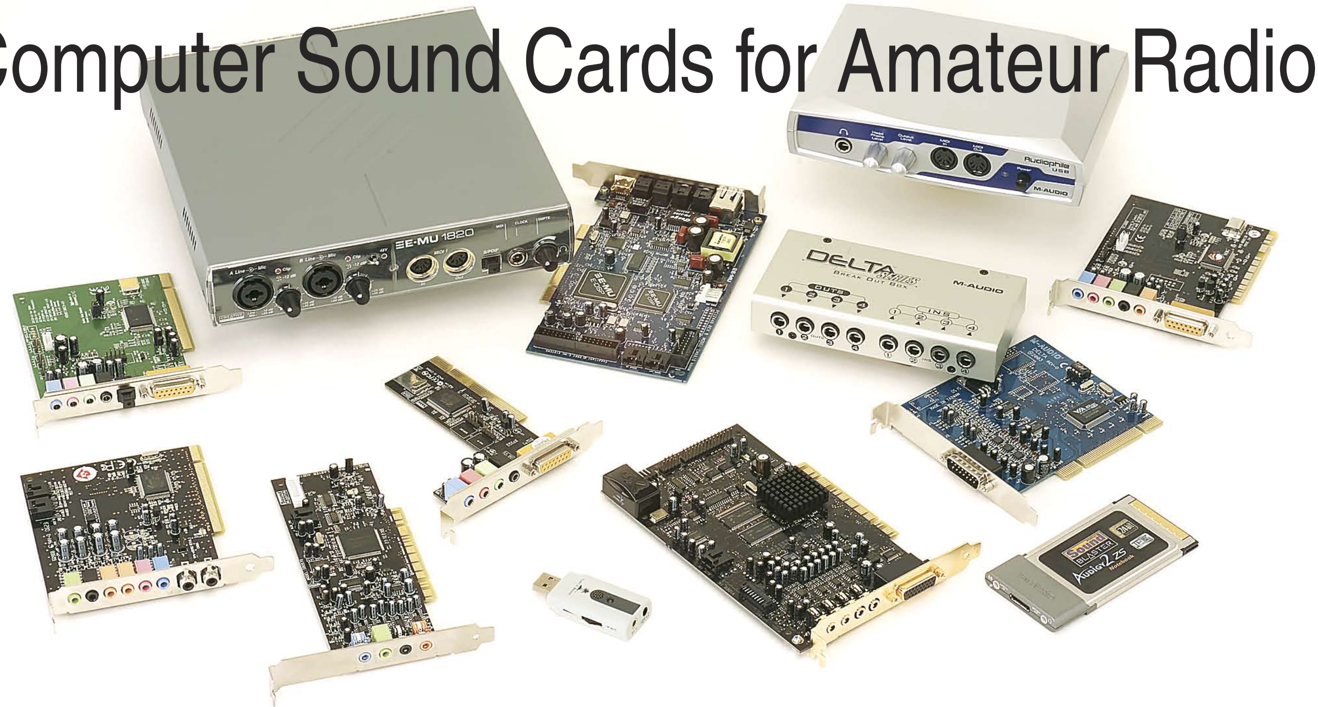
PRODUCT REVIEW

Computer Sound Cards for Amateur Radio



Reviewed by Jonathan Taylor, K1RFD
ARRL Technical Advisor

Not many years ago, the familiar picture of a ham shack featured an HF transceiver front and center. But sift through the last batch of QSLs you received, and you'll see plenty of stations with a big computer screen in the middle. Logging, contesting and digital-mode software is inching the computer closer to the middle of our stations — and in the case of software-defined radios (SDRs), the computer *is* the radio.

The recent growth in digital modes has been fueled by a fresh set of Amateur Radio software applications that put the computer's *sound card* to work. Digital signal processing (DSP) techniques have been around for decades, but only in recent years has it become practical to run DSP algorithms on the main CPUs of home computers. These programs use the computer's sound card as their input-output device, translating signals between the analog and digital worlds. For HF digital modes and certain SDR systems, the computer has become an IF stage, with the sound card acting as its front end — its window to the outside world.

Sound has been a built-in feature of Macintosh computers from the very beginning, but it started out as something of a curiosity on the IBM PC platform. The original Sound Blaster card plugged into one of the PC's industry standard architecture (ISA) expansion slots and eked out noisy 8-bit waveform audio, along with crude sounds from its musical instrument digital interface (MIDI) synthesizer. The brave PC owners who installed it were faced with setting jumpers for several interrupt request (IRQ) lines, input/output (I/O) ports, and direct memory address (DMA) addresses, and then editing their *DOS* or *Windows* configuration files in several places, just to get the thing going — fingers and toes firmly crossed.

Life is easier now with peripheral component interconnect (PCI) slots and Plug-and-Play devices. Usually, adding a sound card is as easy as opening up the PC, sticking in the new card and firing it back up. Ironically, PC motherboards now typically come with sound hardware built in, so we usually install a sound card only if we want to upgrade to a higher quality device or need to add a second device for more flexibility. In fact, there are a number of USB-connected sound devices available that can be great choices as second devices.

But aside from convenience, are some sound cards better than others? After a brief introduction, we'll take a look at what ARRL Lab tests revealed about some popular models. But first, a note about terminology: Computer sound devices are often called "sound cards" even if they're not actually packaged as a circuit board. In this article, the terms *sound card* and *sound device* refer to any device that a computer uses to render and capture sound, regardless of how it's packaged.

Bottom Line

There are clear performance differences between inexpensive 16-bit sound cards and the more expensive 24-bit models. For most digital mode users, any of these cards will perform well. Software defined radios and other high-performance applications will benefit from a high-end card.

Sound Cards 101

The basic function of a computer sound card is relatively simple, yet not widely understood. Your computer's sound device is basically an analog-to-digital (A/D) and digital-to-analog (D/A) converter that operates at audio frequencies. Most sound cards, in fact, can operate in both directions at the same time, doing A/D and D/A in *full-duplex* mode. The sound card runs at a software-selected sample rate. Common sample rates are 8000, 11,025, 22,050 and 44,100 samples per second. For each tick of the sample-rate clock, the card performs one D/A or A/D conversion per channel.

On input, it samples the voltage at the input jack, converts it to a binary value, and stores this number in a memory buffer. After a buffer-full of samples has been assembled, it passes the buffered data to a driver, which hands it off to the application that needs it. The result is a digital representation of the original analog signal. The application can then use digital signal processing (DSP) techniques to convert the stream of bits to

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High End versus Low End Performance

Just how much difference is there between the high end 24-bit sound devices and the most basic 16-bit units? The plots presented here show *RightMark Audio Analyzer* (RMAA) loopback test results for the high end Creative E-MU 1820 (blue traces) compared to the CompUSA generic PCI sound card (red traces). (In the loopback test, the sound card output is connected to its own input.)

On each plot, the horizontal axis is frequency in Hz (Logarithmic scale), and the vertical axis is in dB. In all cases, the data from a single audio channel is shown (the right channel, for most graphs), with the test results from the other channel being similar.

See the "RMAA Performance Tests" section of the text and Table 1 for more information. — *Michael Tracy, KC1SX, ARRL Lab Test Engineer*

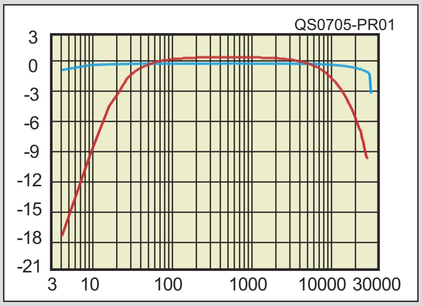


Figure 1 — Frequency response, showing the sound amplitude "roll-off" at high and low frequencies, as well as the accentuation in the midrange. Ideal response would be completely flat at 0 dB.

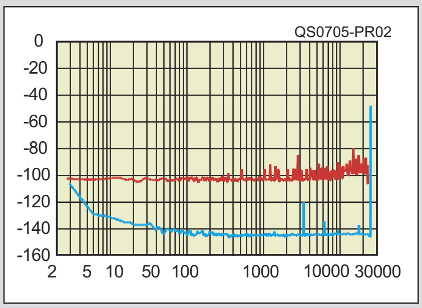


Figure 2 — Sound device internal noise level in dBA. Lower levels represent better performance.

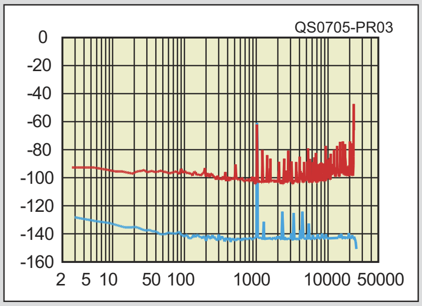


Figure 3 — Sound device dynamic range. A desired signal is generated at 1 kHz at –60 dBA, with other frequencies indicating spurs and noise. Lower spurs and noise show higher dynamic range.

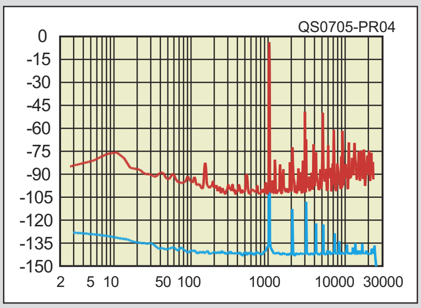


Figure 4 — THD (total harmonic distortion). The desired signal is set to 1 kHz with the amplitude at –3 dBA (close to the sound device's maximum) and distortion products are indicated on other frequencies. The total distortion is reported as a percentage in Table 1 in keeping with the way THD is most often reported.

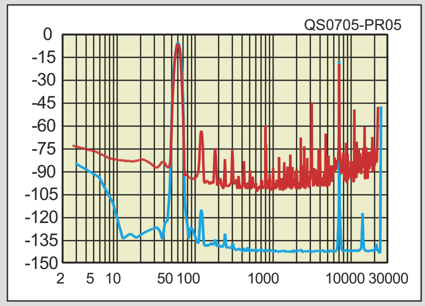


Figure 5 — Two-tone intermodulation distortion, plus noise. Tone frequencies of 60 Hz and 7000 Hz are used.

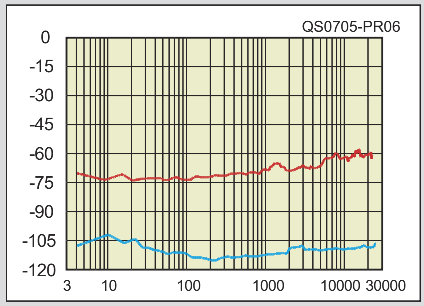


Figure 6 — Crosstalk (leakage), the undesired signal coupling between channels. Lower crosstalk levels indicate better performance.

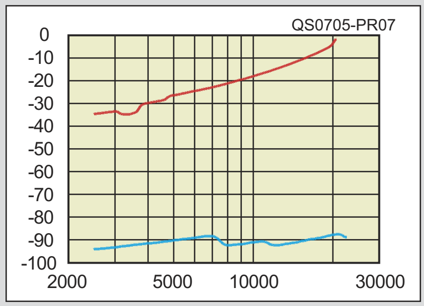


Figure 7 — Swept frequency test of IMD based on two audio tones. This test indicates intermodulation effects that are specific to internal sample rate conversion processing.

something useful, often by filtering it and then detecting its frequency and phase.

In the other direction, the application uses DSP to build a buffer-full of data. It sends it down through the driver to the sound card, and the sound card clocks it out one sample at a time, converting each value to a voltage. After a low-pass filter, the result is a smooth analog waveform, precisely made-to-order by the application that created the buffer.

Modern sound cards usually have plenty of other bells and whistles (almost literally), such as MIDI synthesizers and on-board DSP chips that provide surround sound and reverb effects. These features are rarely used by Amateur Radio sound card applications. For the most part, ham software is built around the basic waveform input/output functions.

Sound devices for PCs fall into four major categories, based on the way they connect to the computer:

- *On-board sound*: Modern PCs, including laptops, usually have a sound chipset directly on the motherboard. These machines have audio input/output jacks installed directly on the cabinet.
- *PCI card*: A sound card can be installed as an option on any desktop PC that has an empty PCI slot, even if the motherboard already supports sound. In fact, more than one sound card can usually be added for special applications. Some PCI devices come in two pieces, with audio jacks on an external breakout box that connects to the PCI card with a custom cable.
- *PC card*: Sound cards are available as PC cards (also known as PCMCIA or CardBus cards) for laptop computers. These credit-card-size devices slide into a slot on the side of the machine, and they can usually be removed or inserted while the machine is still running.
- *USB*: Universal Serial Bus sound devices can plug into a desktop or laptop PC, with the sound chips and audio connectors housed in their own enclosure. This option is typically more expensive than a PCI card, but it can be a lot more convenient.

Sound Card Testing

The ARRL Lab tested 14 makes and models of sound devices, including samples of all four types. Hundreds of sound cards are on the market, and choosing a short list to review wasn't easy. We put together our list by doing an informal survey of the most popular devices available, across a wide range of prices. The cheapest category is the "Brand X" PCI card, which isn't much more than a single chip, a circuit board with a few capacitors, and a set of audio jacks on the back bracket. The most expensive category we tested includes semi-pro devices that are targeted for musical audio production, offering (in theory) the best audio quality and the widest range of input-output options.

Just as a good set of speakers can make all the difference in a stereo hi-fi system, one would expect that a high-quality sound card would give us the best results for DSP applications on the PC. We'll begin by looking at some of the characteristics of sound cards that we can measure.

As we've discussed, sound cards convert analog audio signals to a set of digital samples. This conversion from analog to digital isn't perfect, for several reasons.

Sample Size

When a sound card takes a sample of the input voltage, it expresses it as a binary number with a certain number of bits. This is the *sample resolution,* or *sample size*. The sample resolution determines the number of steps between the smallest and the largest signal the card can measure.

The greater the number of steps and the smaller they are, the more precise the samples will be. Larger steps introduce more *quantization noise*, so a sound card's signal-to-noise ratio is limited by the number of bits of resolution in each sample. For example, a card taking 8-bit samples measures only 256 voltage steps and cannot yield a signal-to-noise ratio (S/N) better than about 49 dB. With 16-bit samples, there are 65,536 steps, and the ideal S/N rises to 98 dB.

Sample Rate

The clock that drives the A/D converter runs at a steady rate, known as the *sample rate*. As you might expect, a higher sample rate is required to accurately capture higher-frequency sounds. A waveform can be accurately captured by sampling at a minimum of twice the highest frequency of interest. Energy at higher frequencies produces *aliases*, so sound cards put a low-pass filter ahead of the A/D converter, running at a cutoff frequency equal to one-half the sample rate. But these filters cannot be perfect, so there's bound to be either some high frequency roll-off, or some distortion due to aliases sneaking through.

Linearity

If the sample steps aren't all exactly the same size, or the clock drifts up and down a little bit in frequency, distortion is introduced. The ideal sound card would have a perfectly linear A/D converter and a perfectly stable clock. Good-quality sound cards do, however, have crystal-controlled oscillators as their clock source.

Other Sound Card Limitations

Even if the clock runs at a stable frequency, we won't get the desired result if it's running at the *wrong* frequency. If we record a sound with a sound card running at 11 kHz, and then play it back through a card running at 12 kHz, we'll hear a very noticeable frequency shift.

Sound cards also have traditional analog audio amplifiers, mixers and filters, all of which can introduce noise, distortion and crosstalk.

A typical sound card has a stereo line-level output, a stereo line-level input, and a monaural microphone input. The microphone input is usually followed by a preamp that can sometimes be switched on or off in the configuration software. Many laptop computers lack a line-level input connector but allow the microphone preamp to be switched on or off in this fashion.

[Continued on page 69]

Table 1
Sound Device Performance Measurements

This table summarizes the performance of the sound devices, as tested in the ARRL Lab. They run the gamut from inexpensive 16-bit PCI cards to those with PCMCIA or USB interfaces, to 24-bit PCI cards with external breakout boxes. For comparison, built-in desktop and notebook chipsets are at the end. Please note: The page references point to the written evaluations; they appear in a slightly different order than the order in which they are listed here.

Sound Device Model	*Freq Response 40 Hz-15 kHz (dB)*	*Noise Level (dBA)*	*Dynamic Range (dBA)*	*THD (%)*	*IMD + Noise (%)*	*Stereo Crosstalk (dB)*	*IMD+N Swept Freq (%)*
16-Bit							
CompUSA, 4-channel PCI (page 00)	+0.74, –5.19	–66.8	66.6	0.495	1.449	–67.5	15.571
SIIG Sound Wave 5.1 PCI (page 00)	+0.28, –3.00	–72.3	68.2	0.205	2.588	–71.1	21.030
Turtle Beach Riviera PCI (page 00)	+0.52, –0.40	–73.0	69.6	0.196	2.688	–74.7	15.751
Diamond Xtreme Sound 7.1 PCI (page 00)	+0.22, –1.64	–83.4	81.4	0.059	0.067	–84.9	1.611
Turtle Beach Audio Advantage Amigo USB (page 00)	+4.08, –5.32	–62.8	62.6	0.113	0.158	–79.3	0.208
RigExpert Standard (page 00)	+2.01, –5.22	–85.3	84.5	0.070	1.046	–86.7	0.066
24-Bit							
Creative Sound Blaster Audigy 2 ZS Notebook (PCMCIA) (page 00)	+6.07, –3.67	–90.6	90.0	0.0033	0.0079	–91.5	2.596
Creative E-MU 1820 (page 00)	+0.04, –0.37	–110.6	110.2	0.0007	0.0014	–113.3	0.003
M-Audio Delta 44 (page 00)	+0.02, –0.07	–99.0	98.8	0.0011	0.0053	–95.9	0.0058
M-Audio Audiophile USB (page 00)	+0.02, –0.03	–94.0	93.8	0.0016	0.0063	–95.4	0.0070
Creative Sound Blaster Audigy SE (page 00)	+0.08, –0.20	–93.7	90.1	0.0067	0.015	–93.3	0.014
Creative X-Fi Fatal1ty Gamer (page 00)	+2.63, –1.81	–101.1	100.7	0.0017	0.0043	–102.0	0.0094
Built-In							
Desktop on-board sound (page 00)	+0.24, –0.69	–78.7	77.3	0.014	0.038	–76.3	0.031
Laptop on-board sound (page 00)	+0.10, –0.16	–74.8	74.9	0.010	0.056	–75.1	0.054

CompUSA 4-Channel PCI
SIIG SoundWave 5.1 PCI
Turtle Beach Riviera PCI

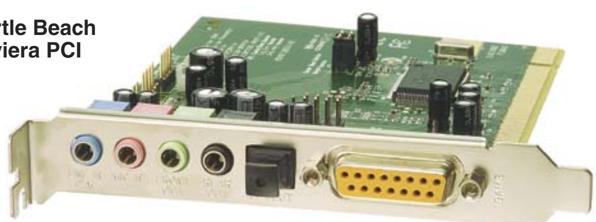
These three PCI cards, all based on the same C-Media chip, are what you're likely to find if you go to your neighborhood computer store and ask for the lowest-price sound cards in the store. The selling point for these devices is that they upgrade your computer's bare-bones audio system to multichannel surround sound. For hams, the main benefit will be the flexibility of having a second (or third, or fourth) sound card in the PC. This lets you dedicate your built-in, motherboard sound device to the usual *Windows* sound effects through the desktop speakers, and run your sound card digital applications through the second card, as long as those applications support it. They're also a good choice for adding basic audio capability to motherboards that don't already have sound built in.

As you might expect, the specs on these 16-bit cards are much less impressive than the 24-bit devices, especially the higher-end models. And beware of claims on the packaging! You might see "32-bit" or "24-bit" mentioned in the descriptions, but these refer variously to the PCI bus width, the digital I/O capability or the playback sample size. All three actually have 16-bit A/D converters inside.

CompUSA 4-Channel PCI

SIIG SoundWave 5.1 PCI

Turtle Beach Riviera PCI



Device type: PCI
Max full-duplex sample size/rate: 16-bit/48 kHz
Mic/line input mixing: No
Switchable mic preamp: Yes
Line input: Yes
Stereo inputs: Yes
Pass-through mixing: Yes
Platforms: *Windows*, *Linux*
List price (US): $19.99 to $29.95

Diamond Xtreme Sound 7.1 PCI

The Xtreme Sound is one step up from the entry-level 16-bit PCI cards, as it has the added capability of 24-bit playback. The input side, however, is still 16-bit. In the ARRL Lab tests, the Xtreme Sound also had significantly better IMD and stereo crosstalk figures than most of its 16-bit colleagues.

As with many other PCI cards, there's a second line-level stereo input (called AUX) mounted on the card itself, but not on the back panel. Diamond also ships the card with a simple but useful sound editor called *Audacity*, which can work with either WAV or MP3 files.

Diamond Xtreme Sound 7.1 PCI

Device type: PCI
Max full-duplex sample size/rate: 16-bit/48 kHz
Mic/line input mixing: No
Switchable mic preamp: Yes
Line input: Yes
Stereo inputs: Yes
Pass-through mixing: Yes
Platforms: *Windows*, *Linux*
List price (US): $59.99

Turtle Beach Audio Advantage Amigo USB

The package reads, "Superior to built-in notebook audio." This USB device is marketed as an audio upgrade for laptop computers, or as a convenient way to plug in a VoIP headset. It's not much bigger than a 9 V battery, but it has a thumbwheel volume control, a mute button and even a built-in microphone in addition to the standard mic and headphone jacks. It comes with a short USB extension cable, but it can also be plugged directly into the USB port if space permits.

A USB device is tops in convenience if you're looking to add a second device to a machine that already has built-in sound. The Audio Advantage Amigo is as small and simple as it gets, but other USB sound devices give you a broader set of input/output options, if small size isn't important.

Device type: USB
Max full-duplex sample size/rate: 16-bit/48 kHz
Mic/line input mixing: No
Switchable mic preamp: Yes
Line input: No
Stereo inputs: No
Pass-through mixing: Yes
Platforms: *Windows 2000* or above
List Price (US): $39.95

Turtle Beach Audio Advantage Amigo

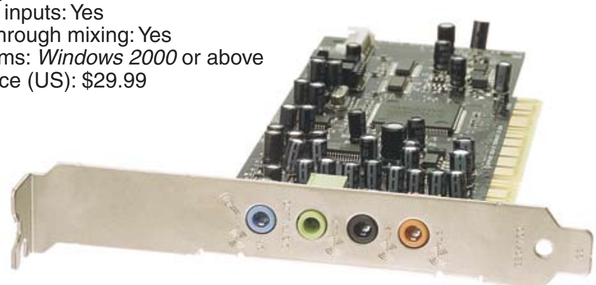
Creative Sound Blaster Audigy SE

The Audigy SE is near the low end of Creative's product line, providing a basic upgrade to the audio on the PC's motherboard. The main benefit is the Audigy's 24-bit sampling capability. This puts it a cut above the basic 16-bit PCI cards, at least for software that can take advantage of the wider samples.

Although the Audigy has a built-in mic preamp, the microphone and line-in channels share the same back-panel connector. This means it isn't possible to connect one signal source to a microphone input and another to a line input and switch between them using the audio control panel.

Device type: PCI
Max full-duplex sample size/rate: 24-bit/96 kHz
Mic/line input mixing: No
Switchable mic preamp: No
Line input: Yes
Stereo inputs: Yes
Pass-through mixing: Yes
Platforms: *Windows 2000* or above
List price (US): $29.99

Creative Sound Blaster Audigy SE



Creative E-MU 1820

In my past life as a radio voice-over guy, this would have been a gem for my home studio if only it had come out 20 years earlier! The E-MU 1820 is a semi-pro device split into two pieces: A PCI card for the PC and a massive desktop or rack-mount unit in its own elegant metal housing, called the AudioDock. The two are interconnected by a shielded cable with RJ-45 connectors. With four sets of stereo inputs and outputs, the outboard box has no fewer than 30 jacks for everything from MIDI instruments to balanced microphones to digital devices. The two-piece combination even takes extra power directly from the PC's power supply through a special power cable. One benefit of the two-piece design may have been that Creative engineers were able to keep the most noise-sensitive analog circuits outside of the PC and safely tucked away in a well-shielded enclosure, leaving the PCI card to handle the remaining digital tasks.

The E-MU 1820, at the high end of Creative Labs' product line, looks like a perfect choice for a home studio, but a less-than-perfect choice for a ham shack. The audio quality is top-notch, but the balanced stereo connections aren't a great fit for the unbalanced, mono audio lines found in radio equipment — plenty of adapters would be required. The sturdy AudioDock unit is great for hooking up various sound sources without having to crawl around to the back of the PC, but other two-piece devices will give you much of the same convenience, and a lot more bang for the buck. Still, SDR enthusiasts are likely to be impressed by its very low noise and very high dynamic range.

Creative E-MU 1820

Device type: PCI with external chassis
Max full-duplex sample size/rate: 24-bit/192 kHz
Mic/line input mixing: Yes
Switchable mic preamp: Yes
Line input: Yes (4 stereo pairs)
Stereo inputs: Yes
Pass-through mixing: Yes
Platforms: *Windows 2000* and above
List price (US): $399.99

RigExpert Standard

We put this device on our list of test candidates because it's one of the few sound devices built specifically for Amateur Radio use. The RigExpert is meant to be a hardware companion to the powerful *MixW* digital-mode software package, providing a seamless transceiver interface that handles PTT, CW, FSK and sound, all through the same USB connection. It works with most other sound card software titles as well, since its drivers emulate standard serial ports and audio devices.

Connections to the rig, including audio connections, are through prewired cables that are provided when the unit is ordered. The outboard box also has a standard serial connector, providing an extra serial port via USB, which always comes in handy in the shack.

As with other USB audio devices, the RigExpert frees up your computer's existing sound card for doing other things, such as *Windows* sounds and multimedia. It has front-panel controls for adjusting the input and output levels, which can be a lot more convenient than fiddling with the sliders on the *Windows* Control Panel, especially for quick adjustments.

Instead of the usual arrangement of stereo inputs and outputs and a separate microphone input channel, the RigExpert has two monaural inputs and one monaural output. The two input lines can be selected, or even mixed together, using the front-panel controls. This is a good fit for most sound card software, which is single-channel only. The 16-bit codec, which can be run at up to 48 kHz, scored at or near the top of all the 16-bit devices we tested, in terms of noise and dynamic range.

RigExpert Standard

Device type: USB
Max full-duplex sample size/rate: 16-bit/48 kHz
Mic/line input mixing: Two mixable line-level inputs
Switchable mic preamp: No mic input
Line input: Yes (two channels)
Stereo inputs: Yes
Pass-through mixing: Yes
Platforms: *Windows*, Mac *OS X*
List price (US): $303 (including cable)

Creative Sound Blaster Audigy 2 ZS Notebook

This is a PC Card (PCMCIA) device that slides into an open port on your laptop. Like its USB cousins, it's touted as an upgrade to basic built-in audio. In this case, the hardware advantage is 24-bit capability, for applications that know how to use it. The card has input and output jacks of its own, and a two-foot dongle for making connections to surround-sound systems.

As with several other Creative products, the input jack is a combination microphone, line and digital input connector. The actual function is software-selectable. This means that it isn't possible to mix line-level and mic-level signal sources together, but most sound cards don't offer that option anyway, even those that have separate mic and line connectors.

Laptop users who need to add a sound device have the choice between USB and PCMCIA. A PCMCIA card might be a better option for a long-term installation, freeing up the USB ports for peripherals that come and go. It feels more like part of the laptop itself, without the extra cable that USB requires, so portable users might prefer it.

Sound Blaster Audigy 2 ZS Notebook

Device type: PCMCIA
Max full-duplex sample size/rate: 24-bit/96 kHz
Mic/line input mixing: No
Switchable mic preamp: Yes
Line input: Yes
Stereo inputs: Yes
Pass-through mixing: Yes
Platforms: *Windows 2000* or above
List price (US): $99.99

M-Audio Delta 44

Like the E-MU 1820, The Delta 44 is a two-piece product consisting of a PCI card and an external breakout box. The two pieces connect to each other with a special cable with 15-pin D-sub connectors at each end. The breakout box provides four sets of audio inputs and outputs and doesn't require any power supply of its own. In the control-panel software, the user can switch the inputs and outputs among several "pro" and "consumer" signal levels (+4 dBu, –4 dBV, or –10 dBV), and the ¼ inch tip-ring-sleeve jacks can be used with either balanced or unbalanced lines, although balanced lines are recommended by the manufacturer for best performance.

The hardware considers the four input and output jacks to be entirely separate lines, but application software typically treats them as two stereo pairs. In fact, the two pairs of stereo inputs can be set up to look like two separate devices to sound card programs. That feature might be handy if you need to run two different applications at the same time.

The Delta 44 has no microphone preamp, so it doesn't offer microphone-level inputs. Clearly, the designers expected that microphones would be connected through a mixing board or an outboard preamp, as might be found in a home recording studio.

M-Audio Delta 44

Device type: PCI with external chassis
Max full-duplex sample size/rate: 24-bit/96 kHz
Mic/line input mixing: N/A
Switchable mic preamp: No
Line input: Yes (2 stereo pairs)
Stereo inputs: Yes
Pass-through mixing: Yes
Platforms: *Windows*, Mac, *Linux*
List price (US): $199.95

M-Audio Audiophile USB

USB devices are tops in convenience, but what's the tradeoff in performance? In the case of the Audiophile USB, not much! As with some of the other devices we tested, this is a 24-bit, 96 kHz box that has solid specs on dynamic range and low noise, yet installing it is just a matter of plugging it in to any available USB port. The USB option means it's equally useful with a laptop, too. In most respects, the Audiophile USB outperformed the 24-bit PCI internal devices we tested.

The Audiophile USB has one set of unbalanced stereo line-level inputs and outputs. The outputs are provided on RCA (phono) jacks, and the inputs on either RCA or ¼ inch tip-sleeve jacks. It's housed in small plastic desktop enclosure with a headphone jack on the front and headphone and output-level knobs. As a combination audio and MIDI device, there are DIN MIDI jacks on the front panel as well. The device isn't powered through the USB port, however, taking its power instead from a 9 V wall-wart supply, which may be a bit inconvenient in some installations.

Curiously, the device doesn't allow its input level, or its master output level, to be adjusted from the *Windows* Control Panel. The manufacturer recommends using the front-panel volume control knob instead.

M-Audio Audiophile USB

Device type: USB
Max full-duplex sample size/rate: 24-bit/96 kHz
Mic/line input mixing: N/A
Switchable mic preamp: No
Line input: Yes (one stereo pair)
Stereo inputs: Yes
Pass-through mixing: Yes
Platforms: *Windows*, Mac
List price (US): $249.95

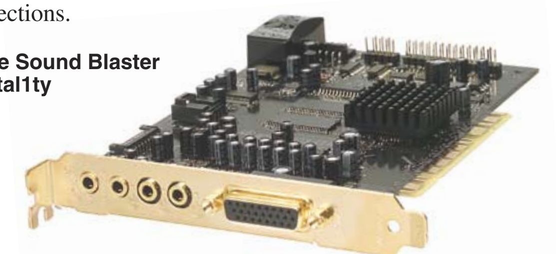
Creative Sound Blaster X-Fi Fatal1ty Gamer

"Hear your enemies before they find you!" screams the product sheet. No, they're not talking about competitive DXing. PC game enthusiasts are an attractive target market for the sound card manufacturers, and Creative aims at them squarely with the X-Fi Fatal1ty, named after a world-renowned PC gamer. (The digit "1" in the middle of the name isn't a misprint.) The cards in the X-Fi series have on-board DSP capabilities with fascinating names such as the 24-Bit Crystalizer and the CMSS-3D Headphone Expander. Depending on which product you buy, and in which mode you run it, these effects promise to add either "stunning audio realism" or "cleaner, smoother music playback." But under the covers is a 24-bit, 96 kHz audio chain with impressive specs, right on the heels of semi-professional sound cards.

The Fatal1ty model can be a one or two-piece device, with a PCI card and an optional input/output box joined by a multi-conductor cable. The breakout box, called the I/O Drive, is designed to fit inside the computer's empty disk-drive bay on the front panel. It features a HEADPHONE jack with VOLUME control, and a set of INPUT jacks for a second line-level or microphone-level input, with a mic gain control. The bracket of the PCI card itself has the primary audio inputs and outputs, so you'll still need to make connections to the back of the computer to get line-level audio output.

Interestingly, the published audio specs of the X-Fi Fatal1ty are exactly the same as another model in the X-Fi line, the XtremeMusic. The main difference between the two is that the XtremeMusic lacks the external I/O Drive breakout box, relying solely on the back-panel audio connections.

Creative Sound Blaster X-Fi Fatal1ty



Device type: PCI with external breakout box
Max full-duplex sample size/rate: 24-bit/96 kHz
Mic/line input mixing: No
Switchable mic preamp: Yes
Line input: Yes
Stereo inputs: Yes
Pass-through mixing: Yes
Platforms: *Windows XP* or above; 1 GHz or faster CPU
List price: $199.99

Motherboard Audio (Desktop and Laptop)

For sake of comparison, we ran the same set of audio tests on the sound devices found on the motherboard of a typical desktop and a typical laptop computer. The audio chipset on motherboards usually comes from only a handful of chip manufacturers that dominate this market. The most common configurations we found were 16-bit, 48 kHz devices, with MICROPHONE, LINE-IN and LINE-OUT jacks on the back panel of desktops, and MICROPHONE and LINE-OUT jacks on laptops, with a few also offering line-level inputs.

Before 2004, the reigning standard for motherboard audio was AC'97, which would provide 16-bit or 20-bit resolution at 48 kHz. More recently, a standard called HD Audio has surfaced, offering more channels and higher quality — up to 32 bits at 192 kHz. If you've just picked up a brand new machine, you might find that it has 24-bit audio built right in. The desktop machine we tested was equipped with the Analog Devices SoundMAX chipset; the notebook computer used the Conexant chips and drivers. Both of these AC'97 devices yielded significantly better performance than the entry-level 16-bit PCI cards we tried.

Motherboard audio capabilities are usually quite basic, but there are a few surprises. Several years ago, a Taiwanese manufacturer came out with a motherboard with a vacuum-tube audio amplifier, apparently to meet the needs of audiophiles who still like the sound of a good tube. The owner's manual had a detailed white paper on the engineering challenges of putting 100 V on the plates of a dual triode that's surrounded on all sides by high-tech silicon. Just think: A perfect match for classic hybrid ham gear such as the Kenwood TS-520 or the Yaesu FT-101!

Device type: Motherboard chipset (AC'97)
Max full-duplex sample size/rate: 16-bit/48 kHz
Mic/line input mixing: No
Switchable mic preamp: Yes
Line input: Yes
Stereo inputs: Yes
Pass-through mixing: Yes
Platforms: *Windows*, *Linux*

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Most Amateur Radio sound card applications have no need for stereo, so they run the sound card in single-channel mode. A notable exception is the FlexRadio SDR-1000, a software-defined radio hardware/software package that has been reviewed previously in *QST*.[1] This system cleverly extracts the in-phase and quadrature (I and Q) components of the received signal, sending I to the left channel of the sound card's input, and Q to the right. This simplifies the design of the RF front end, doubles the effective IF bandwidth, and hands the DSP software precisely the two signals it needs to begin its work.

RMAA Performance Tests

For our performance tests, we chose a software package called the *RightMark Audio Analyzer*, or RMAA. Running RMAA has become a standard technique for measuring sound card characteristics. Several manufacturers publish RMAA test results on their Web sites. The program automatically puts the card through its paces, measuring frequency response, distortion and noise characteristics, dynamic range, and stereo crosstalk, all in just a few seconds. ARRL Lab test results are shown in Table 1, and numbers are shown for the following characteristics.

- *Frequency Response*: An indication of how accurately the sound card converts signals across a wide frequency range. Numbers closer to 0 indicate "flatter" response, which is more desirable. The two numbers show the lowest and highest levels measured across the frequency range, when analyzing a test signal at 0 dB.
- *Noise Level*: A measurement of how much noise is detected by the sound card, relative to a full-power signal, during a period of silence. Since this is a loopback test, the measurement includes noise introduced in both the capture and playback chains. The inverse of this number is one way to express *signal-to-noise ratio* (S/N); think of it as the ratio of the smallest and largest signals the sound card can convert.
- *Dynamic Range*: An estimate of the noise level in the presence of a small signal, or the weak-signal signal-to-noise ratio. Higher values indicate better linearity when converting weak signals. A sound card that has a very low noise level might still perform poorly on weak signals.
- *THD (total harmonic distortion)*: The amplitude of the sum of harmonics of a sine-wave test signal, as a percentage of the test signal's amplitude. This measures undesired signals introduced by the sound card itself when recording or playing back a single, strong tone.
- *IMD+Noise (intermodulation distortion plus noise)*: A measurement of distortion and noise on a complex test signal. This test uses a two-tone test signal, and measures the undesired harmonic and mixing product energy (and noise) at other frequencies.
- *Stereo Crosstalk*: The amount of leakage between the left and right channels of a stereo pair. A test signal is fed into one channel, and measured on the other. Also called *stereo separation*. Larger-magnitude negative numbers indicate better separation.
- *IMD+N Swept Freq (intermodulation distortion plus noise)*: This is similar to the IMD+Noise test, except that the tests use a pair of tones that sweep across the card's entire frequency range.

There are several ways to use RMAA, and we chose the "loopback" mode, which measures the input and output characteristics of the card in the same test. An audio cable is connected from the card's output to its input; the program generates test signals and simultaneously takes measurements, running the signal through the DAC and the ADC at the same time.

Loopback mode testing means we won't be able to distinguish between input and output performance, because these characteristics are being lumped together in a single set of results. This means, for example, that our dynamic range measurements might not match the manufacturers' specs for input or output performance measured individually. But it is still useful to us for doing side-by-side comparisons among sound cards, since we're equally interested in both input and output performance for most Amateur Radio applications.

It's also worth noting that RMAA checks the actual performance of the entire card, not just the specs of the sound chips themselves. This gives us a more accurate picture of how a sound card will truly perform. See the sidebar, "High End Versus Low End Performance" for more insight.

Sound Card Applications

Amateurs have used sound card software for modulating and demodulating virtually any kind of signal that can be expressed as an audio waveform. Frequency-shift and phase-shift modes such as RTTY and PSK31 can be produced by generating an audio signal and feeding it to the microphone jack of an SSB transmitter, and then demodulating it using a receiver with a BFO. The same setup works for analog modes such as slow scan TV (SSTV), since the continuously-varying signal can be expressed as audio. It also works for FM digital modes including packet, in which an audio signal frequency-modulates a carrier.

Sound card applications are useful for conventional voice modes, too. A sound-equipped PC can be used as a voice keyer or a digital audio recorder by storing an audio signal as a disk file, and then playing it back. You can put a full-duplex sound card to work as a digital audio filter or as a transmit speech processor. A computer with a sound card and Internet access can function as an Internet voice gateway for systems such as *IRLP* or *EchoLink*.

A Real-World Test

Signal processing for HF digital modes probably won't push PC sound cards to their limits. Signals arrive with plenty of noise of their own. Their bandwidth rarely exceeds a couple of kilohertz, and modes such as RTTY and PSK31 are interested only in the signals' frequency and phase characteristics, rather than amplitude. But to prove the point, we tested two sound cards at opposite ends of the performance spectrum to see how they would handle a real-world PSK31 signal. The two devices chosen for this test were the entry-level *CompUSA* PCI sound card and the high-end *M-Audio Delta 44* device.

For this test, we installed both devices in a *Windows XP* machine, and then fired up two instances of *DigiPan* PSK31 software, configuring one instance for each device. We then connected the output of an Elecraft K2 transceiver to both sound card LINE inputs using a Y-adapter, tuned to a noisy (and busy) PSK31 frequency, and noted the results.

Not surprisingly, the two *DigiPan* outputs were nearly identical. All signals that produced solid copy on one sound card also gave solid copy on the other, and those with a few "hits" due to fading and noise produced about the same number of missed characters on both. As we expected, the significantly weaker signal-to-noise and THD figures of the generic 16-bit sound card didn't affect *DigiPan's* ability to detect 180° phase shifts in the received signal or its ability to isolate the desired signal from the interference.

Software-defined radio applications are quite a different story. For SDRs that use the PC's sound card as their A/D device, the performance of the card is critical to the overall performance of the system. The limiting factors in DSP systems are the precision of the numbers they use, the speed at which they can be manipulated, and the quality of the samples themselves. If the analog part of an analog-digital system is its weakest link, it's not unreasonable to believe that a high-quality sound card will give us noticeably better performance for our SDR.

Timing Issues

Sample rate accuracy can be important for analog modes that aren't continuously synchronized. For these modes, one sound card is generating a signal and the other is receiving it, and the two cards are expected to be running at exactly the same sample rate.

[1]R. Lindquist, "FlexRadio Systems SDR-1000 HF + VHF Software Defined Radio *Redux,*" Product Review, *QST,* Oct 2006, pp 66-71. *QST* Product reviews are available to ARRL Members on the Web at **www.arrl.org/members-only/prodrev/**.

Windows Vista

Earlier this year, Microsoft released its first major update to the *Windows* operating system in more than five years. A long list of enhancements, fixes, and other changes are bundled into *Windows Vista*, which now comes preinstalled on most new PCs.

As you might expect, *Vista* has stiffer hardware requirements than any of its predecessors. Some computers sold within the past year are labeled WINDOWS VISTA CAPABLE, which means they'll be able to take advantage of most, if not all, of the new features. But older PCs are a different story, and it's wise to visit the Microsoft Web site and run a special compatibility-checker program first if you're thinking about an upgrade.

When *Windows XP* was introduced, the companies that make add-on components like video cards, sound cards and network interfaces scrambled to get a fresh set of bug-free device drivers out the door. No doubt we'll see the same mad rush with *Vista*, so watch Web sites closely for frequent updates. *Vista* has some marvelous new features, but many of them require cooperation from the innards of your PC, and it may take a while for all hardware makers to come on board. When I had trouble getting a new serial port card to work in my *Vista* PC, the speedy e-mail reply from the manufacturer was, "This device cannot be used with *Vista.* Thank you for your inquiry."

Software compatibility may be an issue as well. For example, the 64 bit version of *Vista*, which runs on top-end PCs, no longer supports 16 bit applications written for DOS or the original versions of *Windows* (prior to 1995). Unfortunately, a few popular Amateur Radio titles may still fall into this category. It might be wise to make a list of programs you expect to run on your shack PC before upgrading.

For those of us who use sound card applications, there are some big changes in *Vista*. The entire audio "stack" has been rewritten, providing glorious opportunities for programmers, but potential headaches for users of existing programs. The most obvious change is that the *Windows* Volume Control applet is completely different.

If your sound card has several different inputs to choose from — for example, Microphone and Line In — *Vista* lists each input as if it were a separate device entirely. On the RECORDING DEVICES panel, the one labeled with a green check mark is the *preferred* device, which is the one used by any program that doesn't give you a choice. If you're using a program that gives you a choice of input devices, don't be surprised to suddenly see separate choices for Microphone and Line inputs.

If you installed more than one sound card in other versions of *Windows* and used a program that let you choose which one to use, you'd see the name of each sound card listed. With *Vista*, the list is a bit different, showing the name of each *line* rather than the name of the card. So instead of "CREATIVE SOUND BLASTER," it might show "SPEAKERS (CREATIVE SOUND BLASTER)." This probably makes more sense, but it's a change to keep in mind.

Some newer Amateur Radio sound card programs use a layer called *DirectSound*, part of the DirectX technology Microsoft introduced a number of years ago. Many sound card manufacturers began offering hardware acceleration for *DirectSound*, but this boost is no longer available in *Vista* because of its new audio architecture. Microsoft assures that *DirectSound* will still work properly, albeit without the boost.

None of the sound cards tested for this article mentioned *Vista* compatibility in the installation manuals, but several of the manufacturers have now posted official *Vista* drivers on their Web sites for download. If you're planning on a purchase, check the manufacturer's site if *Vista* compatibility is important to you. *— Jonathan Taylor, K1RFD*

Distortion, or even loss of data, can occur if the rates are slightly different.

In SSTV, if the receiving card is running slightly faster or slower than the sending card, the image may appear slanted. To compensate for this, SSTV software usually includes a slant adjustment which lets the user run the sound card at a slightly slower or faster rate.

Voice over Internet protocol (VoIP) applications such as *EchoLink* and *IRLP* send a continuous stream of data from one sound card to another over the Internet. If the sender and receiver aren't running at exactly the same rate, it can cause audio drop-outs, as the buffer at the receiving end becomes empty or overflows.

This is usually less of an issue for modes such as RTTY and PSK31, which synchronize the receiver with the sender frequently. In these modes, the sound card's clock is still used as the timing reference, but exact sound card timing is far less critical.

Conclusions

Although sound has become a standard built-in feature of modern PCs, there are plenty of options for adding internal or external sound devices for more flexibility and better performance. Sound cards run the gamut in price, with the less-expensive ones suited for simple audio tasks, and the higher-end models geared toward hi-fi sound systems and professional-grade musical recordings.

The top performers in these tests were two-piece PCI devices, but the choice between internal and external is mostly a matter of price and personal preference. Fishing around the back of the PC to find poorly marked audio jacks isn't my idea of fun, so I appreciate the convenience and flexibility of the external boxes. Others might favor the internal models to keep the desktop uncluttered.

Most Amateur Radio sound card applications should work perfectly well with the lower-end models. The 24-bit devices do have better specs, but very few digital-mode apps are designed to take advantage of the higher precision. On the other hand, critical applications such as software-defined radios can definitely benefit from the superior noise, dynamic range, and distortion characteristics of the high-performance cards. The best choice for *your* shack will depend on your budget, the flexibility you need, and whether you expect to dive into sound card-based SDRs or other high-performance sound card software.

Jonathan Taylor, K1RFD, ventured into Amateur Radio as WN8TTP in 1974 after tinkering with an old Hallicrafters shortwave set, and has been a CW enthusiast ever since. He's been active in many different aspects of ham radio, including traffic handling, FM and repeaters, RTTY, contesting, and homebrewing. A second interest of his has been computers and software, dating back to the CP/M machines of the 1970s, and leading to the development of the popular EchoStation and EchoLink programs. He received the ARRL Technical Innovation Award in 2002 and the Dayton Hamvention Special Achievement Award in 2003.

Jonathan's first career was as a radio announcer, a journey with stops in Rochester, New York, St Louis, and New York City. He joined the pioneering PRODIGY online service in 1989, where he developed an early audio-streaming system and one of the first browsers for the World Wide Web. He now serves as Vice President, Architecture at a major e-commerce company.