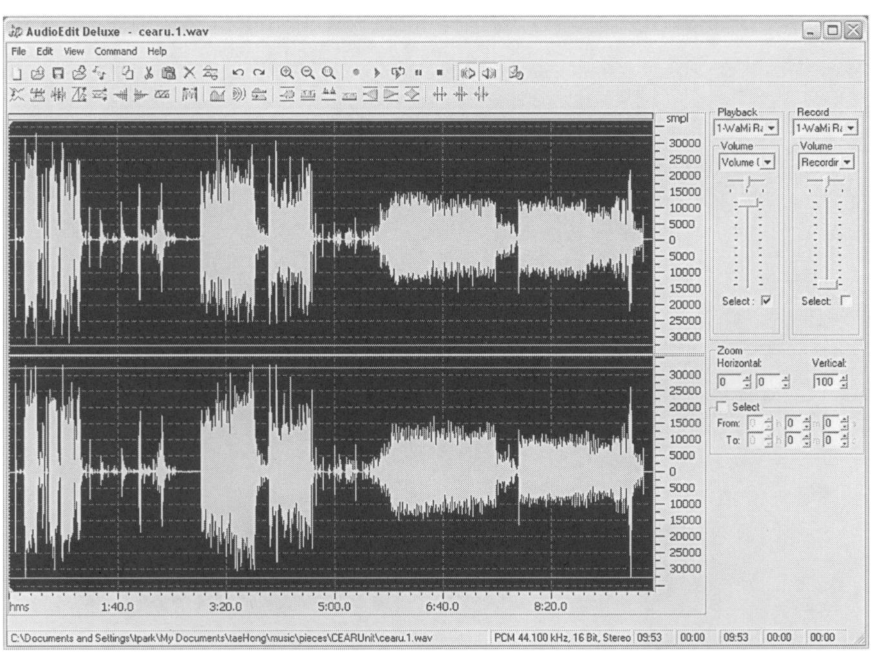
Products of Interest

AudioEdit Deluxe 3.0 Audio Editing Software

Mystik Media has released AudioEdit Deluxe version 3.0, a Windows-platform all-in-one audio editing software package (see Figure 1). Although the program does not particularly exemplify any state-of-the-art technological innovation, it has all the essential and frequently used functionalities that one would and should expect from a soundfile editing package, with added bonus features.

The program can open more than 20 audio formats, and display the waveforms with zoom in/zoom out capabilities. It comes with a built-in audio player and recorder. Standard, but important, features include visual copying/pasting/deleting selections, deleting silence regions, pasting from file, and mixing from file. DSP effects that can be added to the audio samples include amplify, compress, delay, equalize, fade in/fade out, flanger, invert, normalize, phaser, reverb, reverse, silence, shrink, stretch, and vibrato. There are also additional filters such as a band-pass, high-pass, high-shelf, low-pass, low-shelf, and notch. Audio files can be easily embedded with album, artist, copyright, genre, title, and year information.

Figure 1. Screenshot from AudioEdit Deluxe 3.0 audio editing software.



AudioEdit Deluxe 3.0 can read the following audio file formats: WAV PCM; Compressed WAV (ADPCM, GSM, DSP, A-LAW, U-LAW, ALF2, and more); MP3 (MPEG Layer-3); OGG Vorbis; VOX (Dialogic ADPCM); WMA (Windows Media Audio); RAW (PCM, A-LAW, U-LAW); CDA (Audio CD tracks); AU (Unix Audio); and AIFF (Apple Audio). It also provides conversions to and from CD to WAV, CD to MP3, CD to WMA, CD to OGG, WAV to MP3, MP3 to WAV, WAV/MP3 to WMA, WMA to WAV/MP3, WAV/MP3 to OGG, OGG to WAV/MP3, and WAV/MP3/WMA/OGG to CD (burning).

AudioEdit Deluxe 3.0 runs on Windows 98/Me/2000/XP, and may be purchased online from the company's Web site for US$ 40 (special group purchases and site licenses are available). Contact: Mystik Media, 296 Captain Beam Boulevard, Hampstead, North Carolina 28443, USA; telephone (+1) 775-924-4436; electronic mail info@mystikmedia.com; Web audioedit.mystikmedia.com/.

DirectiXer 2.5 VST Adapter for DirectX Host Applications

Kirill "Big K" Katsnelson, a software engineer and computer music zealot running Tonewise, brings us DirectiXer 2.5, a VST adapter for DirectX host applications. The program is designed to wrap VST plug-ins into DirectX-compatible formats. It can be used with standard DAW applications such as Cakewalk, Sonar, Sonic Foundry Vegas and Sound Forge, Syntrillium CoolEdit, and SEK'D Samplitude, among others.

DirectiXer fully supports VST 2.1 instruments, and exposes them as DXi2 instruments, enabling parameter automation for recording and playback. DirectiXer also supports internal plug-in presets and VST banks (FXB) as well program files (FXP), and automatically compensates for VST plug-in delays.

System requirements are DirectX 8 or higher, and Windows 98 or higher. DirectiXer will not run on Windows NT 4.0. The demo version can be evaluated for 30 days after which it can be purchased online for US$ 39. Contact: Tonewise; electronic mail kkm@pobox.com; Web www.tonewise.com/.

Lake Technology Huron Virtual Acoustic Reality Simulation

Lake Technology's Huron system is a custom hardware and software package for acoustic simulation via multiple loudspeakers and headphones. The audio platform is optimized for real-time rendering of 3-D acoustic environments. The hardware is accompanied by a collection of software, ranging from acoustic simulation to digital signal processing (DSP) programming applications.

The expandable Huron PCI and Huron 20 form the core of the Huron audio hardware. Both consist of an expandable Time Division Multiplexed (TDM) bus, allowing the addition of extra DSP expansion boards and input/output (I/O) modules. The bus operates up to 256 channels at 48 kHz sampling rate/24-bit resolution. The DSP boards include four Motorola 56K-series chips along with 12 MB of zero-wait-state dynamic RAM and 1.5 MB fast static RAM. The I/O board allows eight dual-channel I/O modules to be mounted. Both analog and digital input/outputs are provided. The Huron PCI comes with two DSP and one I/O board and is expandable to seven boards. The Huron 20 comes with four DSP and two I/O boards and is expandable to 19 boards. The hardware is controlled by a PC workstation running Windows NT. Multiple Huron systems can be linked together for extra processing power.

The Huron system provides a large set of tools for spatial audio simulations. These simulations can be built around complex geometric models of various spatial environments and can include multiple users. Acoustic simulations can be produced for four to fifty loudspeakers and for stereo headphones using binaural technology.

System Tools includes a virtual rack and patch bay for controlling the audio signal, an I/O manager, a virtual mixer, and a high-resolution equalizer. The simulation software available with the Huron system includes: MultiScape, a framework for simulating multiple users; AniScape, a single user simulation featuring real-time movement; HeadScape, a high-resolution headphone simulation featuring head-tracking along with acoustic rendering developed by CATT, a Swedish company specializing in acoustics and DSP software; SpaceArray, acoustic simulation for large audiences with a wide listening area; and BinScape, a binaural audio simulation for multiple users.

Engineering Tools consists of a convolver that provides up to 278,244 taps in real time, a measurement tool for saving multiple impulse responses simultaneously, and a MatLab interface. Programming Tools are provided to allow the user to create custom applications to control the Huron hardware. A C++ DSP library interface is supplied, along with additional coding and debugging tools.

Integration with third-party Virtual Reality applications is facilitated in part by Huron's support for the Spatial Network Audio Protocol (SNAP) library and CATT-Acoustic (a Windows application for room modeling and room acoustics; for further information, consult www.netg.se/~catt/).

Contact: Lake Technology, Ltd., P.O. Box 735, Broadway PO, New South Wales 2007, Australia; telephone (+61) 2-9213-9000; fax (+61) 2-9211-0790; electronic mail info@lake.com; Web www.lake.com.au/. In North America, contact: Lake Technology Corporation, Suite 201, 340 Brannan Street, San Francisco, California 94107, USA; telephone (+1) 415-861-1147; fax (+1) 415-861-5197.

Tascam GigaPulse Reverberation VST Processing Plug-In

Tascam's new GigaPulse is a high-quality VST-based processing plug-in using impulse responses to render convolution-based reverberation. With the availability of ultra-fast processors in the 1-GHz range, and with enormous hard drives in common use, modeling-based convolution is becoming an even more a realistic alternative to today's ubiquitous delay line/comb filter reverberation algorithms.

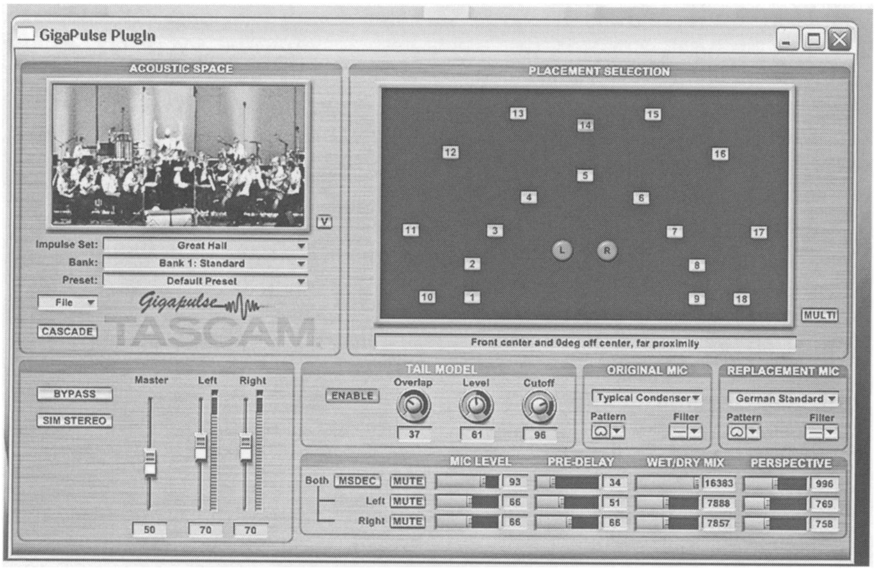
The impulse responses are based on actual recordings of acoustic spaces which have been recorded at up to 18 locations in a given room. Some of the features of GigaPulse include: allowing the source to be virtually moved in space relative to microphone positions; making it possible to move the virtual microphone by means of a user interface to control reverberation while maintaining phase imaging information; a cascading function to morph multiple impulse recordings into one new impulse; and built-in microphone modeling algorithms (see Figure 2). This modeling feature can render vintage Neumann, RCA ribbon, and AKC tube microphones, among others. GigaPulse also gives the user the flexibility to import custom-made impulse responses.

The GigaPulse VST plug-in is available for Windows-based audio workstations supporting the VST protocol. It is listed for US$ 299. Contact: Tascam, 7733 Telegraph Road, Montebello, California 90640, USA; telephone (+1) 323-726-0303; electronic mail custser@tascam.com; Web www.tascom.com/.

junXion 1.2 for Mac OS X

Those of us who use USB game controllers for playing video games and who have tried to hack them into serving as musical controllers should be happy about the release of Amsterdam-based STEIM's junXion (version 1.2) software application for

Figure 2. Screenshot from Tascam GigaPulse user interface.



controlling MIDI-based modules. JunXion, an offspring of MidiJoy (which also happens to be the name of a Japanese rock band as well as a music production and artist development company in Australia), runs on Mac OS X (10.2 or higher) and allows any action by a USB game controller to be translated into MIDI data. The translated MIDI data from keys and joystick maneuvers can be used internally on the computer or transmitted to external MIDI devices via MIDI interfaces (see Figure 3).

Some features of the software include: scaling and polarity selection of incoming data; straight mapping from incoming data to available MIDI events; data mapping of one input to the output of another input (e.g., triggering one note with a button and controlling the pitch of that note with the joystick); MIDI output port selection; simultaneous connection of eight USB input devices (with independent channel and port assignments); and the saving of custom settings as user presets.

STEIM (Studio for Electro-Instrumental Music), headed by Michel Waisvisz, a pioneer in developing human computer interfaces for musical applications, plans to add functionalities such as editable response curves for translation of gestures and an extended set of data conditioners in a future update.

JunXion can be purchased as a download from the STEIM Web site for US$ 29. A demo version is also available. Contact: STEIM, Achtergracht 19, 1017 WL Amsterdam, The Netherlands; telephone (+31) 20-622-8690; fax (+31) 20-626-4262; electronic mail knock@steim.nl; Web www.steim.org/.

Muse Research Receptor Signal Processing Plug-In Player

Muse Research specializes in real-time, media-rich software design, embedded systems engineering, signal processing algorithms, and user interface designs. The company has announced its first product, Receptor, a signal processing hardware and software system.

This player is primarily based on the VST and VSTi plug-in standards developed by Steinberg. The two-space, rack-mountable hardware unit (see Figure 4) comes with a large collection of features and is said to have been designed to stress mobility, reliability, and usability. It functions as a synthesizer, generic effects processor, and guitar processor using 24-bit/96-kHz converters to crunch numbers on a 32-bit mixing architecture.

The player interface includes the familiar knobs and buttons with a small LCD screen, but the unit can also be connected to a computer monitor with keyboard and mouse. This enables the interface to expand into a full-fledged GUI interface (see Figure 5).

Receptor comes with a built-in hard disk populated with a large number of "dial and play" plug-ins which can be updated from the company's dedicated Web site (plugorama.com). Almost every imaginable input/output option is available: a front panel high impedance TRS input; a headphone connector; a USB interface; rear panel balanced analog inputs; S/PDIF stereo digital I/O; MIDI ports; and mouse, keyboard, and computer monitor connections.

Receptor can control over 16 channels of audio with up to 57 effects which can be fully automated via MIDI. The system is compatible with both Macintosh- and Windows-based platforms. Remote communication via Ethernet allows easy management and control from the desktop. It can be integrated into existing audio environments such as Digidesign Pro Tools, Steinberg Nuendo and Cubase, Cakewalk Sonar, MOTU Digital Performer, and Emagic Logic.

Contact: Muse Research, Inc., 970 O'Brien Drive, Menlo Park, California 94025, USA; telephone (+1) 650-

Figure 3. *Screenshot of junXion user interface.*

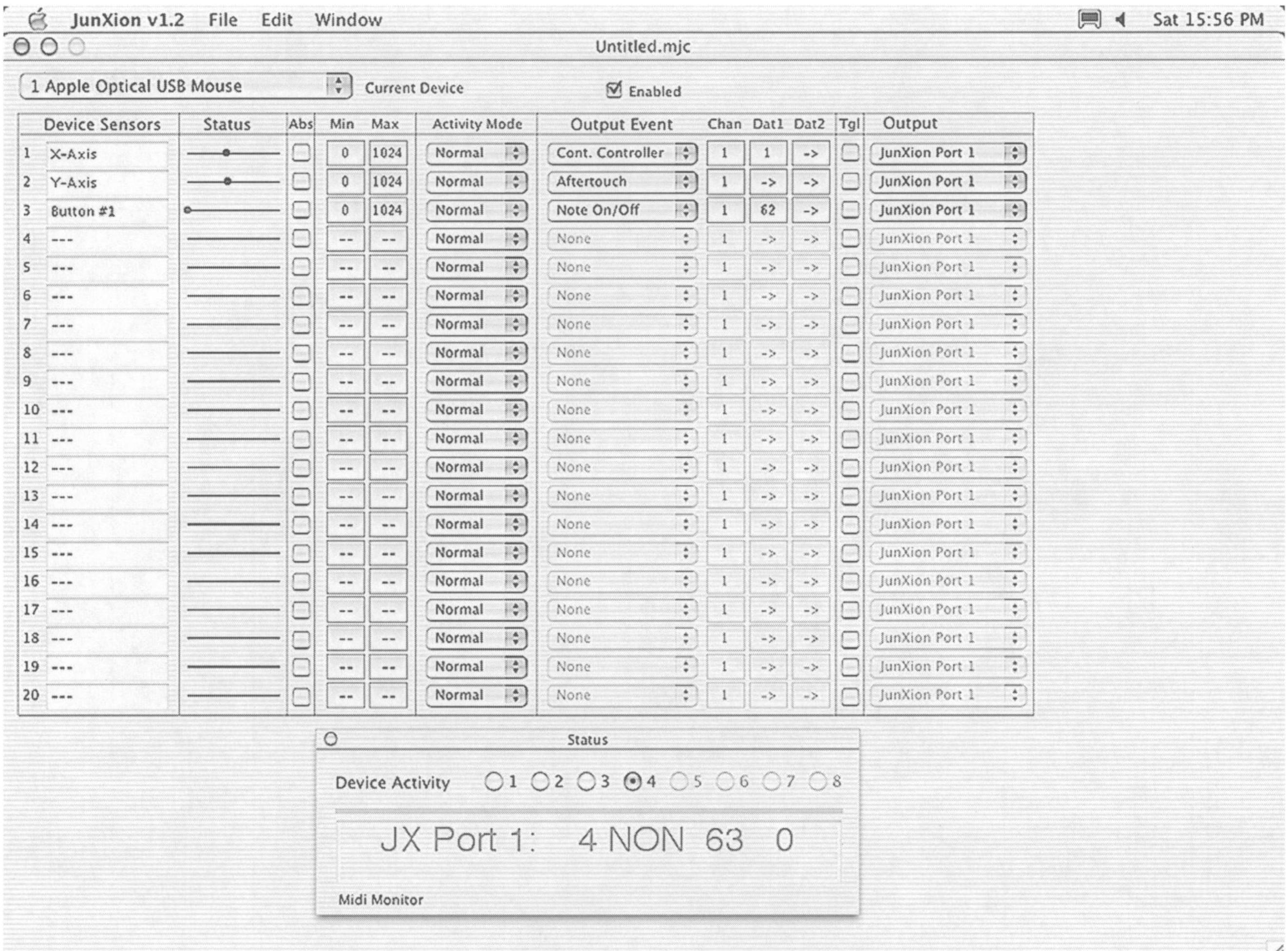


Figure 4. *Muse Research Receptor plug-in player.*

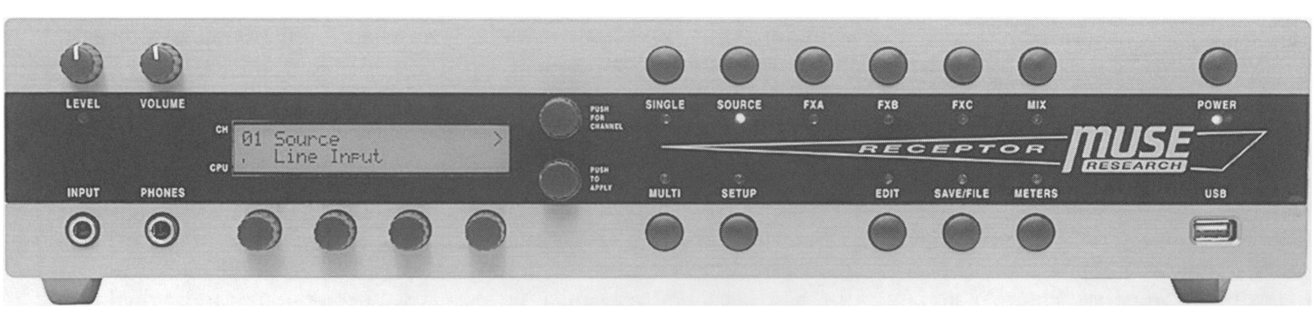
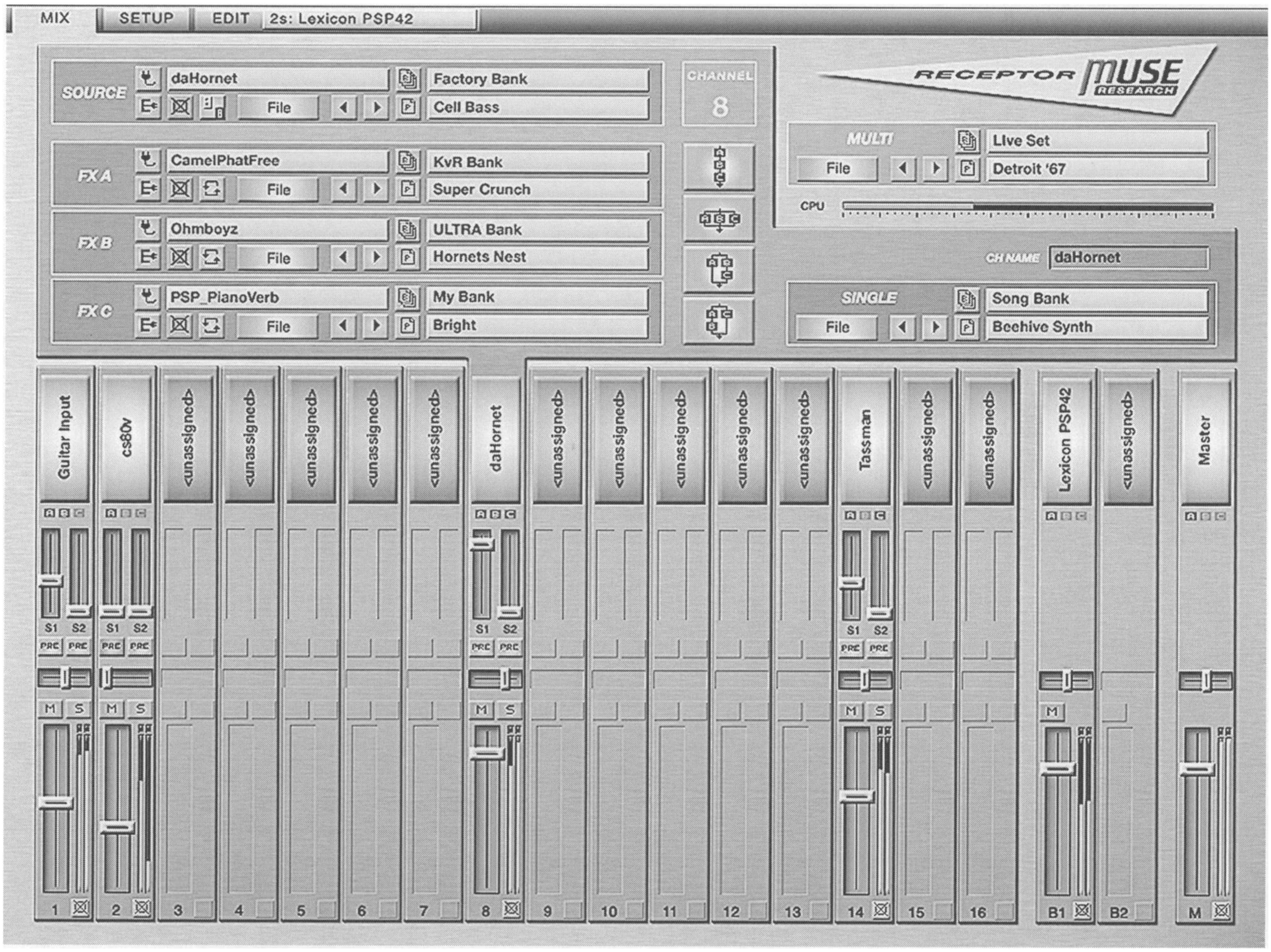


Figure 5. Muse Research Receptor graphic user interface.



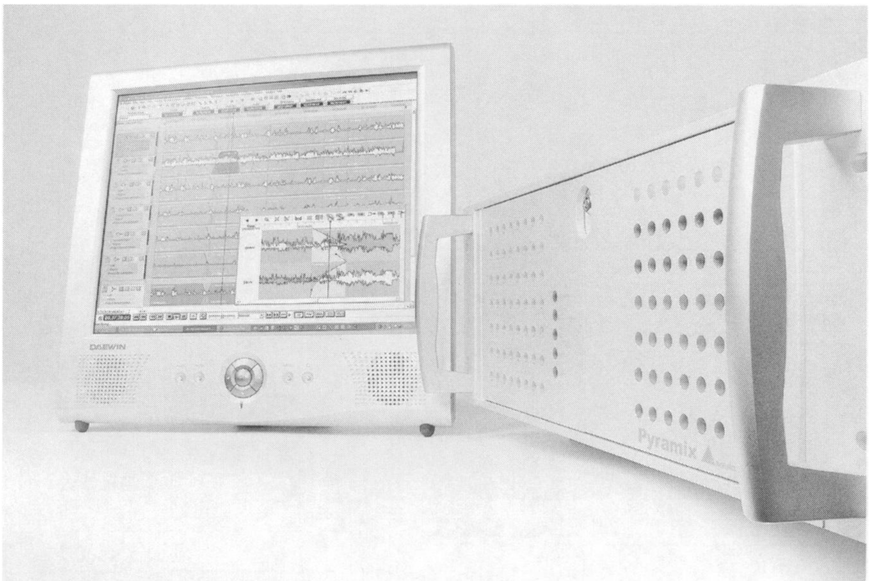
326-5400; fax (+1) 650-326-5401; electronic mail info@ museresearch.com; Web www.museresearch.com/.

Merging Technology Pyramix Virtual Studio 4.1

Merging Technology's Pyramix Virtual Studio is a Digital Audio Workstation that provides hard disk recording, mixing, editing, and effects processing. The combination of the Mykerinos DSP board and the Pyramix software (see Figure 6) allows for sampling rates of 32 kHz to 382.8 kHz, making it appropriate for use with CD, DVD, DVD-A, and DSD (Digital Direct Stream) technology. Pyramix is used for multi-track recording, editing, mixing, and mastering for music, multimedia, and film and television post-production. Mixing and effects processing are carried out in real time.

Among the editing facilities provided by the Pyramix Virtual Studio 4.1 is an advanced cross-fade editor. Symmetrical cross-fades are easily created by dragging adjacent sound clips across each other, and a range of fader controls gives further editing capabilities. The Mixer interface can be controlled in real time by the mouse or by a dedicated hardware controller. Pyramix can support all standard MIDI controllers, control surfaces, and digital consoles that use MIDI. A large number of mixer presets are included and can be edited and recalled using a Configuration Wizard. All surface controls and plug-in parameters can be fully automated. The time code facility of the system is used to store the automation data. The automation controls can be applied to individual mixer

Figure 6. Merging Technology Pyramix Virtual Studio 4.1.



controls or to the entire mixer and can be easily edited with the mouse.

A Phase Meter is included that can be used on a stereo channel or any given pair of channels. A Surround Sound Controller provides various visual and level meter views to aid in the spatial organizing of audio for multi-channel configurations. A Signal Generator can create sine, saw-tooth, and pulse waves as well as noise signals.

The virtual studio comes with a core set of processing tools including a four-band parametric equalizer that offers control of boost, cut, frequency, and bandwidth for each band. A Dynamics Processing module includes a gate, expander, two compressors, a limiter, and a de-esser. A three-band parametric equalizer, a compressor, and an expander are combined as a set of Strip Tools. A three-band parametric equalizer and an advanced limiter are combined as a set of Bus Tools. Both the Strip Tools and Bus Tools modules aim to provide the most frequently used processing and mastering processing tools together in a format that uses minimal DSP processing power.

Pyramix supports VST and TC Works plug-ins. Many third-party plug-ins have been developed for use with the application. These include the De-Noiser, De-Scratcher, and Nova Audio Restoration by Algorithmix. A number of plug-ins by VB Audio Software are also compatible with Pyramix, including Aphro Reverb, C-Limiter, Stereomanager, VB Striptool, C-10 Limiter, EQ 3 Pro, MultiTap Delay, Frequency Analyzer, and Stereo Oscilloscope. The recently developed MPEX2 algorithm has been included in two new plug-ins. Designed in conjunction with Prosoniq, this algorithm allows for time compression and expansion, pitch-shifting, and a 24-to-25/25-to-24 frame time stretcher for film.

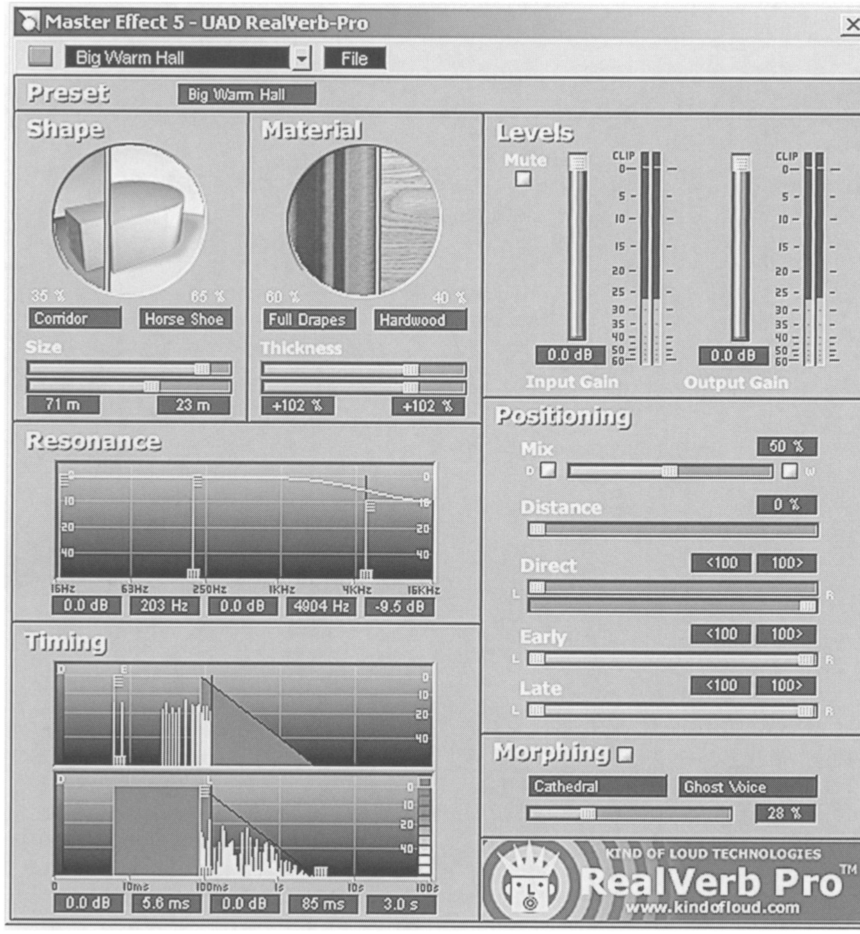
Pyramix offers recording, editing, and mixing at up to 352.8 kHz with 32-bit floating point technology, making it capable of producing audio in multi-track DSD/SACD formats. Surround and stereo SACD Master CDs can be created. The Mykerinos high-performance Audio PCI Card allows for up to 64 inputs and outputs, 64-track playback, and 24-bit/96-kHz monitoring output. Up to eight such boards can be used in parallel or daisy-chained. The DSP card has a modular daughter card architecture with options for ADAT, SDIF, TDIF, AES-EBU and MADI I/O.

High-quality video integration is possible through FireWire (IEEE 1394) support. A Virtual Transport Client-Server architecture allows many applications to interact using one timecode for synchronization. A Cue Sequencer can play cues as individual stereo or multi-channel tracks. These can be entered manually or via MIDI timecode. 16mm and 35mm film formats can be supported in feet or frames.

Pyramix comes in pre-configured software packs for broadcasting, music, mastering, DSD, and post-production, and can also be tailored for user needs. Minimum system requirements demand a Pentium III 500 MHz PC running Windows 2000, NT 4.0, or XP, with 256 MB RAM. An IDE or SCSI hard disk is required and the IDE controller must have bus mastering capabilities. A comprehensive list of recommended PC configurations is available on the Merging Technology Web site.

The Pyramix Virtual Studio comes in various configurations, with related pricing options. Contact: Merging Technology, Le Verney, CH-1070 Puidoux, Switzerland; telephone (+41) 21-946-0444; fax (+41) 21-946-0445. In North America, contact: Merging Technology Sales Office, 3000 Dundee Road, Suite 316, Northbrook, Illinois 60062, USA; telephone (+1) 847-272-0500; fax (+1) 847-272-0597; electronic mail info.america@merging.com; Web www.merging.com/.

Figure 7. Screenshot of Universal Audio RealVerb Pro user interface.



Universal Audio UAD-1 Signal Processing Package for Power Macintosh G5/OS X

Universal Audio has announced the release of its UAD-1 package with software newly optimized for the Power Macintosh G5/OS X platform. The UAD-1 is comprised of a proprietary DSP card that provides the processing power for a range of audio plug-ins, collectively called Powered Plug-ins.

The Powered Plug-ins is a set of tools for audio processing that run on the dedicated DSP card. The currently included plug-ins are based on vintage audio hardware. The 1176LN and Teletronix LA2A are software equivalents of vintage compressors, the former concentrating on vocal compression. The Pultec EQP-1A is based on a vintage valve equalizer, and the Cambridge EQ plug-in offers extensive editing capabilities.

RealVerb Pro (see Figure 7) is a reverberation plug-in that allows the user to define room size, shape, and the material used (for determining reflection/absorption). Dreamverb is built upon this technology and provides a list of room shapes and materials to choose from. The user can further alter these settings by combining and morphing the supplied room settings. The density of the air in the designed room can also be changed.

Nigel is a guitar amplifier simulator made up of seven modules that can be used individually or collectively (see Figure 8). A gate/compressor, phaser, and modulation filter are available on the input. The output side of the amplifier offers tremolo/fade, modulation delay, and echo. The main amplifier module is called Preflex. This provides equalization on the pre- and post-gain signals along with a Morph feature that allows the user to create transitions

Figure 8. Screenshot of Universal Audio Nigel user interface.

Figure 9. Mackie dXb Digital Production Console.

between or mixtures of different types of amplifiers.

Finally, the CS1 channel strip is comprised of the EX1(a five-band parametric EQ), the DM1(a delay modulator), and the RS1(a reflection engine). Every parameter of each plug-in can be automated.

Along with the plug-in software, a performance meter for the UAD-1 card is included with the package, so the CPU and memory usage of the card can be monitored. The card can run all plug-ins at up to 32-bit resolution/192-kHz sampling rate, making it suitable for working with SACD (Super Audio CD) and DVD-A (DVD Audio). The processor uses floating-point values to provide high-resolution and distortion-free signal processing. Support for the Macintosh platform has been provided through Audio Units. VST, DirectX, and MAS (Mark of the Unicorn Audio System) support is also provided. The UAD-1 card is PCI 2.1 compliant and is compatible with the Power Macintosh G5 PCI-X slots.

OS X support is available through Version 3.3 of the UAD-1 software. This allows the use of the Powered Plug-ins with VST and Audio Units on G4 and G5 computers. A beta version is available for download for registered users on the Universal Audio Web site.

The UAD-1 package is listed for US$ 995. Contact: Universal Audio, 330 Encinal, Santa Cruz, California 95060-2101, USA; telephone (+1) 831-466-3737; fax (+1) 831-466-3775; electronic mail info@uaduio.com; Web www.poweredplugins.com/.

Mackie dXb Digital Production Console

When Mackie came out with its d8b digital console, it handily filled the niche between small digital mixers intended for project studios and full-blown, large-scale consoles costing upwards of US$ 100,000 marketed for high-budget studios. The d8b is easy to use, built from reasonably high-quality components (such as 24-bit converters), provides for upwards of 24 inputs/outputs both analog and digital, and includes powerful built-in effects and a range of optional extras. The new dXb sets its sights on a slightly higher market target, more or less that of the Sony DMX2000 digital console (in the neighborhood of US$ 20,000 rather than US$ 10,000 for the d8b). At the same time, it incorporates a number of interesting innovations of design and user interface.

Probably the most striking new elements of the dXb are the two built-in 15-in. touch-screen monitors, displaying levels, routing, dynamics processing, surround panning, and effects for up to 72 channels. In addition, most functions and adjustments can be activated by touching these screens, as well as by manipulating the console's faders, knobs, and buttons, or by using the integrated mouse interface (see Figure 9).

Other features include: a standard operating sampling rate of 96 kHz (24-bit resolution), switchable to 192-kHz operation; a 72-channel input/output matrix (36-channel matrix in 192-kHz mode), with any input able to be routed to any output; Firewire input/output cards for streaming audio directly to/from a computer, including support for PC ASIO or Mac OS X Core Audio drivers; two USB ports for backing up session files to Flash media, etc.; 25 100mm Penny & Giles motorized optical touch-sensitive faders (grouped into two banks of 12 channels, along with a master fader); dynamic effects and parametric equalization on every channel, input and output; on-board support for select VST plug-ins; 8-channel surround mixing and monitoring capabilities; additional built-in effects, including Universal Audio LA-2A and 1176LN compression, Pultec Program EQ, Cambridge EQ, and Dreamverb (accessed through the integrated UAD-1 Powered Plug-ins card); tape-style transport with memory location recall and a Jog/Shuttle wheel; automation system with event editing and dynamic/scene automation; Mackie Control Universal mode for control of digital audio workstation applications (such as Digidesign Pro Tools, Emagic Logic, etc.); various input/output options enabling support for all types of analog or digital connections; and RS-422 (Sony 9-pin), LTC, MTC, MMC, and Word Clock synchronization. The 24 built-in pre-amplifiers feature digitally controlled trim with recallable settings.

The power for the dXb is provided by a Pentium-based CPU running proprietary Mackie software on an Embedded Windows XP operating system.

Depending on the options selected (I/O cards, etc.), the Mackie dXb costs around US$ 20,000. Contact: Mackie/LOUD Technologies, 16220 Wood-Red Road NE, Woodinville, Washington 98072, USA; telephone (+1) 800-858-LTEC or (+1) 425-892-6500; fax (+1) 425-487-4337; Web www.mackie.com/products/dxb/.

New Audio Processing and Recognition Technology from Fraunhofer Institute

Fraunhofer Institute for Integrated Circuits IIS, known above all for its MP3 encoding technology, has released several new applications for processing digital audio. Among these are three products of particular interest to computer musicians.

AudioID is a new system for automatic identification and fingerprinting of audio. This automatic identification/recognition utility works from a database of registered works. The underlying technology is built on the MPEG-7 audio standard. The fingerprinting process extracts a unique "signature" from the signal on the basis of information packaged with what should become the standard MPEG-7 metadata set. The system is designed to be robust, carrying the ability to match the signature of a signal with those stored in a database even when the signal is subject to acoustic interference.

Fraunhofer has also released a melody recognition system. Query-by-Humming is based on the melody representation format defined in the MPEG-7 audio standard. The melody, either sung, hummed, or played on an instrument, is recorded and analyzed for melodic and rhythmic features. Pre-processing reduces background noise, after which fundamental frequencies are analyzed and transformed into a pitch contour, with each note measured for temporal duration. This contour is divided into "phrases." These attributes then form the basis of a database search for matching information. The melodies most closely matching are presented in ranking order for final verification. Additional information about the songs is also presented, such as title, composer, artist, lyrics, etc.

IOSONO is an application of wave field synthesis to allow the reproduction of accurate spatialization of sound within a larger listening area (an expanded "sweet spot"). The basis for this technology comes from the principle of wave propagation whereby the wave field of a source may be reproduced by many sources located on the perimeter of the original wave field. Arrays of closely spaced loudspeakers are driven by unique signals that depend on the position of the virtual source in space and on the position of each loudspeaker. In this implementation by Fraunhofer, the level, position, and distance of each sound source is recorded and processed separately from the acoustic characteristics of the room, enabling the information to be manipulated independently. The 3D-Audio profile of the MPEG-4 standard is used to store and distribute the audio sources and their positions and characteristics. In one installation, at the Lindenlichtspiele movie theater in Ilmenau, Germany, the room is lined with 24 panels of eight two-way loudspeakers, each amplified independently. The IOSONO software is integrated with a Dolby Digital sound processor, enabling material encoded for 5.1 surround-sound to be presented, along with stereo sources and other formats.

Contact: Fraunhofer Institute for Integrated Circuits IIS, Am Wolfsmantel 33, 91058 Erlangen, Germany; telephone (+49) 91-31-7760; fax (+49) 91-31-7769-99; electronic mail info@iis.fraunhofer.de; Web www.iis.fraunhofer.de/.

SRS Circle Surround VST Pro Audio Processor

SRS Labs has released a suite of plug-in processors for encoding and decoding surround-sound audio. Designed as an alternative to Dolby Pro Logic II, the SRS Circle Surround Pro encodes signals up to 6.1 format for stereo transmission or storage, and decodes two-channel signals for distribution over multi-channel formats up to 6.1 surround. The plug-ins conform to the VST protocol and can be applied from within any VST-compatible audio workstation environment (see Figure 10).

The SRS plug-ins can handle sampling rates up to 96 kHz. The Encoder can accept audio projects in the following formats: LCR, LCRS, 5.0, 5.1, 6.0, and 6.1. Full-bandwidth playback of three distinct rear channels has been implemented for 6.1 surround projects. Selectable high-pass filters are provided on the main channels for greater control over low-frequency content and the low frequency effects (LFE) channel. The

Figure 10. Screenshot from SRS Circle Surround user interface.

Encoder is compatible with compression codecs running stereo bit rates as low as 64 kbds (optimum audio performance requires a bit rate of 192 kbps). It is also said to be backward compatible with matrix decoders such as Dolby Pro Logic and Pro Logic II (playback performance is subject to the limitations of the specific decoder).

The SRS Decoder operates in modes for circle surround, LCRS, stereo, and mono outputs. SRS has developed two proprietary processing features to enhance the decoding stage: Dialog Clarity, which applies vertical head-related transfer functions to enhance the content sent to the center channel; and TruBass, which selectively boosts low bass harmonics.

The SRS Circle Surround Pro plug-ins are intended to be equivalent in quality and ease-of-use to the SRS CSD-07 Circle Surround Hardware Encoder/Decoder. System requirements include Windows XP (1 GHz processor or higher), minimum 384 MB RAM, and an approved MME or ASIO-compliant soundcard; a Macintosh system requires a G4 500 MHz processor or higher running OS X 10.2 or higher, and minimum 384 MB RAM.

The SRS Circle Surround VST Pro Suite is listed for US$ 499.95. A seven-day trial version is available for download. Contact: SRS Labs, Inc., 2909 Daimler Street, Santa Ana, California 92705, USA; telephone (+1) 800-243-2733 or (+1) 949-442-1070; fax (+1) 949-852-1099; electronic mail sales@srslabs.com; Web www.srslabs.com/.

Waves Transform Signal Processing Software Bundle

Waves has released a new software bundle, expressly designed for sound transformation. The Transform Bun-

Figure 11. Screenshot of Graphic Tool interface for Waves Sound-Shifter processor.

Figure 12. Screenshot of user interface for Waves Doubler processor.

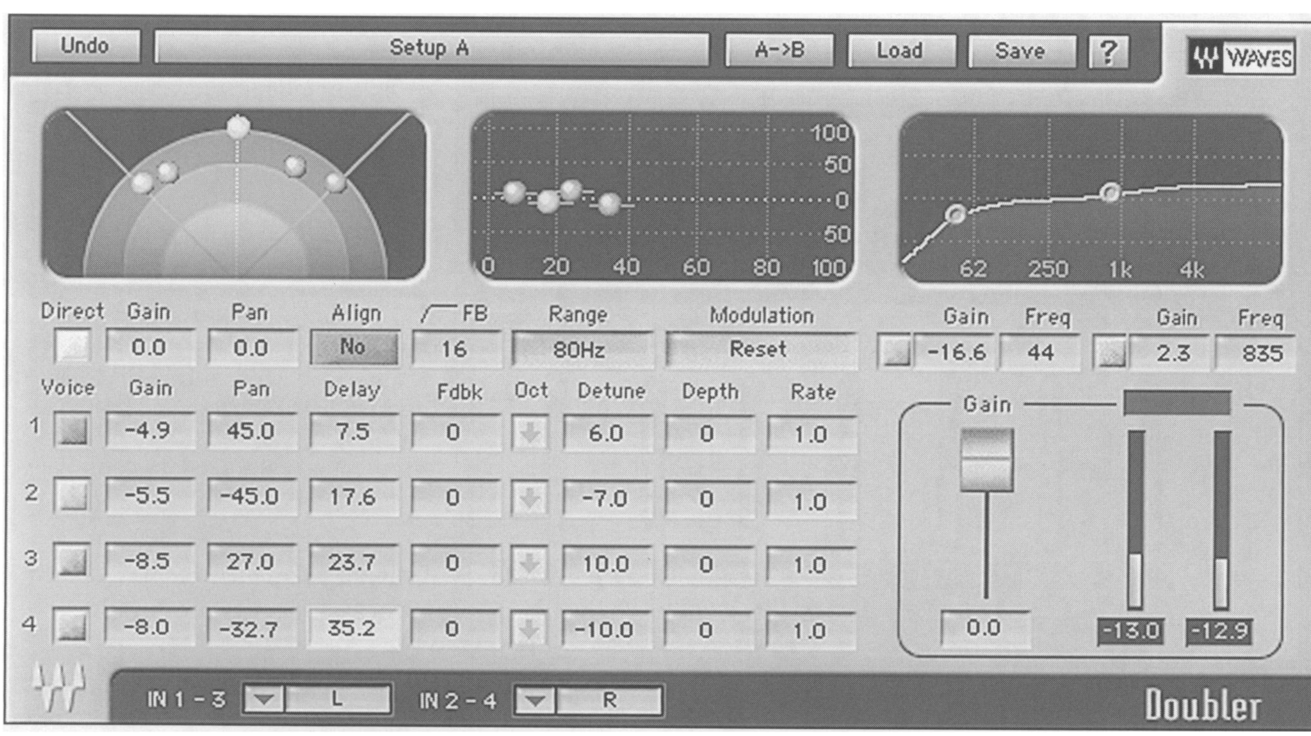
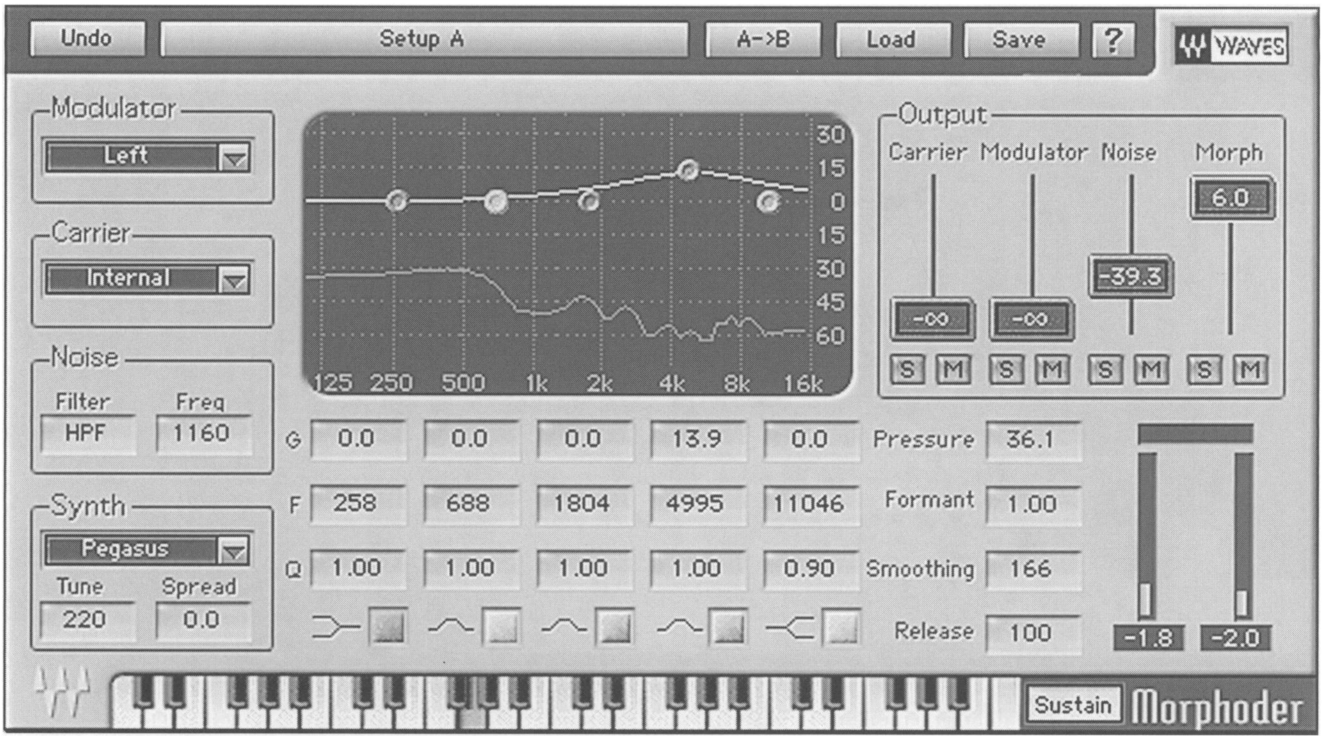


Figure 13. Screenshot of user interface for Waves Morphoder processor.



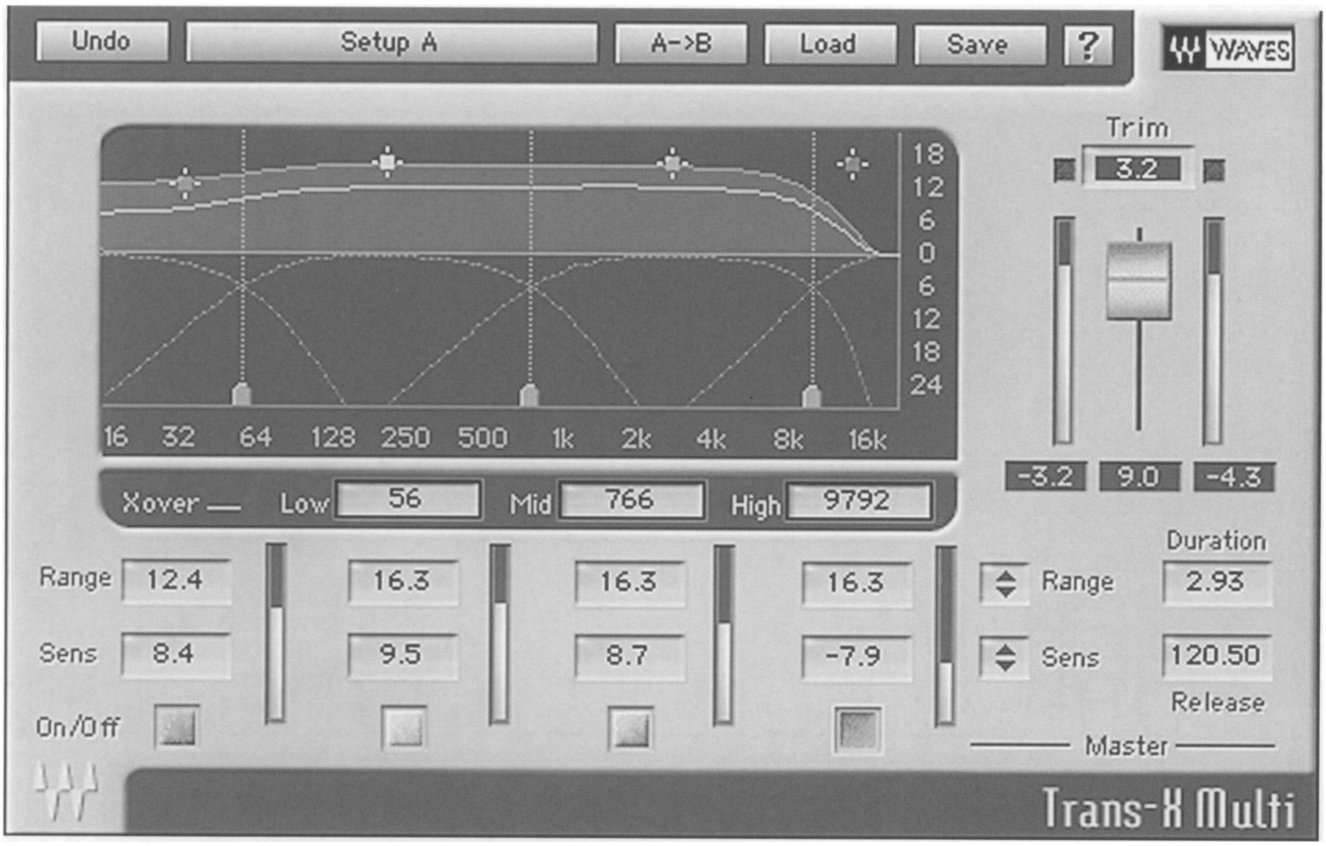
dle, available in Native and TDM versions, is designed to run within Digidesign Pro Tools (Windows-based platforms as well as Macintosh OS 9 or X), and Version 2 supports all plug-in formats (AudioSuite, RTAS, VST, MAS, Audio Units, DirectX), enabling the software to be run within all common audio host applications. The Transform bundle consists of four processors: SoundShifter, Doubler, Morphoder, and Trans-X. Extensive automation and setting storage is integrated into all of the plug-ins.

The SoundShifter is designed for pitch-shifting and time stretching/compression, with particular attention having been paid to preserving transients, clarity, and timing accuracy. The plug-in has three tools for manipulating the soundfiles: Parametric, for off-line (not real-time) time/pitch scaling with fixed ratios; Graphic, for scaling with variable automated ratios (see Figure 11); and Real Time, for immediate pitch shifting. There are four algorithm modes, to enable the user to optimize the processing for specific needs: Sync, Smooth, Transient, and Punchy. There are also four link modes, for defining the relationships between pitch and time manipulations: Unlinked, Time, Pitch, and Strapped. The interface can display time and pitch in whatever ways are most useful for the project: time, tempo, bars, samples, SMPTE, feet & frames; pitch interval, or frequency. For those working with Pro Tools TDM systems, SoundShifter can be integrated with the TDM Time-Trimmer Tool.

Doubler is for creating sophisticated "track doubling" effects. Two or four voices can be created, with independent controls on each for Detune (up to 100 cents), Pan, Delay, EQ, Octave, Feedback, and Volume (see Figure 12). Modulation effects can range over a depth of ± 200 cents, and the LFO (low frequency oscillator) has reset and sync utilities for synchronizing these effects to the beat or to the start of the sound. A monophonic sound can be output as a stereo track, and stereo sounds can have their panning characteristics enhanced.

Morphoder is a vocoder which is capable of utilizing an eight-voice stereo synthesizer as the modulator signal, packaged with ten built-in waveforms optimized for the vocoding process. An on-screen keyboard enables note adjustments to be carried out intuitively (see Figure 13). A noise generator is also provided, along with a filter for controlling the noise band, to mix with the modulator and carrier sources. The aim is to

Figure 14. Screenshot of Multi Tool interface for Waves Trans-X processor.



shape articulation, "edge," and "air" in the output sound. A five-band linear phase EQ utility is provided for refining the Modulator signal, and the graphic display also shows the carrier signal's frequency profile for ease-of-use in shaping the vocoder process. Additional controls include Formant Correction, Release Control, and Pressure Control (for compression of the Modulator).

Trans-X is a utility for shaping signal transients. There are three tools for working with the signal in different ways: Multi (see Figure 14), Wide Band, and 4-Band (for greatest control over all shaping parameters). Attacks can be redesigned to be softer or punchier, and frequency regions can be controlled independently where desired. there are controls for room sound and microphone distance. In addition to transient shaping, Trans-X may also be used to highlight the sustain portion of the signal.

The TDM version of the Transform Bundle is listed for US$ 1,800; the Native version lists for US$ 1,200. Contact: Waves, Ltd., Azrieli Center, The Round Tower, 21st Floor, Tel Aviv 67011, Israel; telephone (+972) 3-608-1648; fax (+972) 3-608-1656; electronic mail info@waves.com; Web www.waves.com/. In North America, contact: Waves, Inc., 306 West Depot Avenue, Suite 100, Knoxville, Tennessee 37917, USA; telephone (+1) 865-546-6115; fax (+1) 865-546-8445.