Products of Interest

Spitfire Audio BBC Symphony Orchestra Sample Library

UK company Spitfire Audio has released a new BBC Symphony Orchestra sample library that runs as a plug-in. The samples were recorded at Maida Vale Studios in London, which has been the home of the orchestra since the 1930s. More than 200 hours of recordings were made, generating more than one million samples. The library uses a lossless compression format that results in a size of 565 GB.

Fifty-five instruments were recorded, across all string, brass, woodwind, and percussion sections of the orchestra. These include soloists and groups, various dynamics, release triggers, recorded legatos, and 418 different playing techniques. There is a legato patch available for each of the main instruments, including the groups. In the case of strings, the legato patches include portamento, slurred transitions, and short notes. The legato patches for the flute and piccolo include slurred transitions, fast run transitions, and short staccato notes.

The user has a choice of eleven different microphone positions: mono, close, tree, ambient, outriggers, mono close, leader, stereo Coles 1438s, close wides, balcony, sides, and Atmos, which is designed for Dolby Atmos mixing. There are also five spill signals (strings, woodwinds, brass, percussion, and full), which can be used to add to the sense of depth and three-dimensional sound. Two Jake Jackson stereo mixes are also included.

The library plug-in is compatible with both Windows and Macintosh and is available in VST2, VST3, AU, AAX, and NKS formats. It can be downloaded as four separate files: strings, brass, woodwinds, and percussion. However, Spitfire Audio recommends buying their external SSD, which is a 1-TB Samsung 860 EVO drive with a hard case. The library is preloaded onto the SSD drive and can be run directly from it. Alternatively, a standard hard drive is also available at a lower cost. A computer with an Intel 2.8-GHz i7 (six-core) processor or similar is recommended for use with the library.

BBC Symphony Orchestra is listed for US$ 999 for the library. The SSD version is an additional US$ 249 and the standard hard drive an additional US$ 99. Educational discounts are available. Contact: Spitfire Audio; Web www.spitfireaudio.com.

Sonnox Oxford Drum Gate Plug-In

Sonnox's Oxford Drum Gate is a dynamics processor for drums in plug-in format. Unlike standard drum gates that apply a blanket gate across the entire signal when a specified threshold is reached, the Oxford Drum Gate first applies a detection algorithm to identify snares, kick drums, toms, and cymbals, so that specific parts of the signal can be kept or gated.

There are three tabs on the plug-in interface: Detection, Decay, and Leveller. The Detection section uses transient detection and a machine learning algorithm to decide which part of the signal is to be kept and which part should be gated. A graphical representation of the signal shows the processing of transients. Detected transients are marked with an arrow and a different color is assigned to those transients that are above and below the threshold value. There is a large slider for adjusting the value of the threshold setting. The user can also adjust a transient sensitivity slider if there are too many or too few markers for transients detected.

When more accurate detection is required, a Match Transients option can be used. The user selects kick drum, snare, or tom, and a machine leaning algorithm then detects those drums in the signal and only lets that component of the audio through the gate.

The third function in the Detection section is Learn Unmatched/Remove Matched. This fine-tunes the detection by allowing the user to manually select detected transients. Clicking on Learn Unmatched allows that transient to pass through the gate, while selecting Remove Matched gates transients that aren't removed by the automatic detection options.

The Decay tab of the interface features controls for the release of the gate. All transients that are allowed to pass through the gate are displayed on a dynamic waterfall graph, with frequency on the horizontal axis and time on the vertical axis. There are sliders provided for adjusting the release time, gain reduction, and shorten decay. There is also a Resonant Decay setting, which allows specific parts of the frequency range to resonate naturally or have processing applied. This setting is adjusted on the visual display by dragging points on a frequency band line.

The third section on the plug-in interface, the Leveller tab, is designed to bring more consistency to drums after gating, while keeping the sound and performance dynamic. It features a large split fader with loud sounds above the fader and soft sounds below. The developer indicates that this may not be needed on all drum hits and types of music but that being able to discern the difference between loud hits and barely audible brushes on snares, for example, ensures a dynamic performance and consistency. There is an automatic

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detection option available or the user can set these loud and soft targets manually.

The transients that are allowed to pass through the gate can be output directly to a MIDI virtual instrument or other MIDI software. The user can control the MIDI note that should be played for each drum hit. Alternatively, for hosts that do not accept MIDI output from an audio channel, there is an option to capture the performance and save it as a MIDI file for future use.

A bypass button is provided for switching between the original and processed sound. A trim control can be used to adjust the output between ±20 dB. There is a Preset Manager, which can be used to save presets from one host application that can subsequently be used with another.

The plug-in supports a range of sample rates from 44.1 kHz to 192 kHz. Latency is 2,889 samples at the lower sample rate and 12,576 at the highest sample rate. Oxford Drum Gate is compatible with Macintosh and Windows, and is available in VST2, VST3, AU, and AAX native formats.

Oxford Drum Gate is listed for approximately US$ 236. Contact: Sonnox, Manor Barns, Finstock, Oxfordshire, OX7 3DG, UK; Web www.sonnox.com.

Emvoice One Vocal Synthesiser Plug-In

The Emvoice One is a vocal synthesizer plug-in. It is available in VST2, VST3, and AU, and AAX formats. The plug-in can be used to simulate a real vocalist for producers, to add backing vocals to a track, or to create vocal tracks with extended techniques and ranges that would not be possible for a human singer. Instead of using synthesis and modeling techniques that run on the user's host computer, Emvoice One uses a Cloud-based sampling approach. The audio processing is carried out on servers, which construct the vocal lines from thousands of pre-recorded samples, to create a realistic vocal sound.

The plug-in itself is free to download and the user must purchase the voices that perform the vocals. The first voice available for the plug-in is Lucy, a female voice with a range of two and a half octaves (E2–A4).

The interface of the plug-in uses a piano roll style grid onto which notes are drawn. Notes that are outside the range of the particular voice being used are grayed out. The user can enter notes manually or use a MIDI Listen function to input them from a MIDI device. Notes are automatically quantized and users can change the resolution from whole notes to sixteenth notes, or they can choose to switch off quantization. Glissandos and vibratos can also be added to notes.

A text box for lyrics appears below each note or phrase that is entered onto the grid. The user can click on this box to enter text. The default is to have one note corresponding to one syllable but multiple notes can be joined into phrases and assigned to a single text box. Words that are used need to be included in the plug-in's dictionary for it to successfully produce sound, so misspellings or unknown words are highlighted in red. The user can select specific pronunciations of a word and create custom pronunciations that can be added to the plug-in's dictionary. Glottal stops can be inserted and specific vowels can be stressed or unstressed. Phrases and words can be copied and pasted, which can assist in building up tracks that use repeated melodies and choruses.

Entire songs can be exported and imported into new projects. The basic building blocks of words used by the plug-in are phonemes. All words and lyrics are converted into their phonemes in the background. It is these phonemes that the audio engine uses to construct the audio track from previously recorded samples.

A demonstration version of the plug-in is available to try out before purchasing.

The Emvoice One plug-in is free to download. The Lucy voice is available for US$ 199. Contact: Emvoice; e-mail contact@emvoiceapp.com; Web www.emvoiceapp.com.

Expressive E Osmose

Osmose is a polyphonic keyboard synthesizer created by Expressive E (see Figure 1). The keyboard has a modular sound design engine, allowing it to be used as a standalone synthesizer or as a controller for external hardware and software synthesizers. It has 49 full-size keys, supports up to 24 voices, and includes layered and split modes.

The Osmose uses the Expressive E's patented Augmented Keyboard Action (AKA) mechanism, which detects polyphonic initial pressure, high quality polyphonic aftertouch, and polyphonic pitch control. This allows the user to add vibrato and pitch bends to notes using a left to right finger movement on the keys. The keys have a physical resistance built into them to give precision in the continuous control for each note, a travel depth that facilitates expressive playing, and high-resolution sensors to ensure accuracy. There are controls for adjusting the sensitivity of the key detections. These features give users extended control over playing the synthesizer, allowing them to create swells, vibratos, and to bend notes at an individual level or polyphonically. There are also two faders for control of pitch and modulation.

Figure 1. The Osmose polyphonic keyboard synthesizer from Expressive E.

The keyboard has a built-in color LCD screen, eight encoders, and nine buttons for controlling settings. The user can configure MIDI equipment directly, without the need for a computer. Osmose supports multichannel MPE and MPE+, and it can send polyphonic aftertouch and global pitch bend messages. Alternatively, the keyboard can be used in piano mode, in which it acts like a simple keyboard controller, sending velocity messages for each note.

The internal sound engine of the Osmose allows it to also be used as a standalone instrument. It features the digital modular synthesizer Eagan Matrix, designed by Haken Audio and used in their innovative Continuum Fingerboard instruments. It is preloaded with hundreds of sounds and textures. The user can also use a software editor, the Continuum Editor, for advanced sound design. A range of synthesis methods and tools are available, including waveshaping oscillators, multimode resonant filters, modal synthesis, sine wave oscillators, Harman filters, kinetic models, and multipurpose delays.

The Osmose has two continuous pedal inputs and these can be assigned to the sustain function or to other synthesizer parameters. A five-pin MIDI input and MIDI output/Thru is included, along with a USB Type B port for MIDI. There are two 1/4-in stereo outputs and a 1/4-in stereo headphone output.

The Osmose is listed for US$ 1,799. Contact: Expressive E, 6 Rue Galilée 93100 Montreuil, France; e-mail contact@expressivee.com; Web www.expressivee.com.

Audient Evo Audio Interfaces

Audient has released two new audio interfaces in their evo range, the evo 4 and evo 8. Both feature a Smartgain mode, which can be used to automatically set the input levels. Users play through the piece they want to record while the interface is in Listening mode and then the most appropriate recording level will be automatically set.

The evo 4 is a two-input, two-output audio interface with two combination microphone/line inputs and a junction gate field effect transistor (JFET) instrument input. Phantom power is available for the microphones and the preamplifiers can provide up to 58 dB of gain. There are two 1/4-in TRS connectors for use with speakers and a headphone output. The interface supports sampling rates up to 96 kHz, at 24-bit resolution. The A-D converters have a dynamic range of 113 dB.

The front panel of the interface has a single large knob at the center, which is surrounded by a ring of LED level indicators. There are buttons on either side of this knob for choosing the function or parameter to be adjusted. These include select buttons for channels 1 and 2, which can be linked to the Smartgain function. Holding down a channel button will mute that channel. There are also Monitor Mix/Monitor Pan and volume buttons for applying the knob control to those functions. The interface supports latency-free monitoring. An on/off button for phantom power and an activate Smartgain button are also provided on the front panel.

The evo 4 interface measures 2.63 × 5.51 × 2.63 in and weighs 0.79 lb. The larger evo 8 model is a four-input, four-output version. It has four combination microphone/line inputs on the rear panel, a single JFET instrument input, two speaker outputs, and two headphone outputs.

The evo interfaces also have a smart muting feature that automatically mutes the monitor outputs if headphones are connected. Audio can be recorded from the user's computer simultaneously with the microphone inputs. A software loopback mixer is included for combining and routing audio. The Evo Control app can be used to set levels, activate features, and control recording from a computer.

The interfaces are bundled with Cubase LE and LE2, Wall of Sound guitar and cabinet presets, the Retologue 2 virtual analog synth, and

2 GB of LoopMasters samples. They are connected to computer used a USB-C connector and are bus powered.

The evo 4 interface is listed for US$ 129 and the evo 8 for US$ 199. Contact: Audient Ltd, Aspect House, Herriard, Hampshire, RG25 2PN, UK; Web www.evo.audio.

Solid State Logic SSL 2 Audio Interfaces

Solid State Logic has released two SSL 2 model USB-powered audio interfaces as part of their new personal studio range. These are two-input, two-output interfaces with classic analog microphone preamplifiers, sampling rates up to 192 kHz, a legacy 4K mode taken from their consoles, and studio-quality monitoring.

The SSL 2 model has two combination Neutrik microphone/line/instrument inputs on the rear panel, along with balanced monitor outputs on $^1/_4$-in TRS ports. The microphone preamplifiers have a two-stage design that use discrete low noise transistors with integrated circuits. They have a gain range of 62 dB and an Equivalent Input Noise (EIN) level of −130.5 dBu. The maximum input level is +15 dBu for the instrument input and +24 dBu for the line level input. The maximum output level for the monitor outputs are +12.5 dBu, with a dynamic range of 112 dB. The connection for the high-current NJM headphone amplifier output is located on the rear panel. It has a maximum output of +10 dBu and offers a dynamic range of 111 dB. There is also a type-C USB 2.0 port on the rear panel, through which the interface is powered. USB C-to-C and C-to-A cables are included. The interface uses AKM A-D and D-A converters and supports sample rates from 44.1 kHz to 192 kHz.

The top panel of the interface has a sloped design, to give the user easy access to all controls and meters. At the top of each channel strip are three small switches, for phantom power, line input, and microphone input. Below that is a five-segment LED meter for the input level, an individual gain knob, and a Legacy 4K switch. This is a new analog enhancement effect that is based on SSL's 4000 series consoles. It combines a high-frequency equalizer boost with a circuit that replicates the subtle harmonic distortion that comes from analog stages such as compressor voltage controlled amplifiers and transistor mix amplifiers. The top panel also has a large knob for level control, a separate knob for the headphone level, a monitor mix knob that blends between the input and USB signals, a stereo switch, and an LED indicator for USB operation.

The SSL 2+ model is a two-input, four output interface. It has an additional MIDI input and output (on five-pin DIN connectors), an extra headphone output, and additional unbalanced outputs. The user can monitor two separate stereo mixes with this interface.

Purchase of the SSL 2 audio interfaces includes SSL Native Vocalstrip 2 and Drumstrip plug-ins, Avid Pro Tools | First, Ableton Live Lite, Native Instruments Hybrid Keys & Komplete Start, and a set of 1.5 GB of samples from Loopcloud. The interfaces are class compliant with macOS 10.11 and above and require ASIO/WDM drivers for use with Windows 8.1 and 10.

The SSL interface is listed for US$ 229.99 and the SSL2+ for US$ 279.99. Contact: Solid State Logic, 25 Spring Hill Road, Begbroke, Oxford OX5 1RU, UK; e-mail sales@solidstatelogic.com; Web www.solidstatelogic.com.

MOTU M2 and M4 Audio Interfaces

The M2 and M4 audio interfaces (see Figure 2) have been designed by Mark of the Unicorn (MOTU) to offer features usually found in high-end interfaces in an affordable, portable device. They use ESS Sabre32 Ultra DAC Technology, have a 120 dB dynamic range, −129 dBu EIN on the microphone inputs, headphone amplifiers driven by the ESS converters, support sample rates up to 192 kHz, and have a full-color high-resolution screen with built-in metering.

The M2 model has two combination XLR/TRS inputs on the front panel for microphone, line, and Hi-Z guitar inputs (see Figure 2). Phantom power, gain control, and a monitor switch are provided for each input. Direct monitoring is also available for the inputs. There are two balanced $^1/_4$-in TRS jack outputs on the rear panel, along with two RCA mirror outputs, five-pin 16-channel MIDI input/output, and a USB-C port, which is compatible with USB type-A. The $^1/_4$-in headphone output with dedicated volume control is built into the front panel. There is also a large control knob here for monitoring volume and a 160 × 120-pixel color LCD for the input and output level meters. The interface is powered through the USB connection. The M2 model measures 7.5 × 4.25 × 1.75 in and weighs 1.35 lb.

The larger M4 interface has two additional balanced TRS line inputs and outputs and two additional RCA mirror outputs on the rear panel. The front panel has a knob for balancing the mix between the live inputs and playback from computer. This interface measures 8.25 × 4.25 × 1.75 in and weighs 1.55 lb.

The M2 is listed for US$ 169.95 and the M4 for US$ 219.95. Contact: MOTU, 1280 Massachusetts Ave,

Figure 2. MOTU's M2 and M4 USB audio interfaces.

Figure 3. The Komplete Audio 6 USB audio interface from Native Instruments.

Figure 2

Figure 3

Cambridge, Massachusetts 02138, USA; Web www.motu.com.

Native Instruments Komplete Audio 6 Mk2 USB Audio Interface

Native Instruments has created an updated version of their Komplete Audio 6 interface, which was first released in 2011. This Mk2 version (see Figure 3) of the USB audio interface supports sample rates up to 192 kHz and 24-bit depth. It is a six-input/six-output interface with analog, digital, and MIDI connections. The front panel of the Audio 6 features two combination $^1/_4$-in TRS line/XLR inputs with optional 48-V phantom power. There is an individual switch for line/instrument source on each of these inputs and a separate gain control knob. There are an additional two $^1/_4$-in TRS line inputs on the rear panel, along with SPDIF digital input/output, five-pin DIN MIDI input/output, and four DC-coupled $^1/_4$-in TRS outputs. Two 25-mW headphone outputs with individual volume controls are provided on the front of the interface for easy access. The frequency response on analog inputs and outputs is 20 Hz to 20 kHz ($\pm$0.1 dB).

Figure 4. Allen & Heath's Avantis digital mixer.

The maximum output level is 11.5 dBu.

The top panel of the interface features a large volume control knob, along with four LED VU meters for the analog inputs, one VU meter for the output, and status indicator LEDs for phantom power, USB, and MIDI. Direct monitoring is possible during recording and there is a switch for moving between channels 1/2 and 3/4 on the front panel.

The interface is powered by USB and is compatible with Macintosh and Windows through ASIO, Core Audio, DirectSound, and WASAPI. It measures 55.6 × 200 × 136.55 mm and weighs 850 g.

A range of software applications is bundled with purchase of the interface. These include Monark, an analog synth; Mod Pack, a set of phaser, chorus, and flanger effects; Replika, a delay with modulation, filters, and phasing; Solid Bus Comp, a compressor; Komplete Start, a collection of synthesizers, instruments, and effects; and music production software Maschine Essentials, Ableton Live 10 Lite, and Traktor LE 3.

The Audio 6 interface is listed for US$ 249. Contact: Native Instruments, 6725 Sunset Boulevard 5th Floor, Los Angeles, California 90028, USA; Web www.native-instruments.com.

Allen & Heath Avantis Digital Mixer

Allen & Heath's Avantis is a digital mixer that features the XCVI FPGA technology also used in their SQ and dLive consoles (see Figure 4). It offers user 64 inputs, 12 stereo FX returns, 42 flexible mix buses, parallel virtual processing cores, a 96-bit bus, and ultra-low latency of 0.7 msec. The mixer has two full HD touchscreens, processing taken from their dLive mix system, and a range of expandable input/output options.

The Avantis mixer has 12 analog XLR inputs and outputs, a stereo AES

Figure 5. The Zylia ZR-1 19-channel recorder with the ZM-1 microphone array.

input and two stereo AES outputs, and BNC word clock input and output. It also has dedicated SLink ports for use with the company's range of GX and DX remote input/output expanders, and an SLink port to connect to another Avantis, SQ, or dLive mixer. Two additional card slot ports are also built into the mixer for use with Dante, Waves, MADI, or gigaACE cards, giving 128 × 128 operation at 96 kHz. A USB 2.0 port is also included and can be used to store and record files, for stereo playback and recording, and for updating the firmware. Full-size and mini headphone outputs are provided, with dedicated volume controls.

The mixer has 24 fader strips organized into 6 layers, giving 144 in total. Each fader strip has an illuminated touch-sensitive fader, along with backlit pre-fade listen, mix, and mute buttons, and a rotary encoder. Each of the fader strips can be configured with an input channel, a mix, an effects send, a Digitally Controlled Amplifier, or a MIDI strip.

The Continuity UI interface displays the fader strips on two 15.6-in Full HD capacitive touchscreens, giving a total of 206 sq in of touchscreen. The touchscreens support gestures such as swipe, drag and drop, and pinch. There are also 24 assignable SoftKeys and 6 assignable rotary controls. Parameters can be copied, pasted, and reset. There are 500 scene memories available, along with Channel Safes, Global, and per Scene Recall Filters.

An Automatic Mix Mixer function is available on up to 64 channels. This can be used to reduce background noise and minimize feedback in situations where a number of microphones are used. Each channel can be assigned a priority level to determine its weighting in the overall gain.

Full processing is available on the input channels, including low-pass filter, high-pass filter, gate, parametric equalizer, compressor, and delay. There are twelve RackExtra FX slots with dedicated stereo returns and twelve Stereo FX with dedicated FX Returns. In additional, the user can choose to upgrade to a dPack option, which adds processing from Allen & Heath's dLive system, including 16 instances of their Dyn8 engines, zero-latency DEEP Compressors, and Dual-Stage Valve preamplifiers on every channel.

The Avantis has a full metal chassis. It weighs 57 lb and measures 10.6 × 36.1 × 24.7 in.

The Avantis is listed for US$ 9,999 and for US$ 11,499 with the dLive effects. Contact: Allen & Heath Limited, Kernick Industrial Estate, Penryn, Cornwall, TR10 9LU, UK; Web www.allen-heath.com.

ZYLIA ZR-1 Portable Multi-Channel Recorder

Zylia's 19-capsule ZM-1 microphone array was introduced to readers in Products of Interest of *Computer Music Journal* 42(1) in 2018. At the time of its release, the microphone required a software application, Zylia Studio, for recording and processing. There was also a separate standalone or plug-in Ambisonics converter available.

Now, Zylia has created the ZR-1, a portable hardware recorder for the ZM-1 microphone that makes location and field recording easier for users (see Figure 5). The recorder supports 19-channel recording at a sample rate of 48 kHz and 24-bit depth. It connects to the microphone using a USB cable.

Figure 6. Nux's B-3 wireless microphone system.

The recorder uses an SD memory card to store audio, which can be then transferred to computer. SDXC cards with a capacity of up to 2 TB are supported, allowing the user to record more than 220 hours of audio in Wave64 file format or approximately 300 hours if using the WavePack file format.

All connections and controls are located on the front panel of the recorder for easy access, and a shoulder strap can be attached for portable use. The available ports are: micro USB port for power, USB port for connection to computer, and a stereo mini jack headphone output for live monitoring and playback. The recorder also has an SD card slot. There is an on/off button, transport buttons, gain knob, volume knob for playback, and a button for pairing the recorder with a Bluetooth phone for remote control using an app.

The recorder can be powered by a 5-V external adapter, over microUSB, or by eight AA batteries. Up to 4 hours of operating time is available when using 2500 mAh NiMH batteries. The recorder measures 8.11 × 7.72 × 2.28 in and weighs 2.69 lb.

Zylia has also produced a development kit to give professional users full freedom to experiment with spatial arrangements. The 6DoF VR/AR Development Kit consist of nine Zylia ZM-1 microphone arrays, the 6DoF recording application for use with multiple microphones, and the 6DoF control panel, which offers gain control and LED control for all connected microphones. This purchase also includes ZYLIA Studio, ZYLIA Studio PRO, and ZYLIA Ambisonics Converter.

The ZR-1 recorder is listed for approximately US$ 1,110, and a bundle with the recorder, microphone, and software applications is listed for approximately US$ 2,222. Prices for the DoF VR/AR Development Kit are available on request. Contact: Zylia, Uniwersytetu Poznanskiego 2, 61-614 Poznan, Poland; e-mail record@zylia.pl; Web www.zylia.pl.

NUX B-3 Wireless Microphone System

The Nux B-3 system is used to turn a standard wired microphone into a wireless microphone. It consists of a wireless transmitter and receiver (see Figure 6). The transmitter has a female XLR port for connecting it directly to the body of a microphone. The receiver has a male XLR port for connection to amplifiers, speakers, mixers, and mixer. A female XLR to 3.5-mm jack cable is also included for connecting the receiver to cameras or mobile phones.

The B-3 system operates using the 2.4-GHz wireless band. It has a distance range of 100 ft, latency of less than 4 msec, and supports sample rate up to 48 kHz at 32-bit depth. The units are powered by a rechargeable battery, which gives up to 5 hours operating time.

Both the transmitter and receiver have six channel indicator buttons, a channel switch button, a RF status LED, an on/mute switch on the side, and a micro USB port for charging on the other side. The devices are paired on channel 1 by default but this can be changed to any of the other five channels by pressing the same channel number button on both the transmitter and receiver.

When the devices are initially turned on, the channel number buttons doubles as a battery gauge, lighting up for 1 second to indicate the level of battery. If only the lower button lights up, for example, this indicates that the battery level is at approximately 15 percent but if all six are lit up, then the battery level is at 100 percent. If there is no input signal for 30 sec, or if the transmitter is not connected to the receiver, both devices will enter sleep mode. They are immediately activated again if a signal is transmitted. If there is no signal for an hour, they will enter a deep sleep mode and, in that case, the user will need to press the channel button or use the on/mute switch to wake them up.

The dynamic range of the system is 108 dB and the total harmonic distortion with noise is reported as less than 0.02 percent (at 1 kHz, −10 dBFS). The B-3 comes with a split USB charging cable that connects to both transmitter and receiver, a hot shoe microphone adapter for use with a camera, and a female XLR to 3.5-mm jack cable. The transmitter and receiver are the same size, measuring 107 × 25 × 26 mm and weighing 125 g.

An additional accessory is available for using the B-3 system with a mobile phone. The B-3MA package includes a phone holder with cold shoe and a 3.5mm TRS jack to TRSS adaptor cable to connect the receiver to the phone.

The B-3 is listed for US$ 179. Contact: Cherub Technology, Rm 507, Block 1, Nanhai E-Cool, 6 Xinghua Rd, Shekou, Nanshan District, Shenzhen, Guangdong 518067, China or Cherub Technology, 11940 Goldring Rd Ste C, Arcadia, California 91006-6013, USA; e-mail

Figure 7. Xvive Audio's U3C wireless microphone system.

info@cherubtechnology.com; Web www.nuxefx.com.

Xvive Audio U3C Wireless Microphone System

Xvive Audio produces a range of wireless microphone and guitar systems. Their latest is the U3C, which turns a condenser microphone into a wireless microphone. It consists of a transmitter and receiver unit, which are made from molded plastic and cast metal (see Figure 7). The transmitter weighs 108 g, the receiver 92 g, and both are 31 × 29 × 98 mm in size. The transmitter has a female XLR port for connecting to a balanced condenser microphone. The receiver has a male XLR port for connection to the microphone input on a mixer, speaker, or recorder. The system uses the 2.4-GHz wireless band for communication between the two and it has a working range of 90 ft.

As it is designed for use with condenser microphones, the transmitter can provide phantom power to the microphone. A switch on the unit body allows the user to switch between 12-V and 48-V phantom power. The user can choose to broadcast on six channels, the default setting being channel 1. A circular LED display on the front of each unit shows which channel is being used and the channel switch button is located in the center. The receiver has an LED indicator showing the status of the connection with the transmitter. The frequency response of the U3C is 20 Hz to 20 kHz (−3 dB) and it has a dynamic range of 110 dB. There is a latency delay of 5 msec for transmitted sound.

The transmitter and receiver are powered by built-in Li-ion rechargeable batteries, which give an operating time of up to 5 hours for the receiver. The battery life of the transmitter depends on how much phantom power is being used by the microphone and ranges from 3 hours for 48-V phantom power at 5 mA, to 7 hours for 12-V phantom power. There is a micro USB port on the side for recharging and a split USB cable is included for charging both units simultaneously. An LED indicator for the battery level remains off if the power level is between 30 and 100 percent. It illuminates a constant red if the level falls below 30 percent and changes to a flashing red if the level falls to 10 percent or less. Both units have a power on/off switch.

Xvive also produce an older model, the U3, which is for use with dynamic microphones. It has similar specifications to the U3C and can deliver a battery life of 5 hours.

The U3C is listed for US$ 219 and the U3 for US$ 199. Contact: Xvive Audio, 1040 E Woodbury Road, Pasadena, California 91104, US; Web www.xviveaudio.com.

Soyuz Microphones Launcher

Soyuz Microphones produces a range of high-end handmade microphones. Their latest product, the Launcher, is an in-line active microphone preamplifier that is designed to add the color of vintage consoles to recordings made with less expensive dynamic microphones and audio interfaces (see Figure 8).

Figure 8. Soyuz Microphones' Launcher in-line active microphone preamplifier.

The unit measures 4.4 × 2.1 × 1.9 in and weighs 0.9 lb. It has a three-pin locking female XLR connector on one end and a similar male connector on the other. It has an all-metal chassis and four rubberized feet on the bottom.

Launcher can be used with dynamic and ribbon microphones, which do not require phantom power. It features a custom transformer and circuit. It can boost the gain by 26 dB and it has a frequency range of 10 Hz to 20 kHz (−1 dB). The maximum output level is 8.3 dBV. Launcher requires 48-V phantom power for operation. It uses 5.0 mA and has an electrical impedance of 1k Ohms.

Launcher is listed for US$ 199. Contact: Soyuz Microphones, 3355 Wilshire Blvd. suite 616 Los Angeles, California 90010, USA; e-mail soyuzmicrophones@gmail.com; Web www.soyuzmicrophones.com.

Cloud Microphones Cloudlifters

The Cloudlifter CL-1, from Cloud Microphones, is an in-line low-noise microphone preamplifier. It is designed to provide active clean gain

Figure 9. The Ambisonics Spatial Mic from Voyage Audio.

to dynamic and ribbon microphones. It can also be used when recording quiet sound sources or when using noisy or low-gain preamplifiers.

The device measures 2 × 4.5 × 2 in and weighs 0.85 lb. It has a metal chassis with an XLR male connector at one end and a female connector at the other. It has four rubberized feet to prevent sliding during use. It uses patented discrete JFET circuits that preserve the natural sound of the microphone. The preamplifier requires 48-V phantom power and it can deliver up to +25 dB of gain. The input impedance is optimized for ribbon microphones at approximately 3k Ohms.

Cloud Microphones also produce a two-channel model, the CL-2, and a rack mountable four-channel version, the CL-4.

The CL-1 is listed for US$149, the CL-2 for US$ 249, and the CL-4 for US$ 499. Contact: Cloud Microphones; e-mail info@cloudmicrophones.com; Web www.cloudmicrophones.com.

Voyage Audio Spatial Mic

The Spatial Mic is a second-order Ambisonics microphone from Voyage Audio. It uses an eight-capsule array to record 360-degree sound (see Figure 9). The microphone outputs all eight channels on a single cable, with a choice of USB 2.0 on a Type-C connector for streaming to a mobile device or computer, and ADAT lightpipe for use with an audio interface. Both the USB and ADAT outputs can be used simultaneously. The microphone can be powered through the USB port or by using an external 5-V power source with a micro USB port. It also has a mini headphone jack for live binaural monitoring. All outputs are located on the bottom of the microphone.

The microphone array is made up of eight matched 14-mm prepolarized condenser capsules with digitally controlled analog front ends. The signal-to-noise ratio is greater than 72 dB (A-weighted). The maximum sound pressure level is 120 dB (at 1 kHz), with total harmonic distortion of less than 3 percent. The microphone supports sample rates of 44.1 and 48 kHz, at 16- or 24-bit resolution. The digital circuit board within the microphone includes a 16-core processor and uses high-resolution SiLabs clocking. A unique calibration profile for each specific microphone is stored on-board.

The Spatial Mic is made from aluminum and nylon with a metal head basket. It measures 6.75 × 2.125 in and weighs 0.54 lb. It has a circle of LED indicators with an anodized aluminum control knob at the center. Pushing the knob cycles it through its control modes: metering, microphone gain, mix, and headphone level. Once selected, the user then rotates the knob to adjust the value or level of that parameter. The microphone capsules can be muted/unmuted by pushing and holding the control knob. The LED indicator ring can be used to show the signal levels of individual capsule signals, the live binaural monitor mix, or it can be turned off.

Two software downloads come with the microphone. The Spatial Mic Converter plug-in uses an internal 64-channel filter matrix to convert the raw audio signal to first- or second-order Ambisonics in AmbiX or Fuma formats. It also allows the user to change the direction of the microphone so that it is aimed at a particular sound source or aligned with audio positioning in a surround video. A Spatial Mic Control app can be used to change hardware parameters of the microphone. It transmits information through the microphone's USB port.

The microphone has a built-in $^1/_4$-in–20 thread mount. The buyer has a choice of a 3-m black nylon braided USB C-to-C or C-to-A cables. A foam windshield is also included and a Rycote shock mount is available as an optional extra.

The Spatial Mic is listed for US$ 899. Contact: Voyage Audio; e-mail info@voyage.audio; Web www.voyage.audio.

Roswell Pro Audio Delphos II Microphone

Roswell Pro Audio's Delphos II is a large diaphragm condenser microphone designed to have a natural frequency response, high sensitivity,

Figure 10. Roswell Pro Audio's Delphos II large diaphragm condenser microphone.

and low noise (see Figure 10). It offers the user a choice of three polar patterns: cardioid, omnidirectional, and figure-of-eight. The cardioid pattern eliminates the capsule's rear diaphragm from the circuit, which increases the sensitivity and signal-to-noise ratio.

The Delphos II features a 34-mm capsule with gold-plated, ultra-thin Mylar diaphragms. It has an optimized transformerless circuit to ensure a clean signal path. Its circuit boards use low-noise NOS transistors and high-precision metal film resistors. The JFET has low input capacitance and is manually biased in production to ensure a high signal level without distortion.

The frequency range of the microphone is 20 Hz to 20 kHz. It has a signal-to-noise ratio of more than 82 dBA, equivalent noise of less than 12 dBA, sensitivity of 35 mV/Pa, and impedance of 200 Ohms. The body of the microphone has switches for controlling the pattern on one side and a 10 dB pad on the other. It measures 208 × 60 mm and weighs 850 g, and has an enamel finish in a dark blue metallic color. Included with purchase of the microphone is a Cutaway shock mount, a microfleece sock, and a flight case.

The Delphos II is listed for US$ 999. Contact: Roswell Pro Audio, 125 S. Main St #118 Sebastopol, California 95472, USA; e-mail info@roswellproaudio.com; Web www.roswellproaudio.com.

Aston Microphones Stealth

Figure 11. Aston Microphones' four-voice Stealth microphone.

Aston Microphones' Stealth is marketed as a broadcast quality microphone for use in the studio and on stage. It can be used as an active or passive microphone and has four separate voice settings to choose from (see Figure 11). Two of the voices are optimized for vocals, one for guitar, and the final voice delivers a vintage style voice similar to a classic ribbon microphone. The voice settings are not created from equalization filters but contour networks, in which the entire signal is attenuated and then specific frequencies added back to the signal at a higher level, reducing the phase distortion that can be created by filters.

The microphone can work with or without phantom power. When used as a passive microphone, the active circuits are bypassed and the microphone uses a simple signal chain similar to a dynamic microphone, producing a clean sound. A built-in autodetect function can detect 48-V phantom power. This automatically changes the microphone to operate in active mode, in which it uses the built-in class-C preamplifiers. This gives it an additional 50 dB of gain. Purple LED lights near the base of the microphone indicate the activation of phantom power but these can be turned off, if needed. A moveable ring just above these lights serves as a rotary switch to move between the different microphone voices.

The polar pattern is a very focused cardioid pattern. The capsule is enclosed in a stainless-steel Faraday cage to protect it from external radio frequency interference. The microphone uses a moving coil transducer. It has a frequency response of 20 Hz to 20 kHz (±2 dB). It has an equivalent self-noise of 10 dBA. In passive mode it has a sensitivity of 1 mV/Pa (at 1 kHz into 1 Ohm). This increases to 150 mV/Pa in active mode. The maximum sound press level is 140 dB with total harmonic distortion of 0.5 percent.

The Stealth microphone features an internal shock mount system constructed from synthetic viscoelastic urethane polymer to dampen vibrations and act as an acoustic shield. A quick release stand mount is included with purchase. The microphone measures 7.72 × 2.28 in and weighs 1.52 lb.

The Stealth microphone is listed for US$ 399. Contact: Aston Microphones, 3 Hunting Gate, Hitchin SG4 0TJ, UK; e-mail sales@astonmics.com; Web www.astonmics.com.

New Releases

Publications

Adam Patrick Bell: *The Music Technology Cookbook: Ready-Made Recipes for the Classroom* (hardcover, 2020, ISBN: 9780197523889, softcover, 2020, ISBN: 9780197523896, New York, NY: Oxford University Press, www.oup.com/).

Michael Clarke, Frédéric Dufeu, and Peter Manning: *Inside Computer Music* (hardcover, 2020, ISBN: 9780190659646, softcover, 2020, ISBN: 9780190659653, New York, NY: Oxford University Press, www.oup.com/).

David Miles Huber: *The MIDI Manual: A Practical Guide to MIDI within Modern Music Production* (softcover, 2020, ISBN: 9780367549985, New York City: Routledge, www.routledge.com/).

Mark Reybrouck: *Musical Sense-Making: Enaction, Experience, and Computation* (hardcover, 2020, ISBN: 9780367222406, New York City: Routledge, www.routledge.com/).

Recordings

David Felder: *Les Quatre Temmps Cardinaux* (CD, 2020, Coviello Contemporary 91916, www.coviellomusic.com/).

Jeff Morris: *Hearing Voices: Human Sounds, Digital Ears* (CD, 2020, Ravello rr8033, www.ravellorecords.com/).

Fabio Selvafiorita: *The Fall* (CD, 2020, Stellage003, www.stellage.store/).