PreSonus® NEW PRODUCTS WINTER 2013

PAGE 12

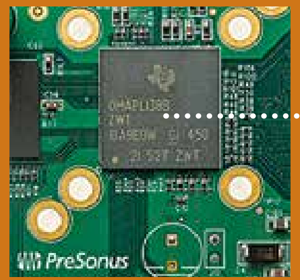
STUDIOLIVE™ Active INTEGRATION

Active Integration.

The powerful product ecosystem that changes how you work and create.

What's the opposite of a bunch of products that don't even know that each other exist?

An ecosystem where every part of your signal chain communicates wirelessly. Seamlessly integrated software with shared interfaces and common signal-processing algorithms. And vastly more processing power than anything that's come before.



To achieve that requires a real computer.

So Active Integration is based on a cutting-edge OMAP™ 5 multicore, 2 GHz Cortex-A computer-on-a-chip. With a 32-bit, 96 kHz multistream CPU, Wi-Fi, 100 Mb Ethernet, USB 2.0, and lavish amounts of 532 MHz LPDDR2 SD RAM, it's the perfect host for PreSonus' current and future hardware and software.

Bottom line: Active Integration is a growing ecosystem that delivers a highly integrated and flexible workflow on stage and in the studio.

NEW StudioLive™ 32.4.2AI. 32 re al channels, a wealth of new features, Wi-Fi LAN adapter included, and the most processing power of any compact digital mixer.

32+1
number of XMAX™ Class A preamplifiers

32
number of accessible channels without have to resort to bank switching

4
number of subgroup buses

14
number of aux (monitor) mixes

48 x 32
input and output streams of built-in FireWire S800 interface

16
number of 31-band graphic EQs pre-inserted on each aux mix and main bus

2 GHz
speed of multicore OMAP 5 CPU

10,000
times more onboard RAM than StudioLive 24.4.2 (Honest! We're not kidding!)

We heard it for years: *"When are you coming out with a 32-channel mixer?"* We could have just stuck eight more channels onto our StudioLive 24.4.2. But we wanted to give you much more than that. More signal processing. More effects. Wireless connectivity. New features that our customers have asked for.

StudioLive 32.4.2AI is a 32-channel, next-generation digital mixer featuring 32 XMAX™ mic preamps, 32 channel line inputs, 4 internal FX buses with reverb and delay, mute groups, Quick Scenes, a 48x34 FireWire S800 audio interface, and much, much more.

The new StudioLive 32.4.2AI can be wirelessly remotely controlled from a laptop, iPad, or iPhone/iPod touch using the included USB WiFi module and a third-party wireless router.

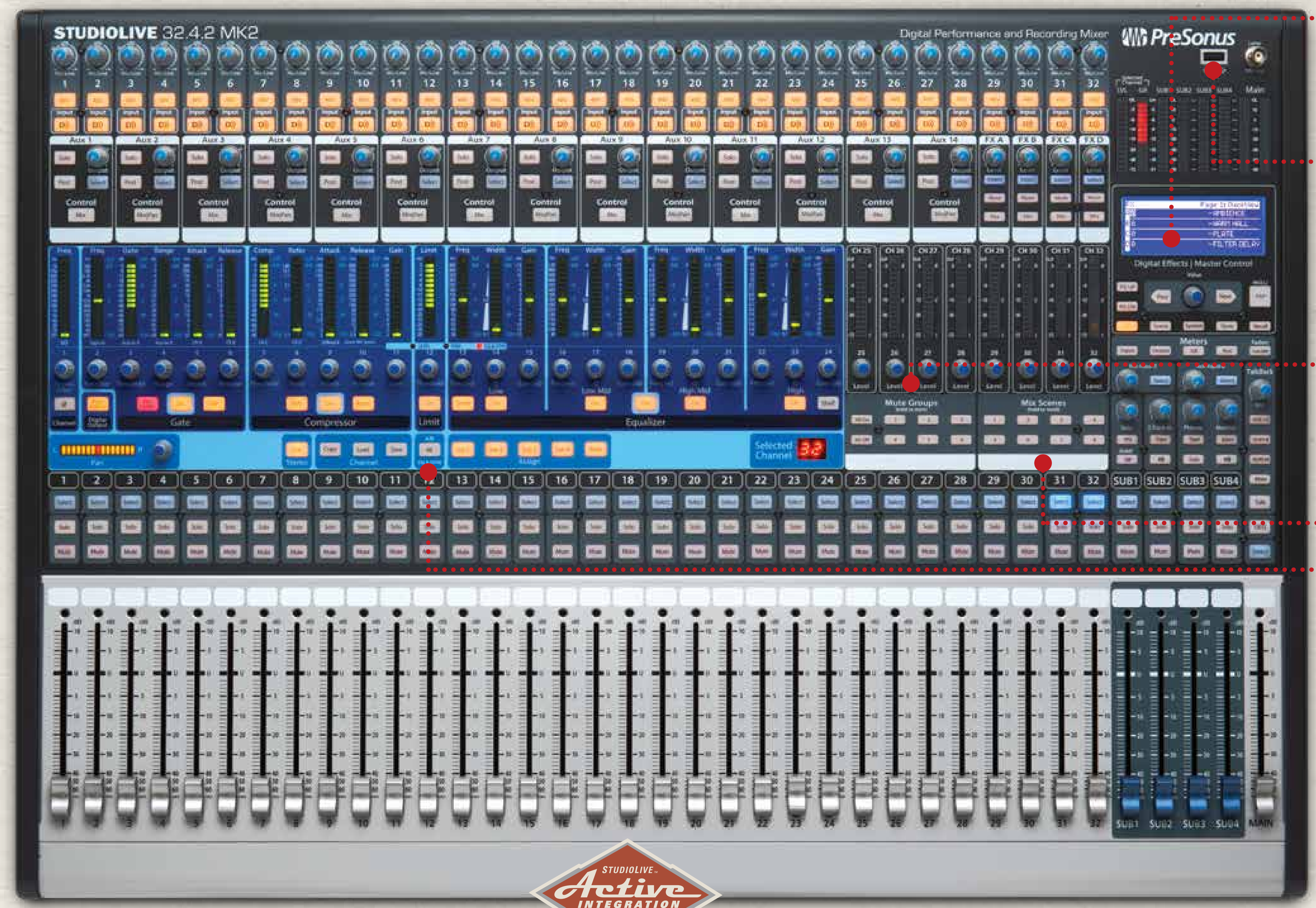
Naturally, it ships with a powerful suite of productivity software: VSL-AI, SL Remote-AI, QMix™-AI, Capture™ 2.0, and Studio One® Artist 2.5 DAW with built-in upload to a free Nimbit® account so that you can share and sell your performances.

A real multicore CPU on board.

Other digital mixers have wimpy little "processor" chips. The new 32.4.2AI has a full-on OMAP™ 5 multicore, 2 GHz, Cortex-A CPU that lets the 32.4.2AI speed your workflow and improve your sound in ways other compact digital mixers can only dream of.

Control 14 different monitor mixes.

QMix-AI handles it without breaking a digital sweat. Both of our newest competitors just dream of doing this. Neener, neener.



Four 32-bit DSP effects engines

…with the extended power of the Active Integration CPU: two processors loaded with rich, detailed reverbs and two armed with precise delay effects.

Six mute groups

…with All On and All Off functions. Mute groups allow multiple channels to be muted and unmuted with a single button. With six mute groups, a FOH engineer can assign drum channels to one mute group, the horn section to another mute group, backup vocalists to a third, etc. So when the horn section takes a break, you can mute the entire section and kill all those open mics instantly. Or when the lead vocalist and guitarist come out for their Page/Plant acoustic jam, you can mute the entire drum kit at once.

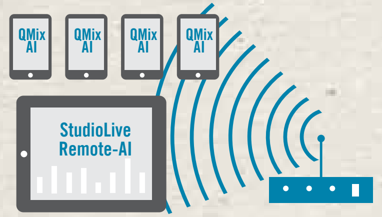
New, obese, double Fat Channel with A/B compare

Dual memories on 54 high-pass filters; 60 gates; 60 compressors; 60 4-band, fully parametric EQs; and 60 limiters give you two different sets of settings at any time (think dark vs. bright overheads, for example)—and the ability to instantly compare them. In fact, it adds up to 468 different processors at your fingertips. That's more than double the processing power of the competition, including Mixer X.

468 PROCESSORS

No computer needed.

Dedicated USB 2.0 port and included Wi-Fi LAN adapter for wireless communications with SL Remote-AI for iPad and QMix™-AI for iPhone.



Six Quick Scene buttons.

Mixing four bands in one night and need to quickly store each scene? The Quick Scene buttons are at your fingertips. These buttons allow you to store six different scenes without having to add them to the library and name them. Now you can get set up for total recall even faster.

Smaart® Room Analysis,

System Delay and Output Check wizards on every bus plus Spectrograph and RTA on every channel and bus.

Smaart®

Browsible Fat Channel presets

are organized into 9 categories (DRM – *Drum*, BAS – *Bass*, GTR – *Guitar*, KEY – *Keyboards*, HRN – *Horns*, PRC – *Percussion*, VOX – *Vocal*, SYS – *System*, MIS – Misc.), so you don't have to scroll through 99 presets to find your favorite bongo preset.

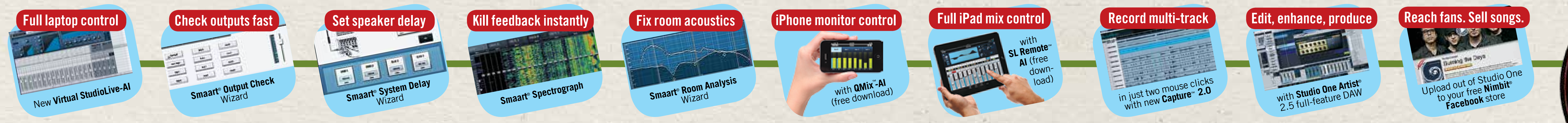
Thunderbolt and Dante options.

StudioLive 32.4.2AI's option card slot ships with dual FireWire S800 ports but by the end of 2013, you'll be able to upgrade your 32.4.2AI mixer to Thunderbolt for ultra-low latency computer connectivity or to Dante for easy networking.

Channel Info Page with…

naming synchronization. New to StudioLive 32.4.2AI is the Channel Info Page. From this screen, you can view the channel or bus number and create a custom name for each channel or bus. These names are stored on StudioLive 32.4.2AI and are broadcast to the entire network, so VSL-AI, SL Remote-AI, and QMix-AI will display names in the channel strips. This works the other way as well: Labeling a scribble strip in software will also broadcast it to the network, including StudioLive 32.4.2AI.

See page 28 for more details about the new StudioLive 32.4.2AI mixer.



Seamlessly integrated AI software suite can optimize your PA system to the venue, then take you from Riff to Release™.

When you compare 32-channel digital mixers, ask some hard questions about the software that is — or is *not* — included. It makes a huge difference.

Does the mixer come with a 1-click multi-channel recording program? Can you "shoot the room" with a professional measurement and analysis program and then optimize the performance of your PA system?

Can onstage musicians control their own mixes with iPhones or iPod touches? Does the iPad program that comes with the mixer do everything you want it to?

Is a real DAW program, with unlimited tracks and loads of plug-ins, included?

Only one 32-channel digital mixer can give you all this and more: StudioLive 32.4.2AI.

XMAX preamplifier

33 XMAX™ mic preamps. Check out the forums. Users think our XMAX preamps sound better than the ones on Mixer *X* or any other compact digital mixer for that matter. Not surprising. XMAX runs on high-voltage (30V) power rails that deliver more headroom, deeper lows, smoother highs, and a richer overall sound. XMAX uses only discrete components — no op-amps— for ultra-low noise and transparency. Finally, XMAX is all Class A circuitry, with no crossover distortion, and delivers purer, clearer, results than the Class AB designs found in many preamps. The net result of the XMAX preamp design is high headroom, low noise, wide dynamic range, extended frequency response, and—most important—musicality and transparency, with smooth highs; solid, deep lows; and everything in between.

Talkback mic preamp with 48V phantom power and independent level control.

Inserts on every channel.

FIRE WIRE 800

48-in/34-out FireWire S800 digital recording interface (24-bit, 44.1 kHz and 48 kHz) makes beautiful music with new Capture™ 2.0 live-recording software. See page 10.

Thunderbolt and Dante options. StudioLive 32.4.2AI's option-card slot ships with dual FireWire S800 ports but soon you'll be able to upgrade to Thunderbolt for ultra-low-latency computer connectivity, or to Dante for easy Ethernet networking.

JetPLL Sync. Synchronization to your computer is stable and robust, thanks to JetPLL™ synchronization technology. JetPLL incorporates noise shaping to virtually remove all audio-band jitter,

JetPLL™

ensuring near-perfect clock performance and optimal converter performance when networking two or more digital-audio devices. JetPLL quickly locks to any digital format through a wide range of frequencies and is extremely robust and tolerant of wide variations in clock frequencies. As a result, you'll experience better stereo separation and clearer, more transparent audio.

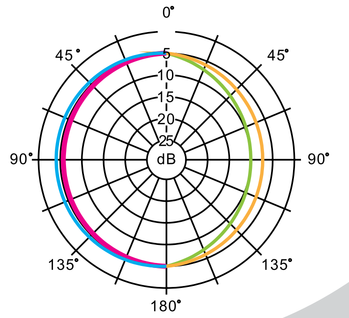
PRM1 Measurement Microphone

With our new PRM1 measurement microphone and free Virtual StudioLive / Virtual StudioLive-AI (VSL) mixer-control software with Smaart® Measurement Technology, all StudioLive digital-mixer owners can have a complete, powerful, and affordable audio-analysis and correction system.

A measurement, or "reference," microphone is special type of condenser microphone that is designed to provide an accurate reproduction of a room's sound characteristics for use with audio-analysis tools, such as real-time analyzers (RTAs) and spectrographs. While measurement microphones can be quite expensive, the affordable PreSonus PRM1 delivers high-quality results.

The PRM1 employs a ¼" pre-polarized electret-condenser capsule and delivers a linear frequency response between 20 Hz and 20 kHz. Like most measurement mics, it has an omnidirectional polar pattern. Sensitivity is rated at -37 dB/Pa, EIN (A-weighted) at 26 dB, S/N ratio at 70 dB, dynamic range at 106 dB, and maximum SPL at 132 dB SPL.

The mic has an all-metal chassis and comes with a clip, foam windscreen, and hard case.



Capture 2.0 has a completely redesigned user interface with a new Start page, main Recording page with switchable marker list, Big Meter view, and dedicated Soundcheck mode. It has an overall darker color scheme for high visibility and legibility in difficult lighting conditions.

NEW Capture™ 2.0.

Capture version 1 introduced the world to the first purpose-built live-performance recording software. The revolutionary, 2-click ease-of-use made live-performance recording accessible to anyone using a StudioLive mixer.

By continuing to innovate, incorporating feedback from our ever-growing user base, Capture 2.0 takes live-performance recording to a new level.

It represents a major leap forward — further streamlining the StudioLive workflow — and further distancing us from our digital-mixer competition, who only offer the compromises of conventional DAWs (and make their users pay extra at that!).

Cool new Start page gives you instant access to new, previous, and recent sessions.

The world's best live-recording software is now even better. And it's free only with StudioLive AI digital mixers.

Enhanced Reliability and Stability

New pre-record buffer. Roll back a recording up to a minute before you actually started recording (no more missing the first few seconds when the band starts early).

Session lock. Inadvertently stopping the recording by leaning on the keyboard is no longer possible.

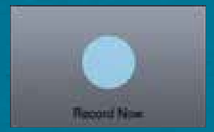
Diagnostics. Remaining-time and disk-performance displays tell you exactly how much recording space you have left and how well your hard drive is performing.

Auto save. No lost work with user-definable Auto-Save intervals. Capture also auto-stores the active session whenever the transport is stopped.

File recovery. Even after a complete power loss, Capture will automatically recover the session and all files the next time it is opened.

96 kHz recording with StudioLive 32.4.2AI

Vastly Improved Workflow



"Record Now" one-step recording. Single-click on the Start page to instantly create a new session and immediately start recording from every input.

Rearrange tracks. Drag-and-drop tracks to reorder the arrangement. This makes importing backing tracks and getting them on the correct channels of your StudioLive mixer quick and easy.

Stereo Playback mode. Capture 2 can be used in Stereo mode with built-in computer soundcards. All tracks are routed to a single stereo output. Want to review and edit that killer show the band just played on your laptop en route to the next gig? No problem.

Virtual-soundcheck tools. Dedicated workflow makes virtual soundcheck incredibly simple by keeping related sessions organized (by artist name, location, etc.) and instantly recallable into a new session. Choose the session to use for your soundcheck, press Play, and do the soundcheck (without the band present). Then click again to exit Soundcheck mode and start recording the new session.

Improved session/file handling. File names and folder structure are organized by artist, performance, location, and song (marker) so you spend less time manually keeping tabs on your files.

Big Meter mode. Turns the computer monitor into a gigantic meter bridge for easily intelligible input metering from any distance.

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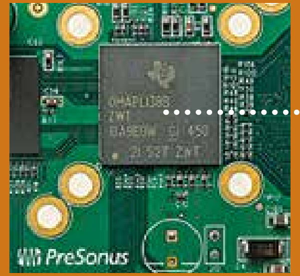

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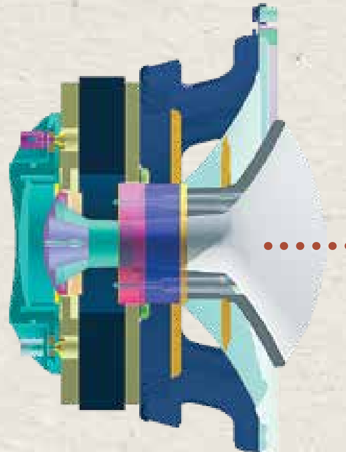
Bottom line: Active Integration is a growing ecosystem that delivers a highly integrated and flexible workflow on stage and in the studio.

NEW StudioLive AI 3-Way Active PA Systems.

The first-ever confluence of advanced acoustic science, massive DSP, full iPad control, and irresponsibly high power.



STUDIO. LIVE. PA SYSTEMS WITH THE accuracy of huge studio monitors. Renown speaker designer David Gunness provided the unique coaxial designs and cutting-edge Temporal EQ™ digital signal-processing technology.

PreSonus provided the new Active Integration™ CPU (with 96 kHz, 32-bit floating-point processing; onboard wireless; and Ethercon) along with a whoop-ass, **2,000 watt**, Class D power amplifier.

Because every StudioLive AI PA full-range system and subwoofer has the equivalent of a $1,500 rack-mount speaker processor built in, they deliver sound quality that simply hasn't been possible before in active systems.

Add the ability to set up, tune, and monitor these speakers using SL Room Control for iPad®, OS X®, and Windows®, and you have the first major PA speaker-design advancement of the decade.

The other fathers of StudioLive AI PA: PreSonus' Jim Odom, Chief Strategy Officer, and Bob "Thinks in Code" Tudor, Chief Technical Officer.

Before cofounding Fulcum Acoustic, **David Gunness** designed pro loudspeakers at Electro-Voice and later at Eastern Acoustic Works (EAW). Gunness holds five patents and writes technology whitepapers, many published by AES. The KF900 large-scale sound reinforcement system, the DSA digitally steered array, and a suite of innovative processing techniques marketed as "Gunness Focusing™" are all results of his career-long emphasis on improving loudspeaker performance with innovative software tools and DSP.

fulcrum ACOUSTIC®

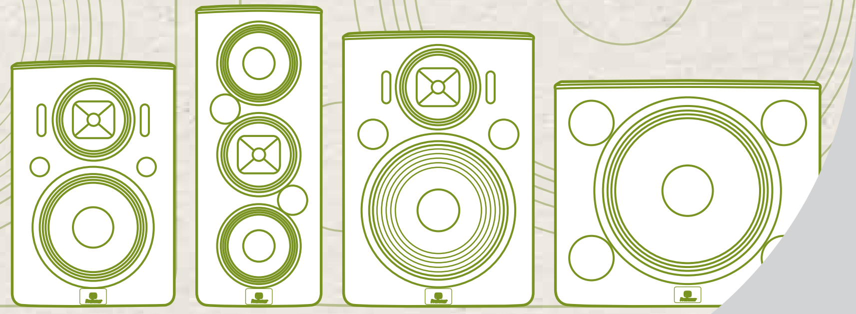
312AI
2,000W
8" CoActual
12" LF
Active Integration

328AI
2,000W
8" CoActual
2 x 8" LF
Active Integration

315AI
2,000W
8" CoActual
15" LF
Active Integration

18sAI 1,000W • 18" LF
Active Integration

StudioLive AI PA



NEW StudioLive AI 3-Way Active PA Systems. Coaxial finally done right.

The text on this page is liberally borrowed from the Fulcrum Acoustic Web site. For the whole story, visit http://www.fulcrum-acoustic.com/technologies/buildingabettercoax

COACTUAL BENEFITS

Crisper stereo image • Greater sound-stage depth • More separation between complex mix components
Increased resistance to feedback • More seamless transitions between distributed loudspeakers
Less fatiguing listening experience at very high SPLs.

The CoActual™ transducer

design used in StudioLive AI PA avoids the typical performance problems associated with designs that use separate, spaced woofers and horns:

Separate woofers and horns can never sum coherently through crossover over a wide listening area. The physical spacing of the horn/compression driver and woofer make this impossible. A coaxial design, with closely spaced HF and LF elements, maintains coherence through the crossover over a wide listening area.

With a separated woofer and horn, the fact that both the LF and HF transducers are providing output slightly above and slightly below crossover results in extreme beamwidth irregularities in the plane in which the drivers are located. When the horn and woofer are co-located, the beamwidth performance can be much smoother through crossover.

In StudioLive AI systems, the woofer and compression driver output are intentionally overlapped to allow for a very smooth beamwidth response through crossover. This cannot be done with spaced devices.

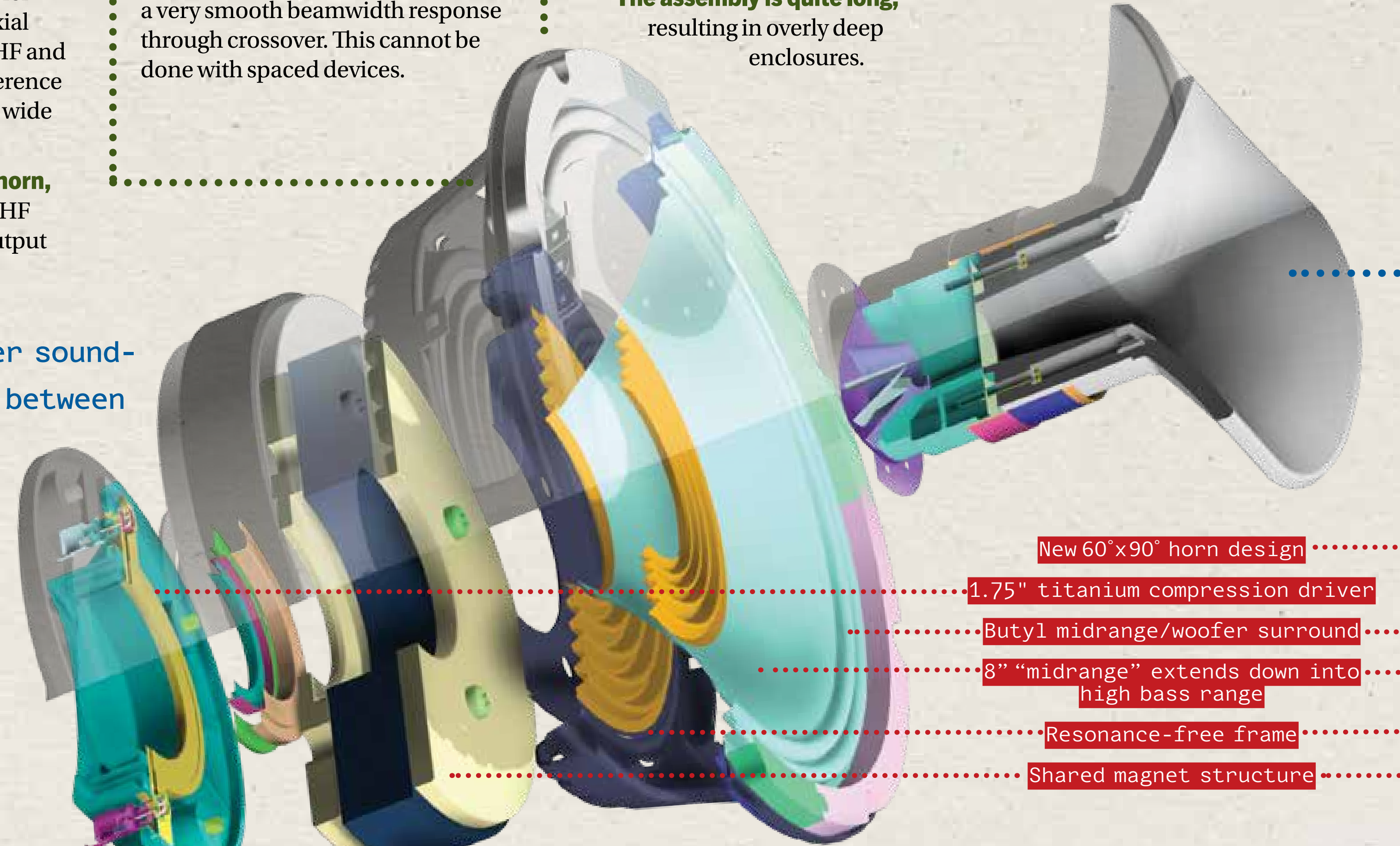
So if coaxial designs solve so many problems, why aren't they more popular?

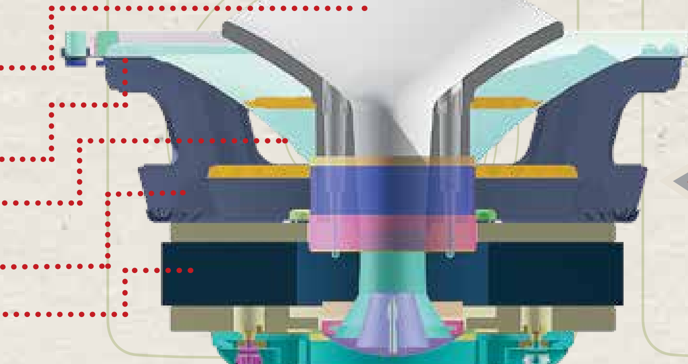
Because conventional coaxial transducers have their own set of problems:

The compression-driver diaphragm and the woofer voice coil are widely separated, resulting in significant distance/time anomalies.

The assembly is quite long, resulting in overly deep enclosures.

The excursion of the woofer modulates the high frequencies, resulting in sonic artifacts that get worse as the driver gets louder (and the woofer-cone excursion increases). The sound character changes dramatically with output level.





The PreSonus CoActual design may look familiar but there are a number of technical innovations that differentiate it from its competition.

The new CoActual driver uses a single magnet structure, shared by the woofer and compression driver. This results in a much more compact, lightweight assembly.

This magnet structure also keeps the compression-driver diaphragm located in close proximity to the woofer voice coil.

A purpose-built high-frequency horn is used for high frequency pattern control and also to keep the HF energy off of the woofer cone. This allows the StudioLive AI PA to remain sonically stable at high levels.

The crossover overlaps the HF horn and woofer, thereby allowing the compression driver to fill in the woofer frequency range that is shadowed by the horn. TQ™ processing allows the compression driver to work at lower frequencies without sonic artifacts.

The woofer's larger radiating surface works in conjunction with the HF horn to improve directional control at the bottom of the horn's operating range. This increases directional control beyond what can be accomplished by the horn alone.

At the low-frequency end of its range, a coaxial horn "leaks" sound back onto the woofer cone. This reflects forward, out of time, and produces tonal colorations that don't respond to conventional EQ. TQ™ eliminates these colorations, thereby removing one of the primary shortcomings of competitors' designs.

STUDIOLIVE Active INTEGRATION TRADE MARK

Q: Why did PreSonus decide to get into the PA loudspeaker business?

Shortly after the launch of StudioLive mixers, PreSonus began dreaming about how to further bring a controlled studio environment to the live venue. StudioLive mixers brought some measure of control but to get to the heart of the issue, we had to go to the source: the powered speaker.

The typical approach used by most powered loudspeakers is to overcome live-sound obstacles with pure brute force. High SPL is great, but a simple two-way system with the HF element directly above the LF unit results in staggered sound arrival times. So this type of loudspeaker cannot possibly have an optimal impulse response over a wide area.

Knowing these problems are nearly universal among the powered loudspeakers on the market today, PreSonus began searching for a loudspeaker technology that would take away as many variables as possible and make a live environment as close to a controlled studio environment as possible.

There was one problem: PreSonus was not a loudspeaker company. That's why we approached Fulcrum Acoustics' legendary engineer, David Gunness. The Fulcrum team instantly understood what PreSonus was trying to accomplish and brought us to the 3-way, 8", coaxial design each StudioLive AI-Series loudspeaker employs.

Temporal EQ is the secret sauce of StudioLive AI 3-Way Active PA Systems. (…which would make Active Integration 's massive DSP the all-meat patty if you're into really labored metaphors.)

The text on this page is liberally borrowed from the Fulcrum Acoustic Web site. For the whole story, visit http://www.fulcrum-acoustic.com/technologies/tqstatemen

temporal equalization tq™

While DSP has long been included in loudspeakers, it has generally been handled as an additional feature, rather than a part of the design of the loudspeaker itself. Designers and engineers would build the best physical loudspeaker they could and then use DSP filters to improve performance and to provide the end user with some EQ options.

Fulcrum Acoustic's TQ™ Temporal Equalization algorithms are included in the design process from the beginning. Rather than choosing a comprise between two competing attributes, like settling for high-frequency horns with imperfect coverage patterns in order to achieve aesthetically pleasing transient response, we physically optimize the attribute that cannot be addressed with DSP and solve the other problem with DSP.

First, TQ uses a complement of Infinite Impulse Response (IIR) high-pass, low-pass, and parametric filters, plus delay, to tune for natural-sounding spectral balance. The result is a pleasing aesthetic balance between highly transient sources (such as percussion instruments) and less transient sources (such as human voice), uncolored vocals, and excellent intelligibility.

TQ provides multiple fully addressable, large Finite Impulse Response (FIR) filters. With this filter we can implement more detailed frequency-response adjustmentsm, and more important, we can implement the precise temporal (time-domain) filters that are responsible for the most remarkable TQ benefits. Loudspeakers tuned with TQ provide a crisper stereo image, greater soundstage depth, more separation between the components of a complex mix, increased resistance to feedback, more seamless transitions between distributed loudspeakers, and a less-fatiguing listening experience at very high SPLs.

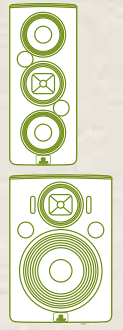
In Fulcrum Acoustic's passive loudspeaker designs, TQ™ is achieved with outboard speaker processors costing between $1,500 and $3,000. In StudioLive AI speakers, it's handled by Active Integration.

Then, because of the addition processing power of Active Integration, we're able to add Fulcrum Acoustic's most powerful tool: a very specific type of fully addressable, large Finite Impulse Response (FIR) filter. This proprietary algorithm lets StudioLive AI PA achieve more detailed frequency-response adjustments.

Q:

What's up with the 328AI design that has two 8-inch bass transducers?

Think of the StudioLive 328AI as a really skinny 315AI for tight, cramped venues where a wider speaker would take up precious stage space and compromise sight lines (i.e., block the heroic profile of your egotistic lead singer). The 328 AI's two 8-inch bass transducers have the same radiating surface as a single 15-inch. And yet it's much skinnier.



All StudioLive AI full-range speakers feature:

2,000 Watt, Triamped, Class D Amplification
- LF 2 x 500 watts
- MF 1 x 500 watts
- HF 1 x 500 watts

Universal Switch Mode Power Supply with Power-Factor Correction

Active Integration 32-bit Floating-Point DSP

Asymmetric Three-Way Crossover

Temporal Equalization:
- Horn reflection
- Linear time and amplitude anomalies correction
- Coaxial Speaker Coherence Alignment
- Performance contouring
- Dynamic limiting
- Excursion limiting

Wireless and wired networking with control over
- Output level
- Operation modes
- …and more with SL Room Control

Four Operation Modes
- Live Performance
- MP3 – Music Playback
- Floor Monitor – Stage Wedge
- User – Custom preset with SL Room Control

External Subwoofer Mode
- Engages custom low-cut filter per model
- Sets phase and time alignment for optimized performance with 18sAI

XMAX™ Mic Preamp with 12V Phantom Power

Balanced Combo Line Input w/ Attenuation

Audio Mix Output

Speaker Attenuation Control

USB 2.0 Port for Included USB Wi-Fi Module and Disaster Recovery

Ethercon Connection

Birch or Plywood Construction

Comfortable, Ergonomic Handles

Integrated M10 Fly Points

Integrated Pole Mounts

1,000 Watt, Class D Amplification

StudioLive 18sAI subwoofer features:

Universal Switch Mode Power Supply with Power Factor Correction

Active Integration 32-bit Floating Point DSP with
- Variable crossover
- Dynamic limiting
- Excursion limiting

Wireless and Wired Networking with Control Over:
- Output level
- User-adjustable contours
- …and more with SL Room Control software.

Custom Designed 18" Ferrite Woofer

Stereo Line Inputs (XLR-¼" Combo)

Stereo Line-Level Throughputs (XLR)

Speaker Level Control (-6 dB to + 6 dB)

Mono Sum Option Sends Summed Signal to Both Outputs

USB 2.0 Port for Included USB Wi-Fi Module and Disaster Recovery

Ethercon Connection

Polarity Invert

Three Onboard Delay Settings:
- 0 ft. – full-range speaker on Sub Pole
- 2 ft. – full-range speaker on speaker stand
- 4 ft. – full-range speaker on stage

Three Operation Modes:
- Normal – live performance
- Extended LF – fatter low end
- User – custom preset with SL Room Control

Optional Dante Networking Card (coming late 2013)

Birch or Plywood Construction

Comfortable, Ergonomic Handles

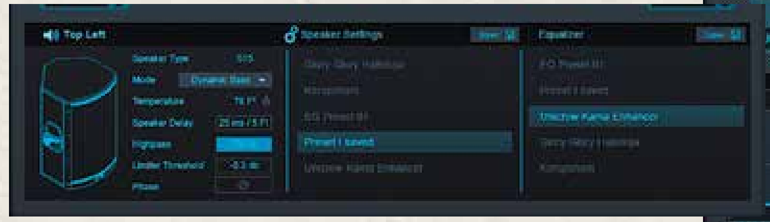
Integrated Pole Insert

Interlocking Stacking Provisions

STUDIOLIVE Active INTEGRATION

Set up, adjust and monitor StudioLive AI 3-Way Active PA Systems with an iPad. It's part of that Active Integration ecosystem of interactive products that we keep going on and on about.

StudioLive AI PA Exclusive: Wireless and wired integration with SL Room Contol



Feature: Speaker Grouping

SL Room Control provides individual and grouped speaker management. Once you have EQ'd and set the delay time and level for each speaker, you can group them together. Grouping speakers allows you to adjust the overall level of your entire FOH or satellite system at once. In addition, you can mute or solo individual speakers in the system while maintaining group level control.

	PreSonus CoActual™ Coaxial								
	HF driver	HF power	MR woofer	MR power	LF woofer	LF power	TOTAL POWER	XMAX mic preamp	Wireless and wired control
S328AI	1.75 -inch	500W	8-inch	500W	2x8-inch	1,000W	2,000W	YES	YES
S312AI	1.75 -inch	500W	8-inch	500W	12-inch	1,000W	2,000W	YES	YES
S315AI	1.75 -inch	500W	8-inch	500W	15-inch	1,000W	2,000W	YES	YES
18sAI	——	——	——	——	18-inch	1,000W	1,000W	——	YES

Wireless and wired integration with SL Room Control

SL Room Control is a speaker-management system and remote-control/monitoring software for StudioLive AI-Series loudspeakers.

It provides remote wireless control over all onboard features, including a 31-band graphic EQ, 8-band parametric EQ, limiter, and speaker delay, as well as network set-up and speaker grouping. SL Room Control also constantly communicates performance parameters such as temperature and clipping.

In short, this powerful software application opens up the power of the onboard DSP, providing optimization tools that were previously only available in stand-alone rack units.

Remote Control Over:

- Onboard Operational modes
- External Sub Mode (full-range only)
- Polarity invert (18sAI only)
- Network setup wizard

Network Scanning to Automatically Detect All Speakers

Network Browser

Performance Monitoring for:

- Excursion limiting
- Over temperature
- ADC clip detection
- Power-amp soft limiting

Group Speaker Management with Level, Mute, and Solo

31-Band Graphic EQ

8-Band Parametric EQ

10 Notch Filters

Limiter

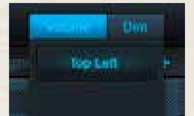
Variable Crossover from 75 – 150 Hz (18sAI only)

Mute

Solo

Speaker Delay (up to 250 ms)

Output Level



EQ Preset Browser

Custom Labels and Comments Field for Each Speaker

Preset Save and Load

Create and Store User Operation Mode Onboard Each Speaker for Use Away from SL Room Control

Windows 7 and 8 (32- and 64-bit)

Mac OS X 10.7 Lion and 10.8 Mountain Lion (32- and 64-bit support)

Apple iOS 5.1 or later

Feature: Speaker EQ.

SL Room Control gives you access to the on-board 31-band graphic and 8-band parametric EQs in your StudioLive AI-speakers. This allows you to set your system EQ per speaker, freeing up your mixer's output EQs for other things.

NEW Sceptre™ Series CoActual™ Studio Monitors.

The first time you hear Sceptre-series CoActual™ 2-way studio monitors, you'll discover fine nuances of your music that can't be reproduced by conventional designs. The Sceptre's panoramic soundstage, fine detail, and stunning dynamics will astonish you. This exceptional performance is the result of an advanced coaxial design that works integrally with a 32-bit, 96 kHz, dual-core processor running Fulcrum Acoustic's TQ™ Temporal Equalization Technology™.

"Rather than choosing a compromise between two competing attributes, we physically optimize the attribute than can't be addressed with DSP… and solve the other problem with DSP."

David Gunness,
co-founder,
Fulcrum Acoustic

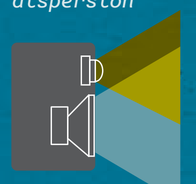
- CoActual technology combines DSP time-correction and point-source design for symmetrical soundstage and micro-definition imaging
- Unique coaxial transducer integrates 8-inch (Sceptre S8) or 6.5-inch (Sceptre S6) midrange driver and 1-inch (25 mm), horn-loaded HF transducer with Coaxial Speaker Coherence Alignment
- 32-bit, 96 kHz, dual-core, active floating-point DSP provides critical Temporal EQ™ with multiple FIR filters
- Acoustic-tuning controls: HF Driver Adjust (0 dB, +1 dB, -1.5 dB, -4 dB) Acoustic Space settings (0 dB, -1.5 dB, -3 dB, -6 dB)
- High-pass filter (Off, 60 Hz, 80 Hz, and 100 Hz Sensitivity (+4 dBu to -10 dBV)
- 200W RMS, Class D biamplification with internal heat sink
- RF shielding, current-output limiting, and over-temperature protection
- Balanced XLR and ¼-inch TRS line-level inputs with A-taper level control
- 102 dB maximum continuous SPL

How Dave Gunness' unique transducer design and Temporal EQ DSP perfected the coaxial monitor.

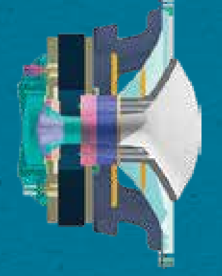
The coaxial conundrum

Speaker designers have long been aware that coaxial designs offer the advantages of a single point source for a consistent acoustic center and a symmetrical dispersion pattern. But previous coaxial designs have had their own problems: The compression driver diaphragm and woofer voice coil are widely separated, resulting in significant distance/time anomalies. The excursion of the woofer modulates the high frequencies, resulting in sonic artifacts that get worse as the driver gets louder (and the woofer cone excursion increases). The sound character changes dramatically with output level. Our CoActual transducers keep the compression-driver diaphragm located in close proximity to the woofer voice coil.

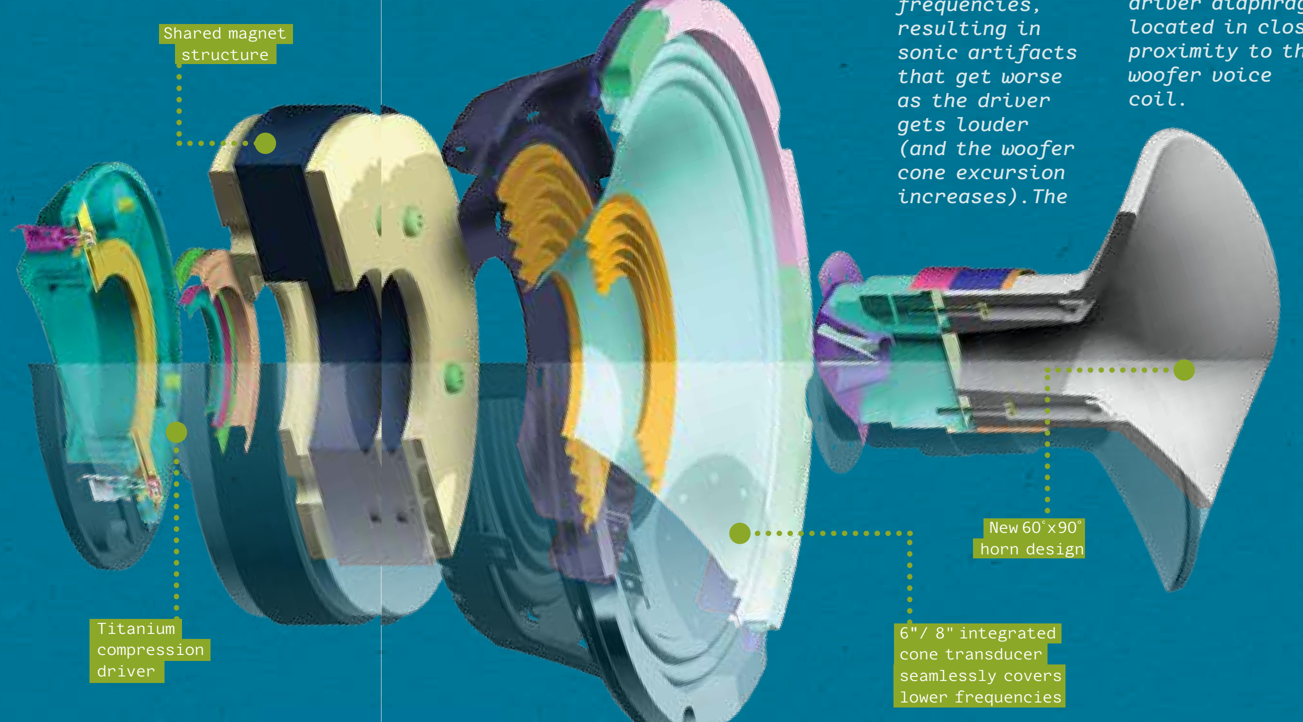
Symmetrical dispersion pattern from a single point source.



Asymmetrical dispersion pattern from two separate sources.



A purpose-built high-frequency horn is used for high frequency pattern control and to keep the HF energy off of the woofer cone. But that's only part of the solution. Fulcrum Acoustic's ingenious (but extremely processor-intensive) DSP algorithms eliminate horn reflections, coloration and spatial variability, and allow high-efficiency bass transducers to cover the vocal range. No other studio-monitor manufacturer has access to this cutting-edge technology. It's what enables Sceptre CoActual monitors to deliver clarity and coherence that has previously only been available in ultra-high-end systems.



NEW

Eris™ Series High-Definition Studio Monitors.

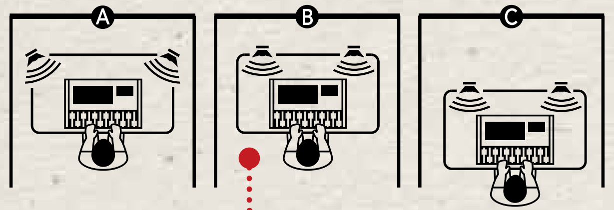
Monitor, mix, and master with affordable, active, 2-way designs that don't treat you like a kid.

Here's a dirty little secret: Most monitors in Eris' price range are designed so that they sound impressive in the listening room — lots of bonky bass and tizzy treble — but they aren't accurate when you're trying to do a serious mix in your studio.

Our Eris Series are true pro monitors with the back-panel acoustic-tuning tools you need to tailor their sound to your room environment and musical genre.

You end up with mixes that sound good everywhere, not just in your studio.

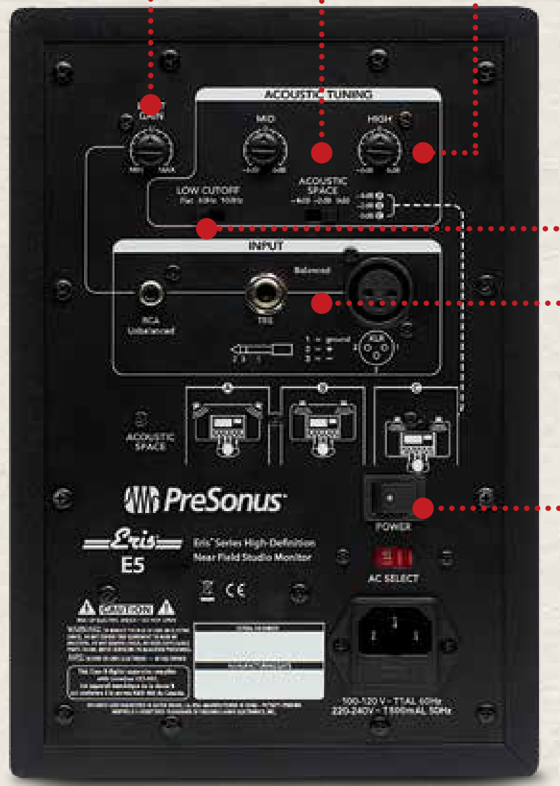
At their heart, the Eris E5 and E8 are state-of-the-art, thanks to long-throw Kevlar™ low-frequency transducers, low-mass silk-dome tweeters, and plenty of responsive Class AB amplification.



Input Gain control for proper level setting

Acoustic Space switch lets you optimize performance in different control-room locations

High and mid-frequency level controls let you customize for each musical genre



Low Cutoff switch allows accurate performance with or without a subwoofer

Balanced XLR / ¼" and unbalanced RCA inputs

Amps deliver high current for fast dynamic response to complex musical waveforms

Eris E5 2-way active monitor

5.25-inch Kevlar™ K100 low-frequency transducer

1-inch (25 mm), ultra-low-mass, silk-dome, high-frequency transducer

70 Watt, Class AB biamplification

Front-firing acoustic port for superior bass-frequency reproduction

Professional control: Midrange (±6 dB, continuously variable), HF (±6 dB, continuously variable), High-Pass (Off, 80 Hz, 100 Hz), and Acoustic Space settings (flat, -2, -4 dB) for accurate mixing contour

Protection: RF interference, output-current limiting, over-temperature, transient, and subsonic

Optimized, resonance-suppressing internal bracing

Balanced XLR / ¼-inch and unbalanced RCA inputs

102 dB maximum continuous SPL

Eris E8 2-way active monitor

8-inch Kevlar™ K100 low-frequency transducer

1-inch (25 mm), ultra-low-mass, silk-dome, high-frequency transducer

130 Watt, Class AB biamplification

Front-firing acoustic port for superior bass-frequency reproduction

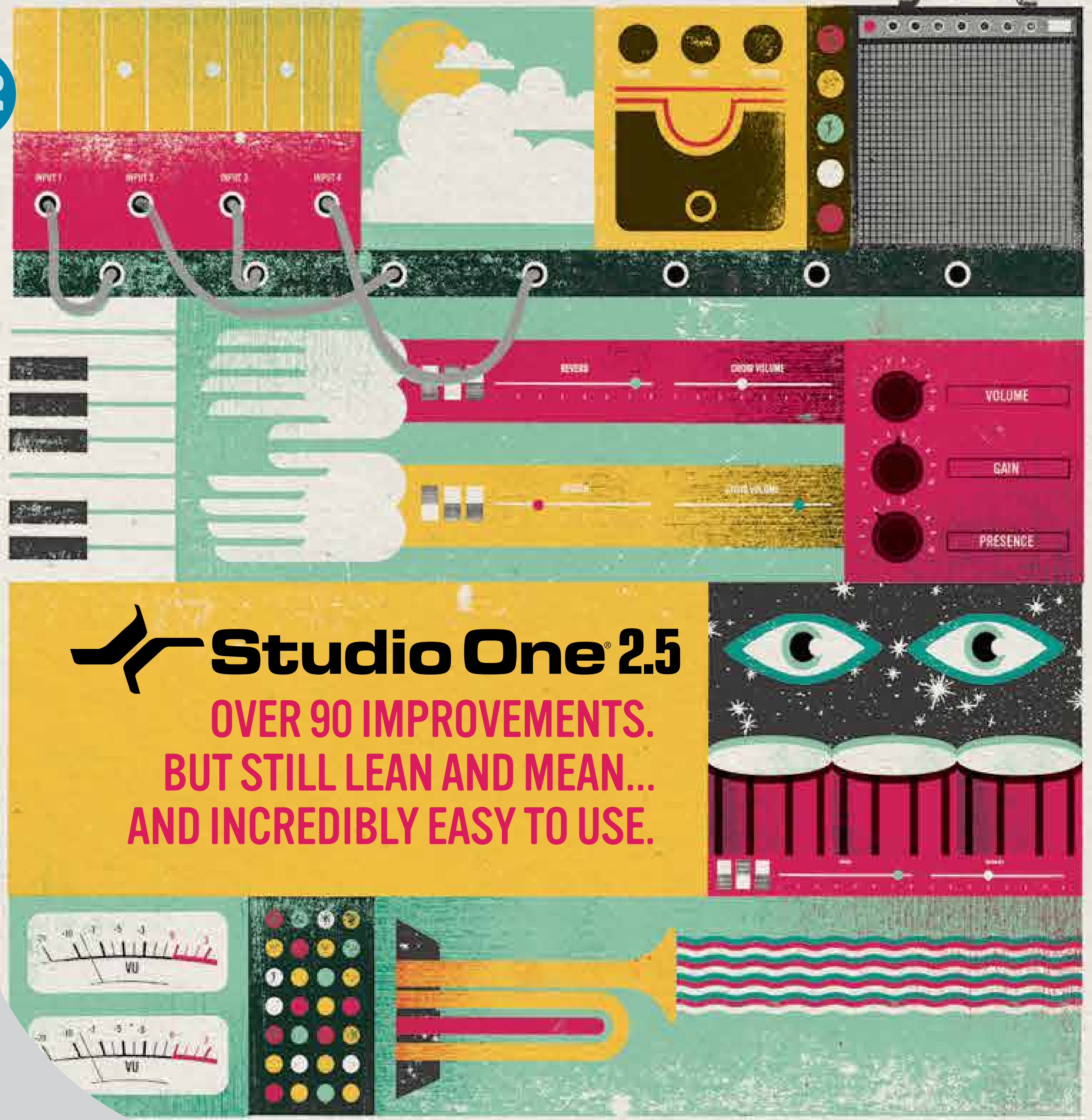
Professional control: Midrange (±6 dB, continuously variable), HF (±6 dB, continuously variable), High-Pass (Off, 80 Hz, 100 Hz) and Acoustic Space settings (flat, -2, -4 dB) for accurate mixing contour

Protection: RF interference, output current limiting, over-temperature, transient, and subsonic protection

Optimized, resonance-suppressing internal bracing

Balanced XLR / ¼-inch and unbalanced RCA inputs

105 dB maximum continuous SPL



An e-mail that came in while we were preparing our brochure... We get accolades all the time but this one seemed especially worth sharing.

“Just a little testimonial

from a FORMER Pro Tools 10 user. Studio One sounds better.

I’ve been using it for 3 hours and am already comfortable doing most regular record/mix functions.

I created a 128-track session (as opposed to the 32-track PT limit). Each track had McDSP 6030 Ultimate Compressor, PreSonus ProEQ, and Waves L-1 Limiter (a total of 384 plug-ins), and I was able to put all tracks in record, punch in/out/in/out, and play back reliably on my 3.4 GHz iMac Core i7 with 16 GB RAM recording to my internal hard drive.

Blown away in San Diego.”

New in Studio One® 2.5

Comping Improvements

Single-swipe comping with automatic beaming into parts makes comping much faster and more powerful.

Folder Track Editing

Edit directly on the Folder Track, with a waveform preview for all tracks in the folder. This simplifies workflow with large track counts.

Track Transform Improvements

Multitrack transform for virtual instruments. Now you can instantly turn complex multi-output virtual instruments into audio tracks and back.

Ampire XT

...has been completely overhauled. All-new cabinet impulse responses, matched with reworked amp models, results in a significant improvement in sound quality.

Automation Enhancements.

Easier and faster editing of automation envelopes.

Audio Export Enhancements.

Rendering multiple outputs at once results in much faster exports—a huge timesaver.

Re-recording from Buses.

This is a completely new feature that enables you to use the output of virtual-instrument or bus channels as the input to an audio track. In this way, you can print from buses or instruments to new audio tracks in real time.

Drag-and-Drop Event Effects.

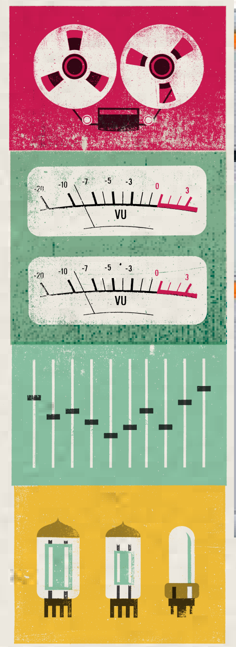
Drag-and-drop workflow is even faster with a new modifier key that lets you do things like drag-and-drop effects directly onto events in the arrangement.

Wet/Dry Mix and Mix Lock for Effects.

Built-in parallel processing on PreSonus Native Effects™. You can drop a compressor on a drum bus, crank the ratio, and drop the threshold to squash the drums mercilessly, and then only mix that squashed sound 20% with the original, unprocessed sound.

Legato and Overlap Correction (MIDI).

Instantly apply or remove legato from MIDI tracks quickly and simply—a huge timesaver.



We are pleased to announce the release of Studio One 2.5, a major update to Studio One 2. This free (to Studio One 2 users) update adds nearly 100 enhancements and features. All versions of Studio One 2 have been updated, including Studio One Artist, Producer, Professional, and Free.

QUALITY & INNOVATION resolution AWARDS 2012 Winner DAW

TEC NOMINEE

AUDIOMEDIA GEAR OF THE YEAR 2013

AUDIOFANZINE.COM

The ADL 700 is now shipping. How one of the world's finest tube preamplifiers led to one of the world's finest tube channel strips.

In 2005, we collaborated with famed tube-circuit designer Anthony DeMaria to create the finest tube preamp that money can buy. Not the finest for the price. The finest—period.

The result was the ADL 600, a distinctive Class A, discrete design that has won a reputation as one of the best-sounding preamplifiers in the world among top recording engineers and producers like Chuck Ainlay, Jimmy Douglass, and Mark Mancina and artists such as Victor Wooten.

The ADL 600 is an ultra-low-noise tube preamp with a big, warm, clear-yet-distinctive sound that makes virtually anything you run through it sound better: smooth, and articulate vocals; deep and tight bass guitars; and rich, full mixes.

A channel of the ADL 600 preamp with with innovative signal processing.

Responding to the requests of professional and amateur musicians and producers alike, PreSonus has now created the ADL 700. It combines a channel of our superb ADL 600 tube topology with totally new compressor and EQ designs from the PreSonus engineering mastermind behind some of our best loved analog circuits (including our award-winning XMAX™ Class A microphone preamp).

A new approach to compressor design.

In our not-so-humble opinion, a preamp as distinctive as the ADL deserved an equally innovative compressor.

Premium channel strips have typically used optical compressors. that are susceptible to temperature fluctuations. As optical compressor components heat up or cool down, the resulting attack and release times can be quite different, even on snare hits in the same song.

The ADL 700 uses a custom-designed FET (Field-Effect Transistor) compressor to emulate a triode tube sound with great reliability. This type of compressor provides a faster attack time and precise repeatability.

Also note that the compressors of two ADL 700s can be stereo linked, allowing for more accurate stereo imaging.

Baton Rouge EQ instead of "British."

The ADL 700 also includes a custom-designed, 4-band semi-parametric EQ that was designed with musicality in mind. The combination of isolated filters and optimized-per-band Q provide subtler signal-shaping without harsh artifacts.

Switchable compressor / EQ signal flow.

Placing the compressor before EQ allows you to make dramatic changes to the EQ settings without needing to alter the compressor setting. But, if you place EQ before the compressor, you get better control over different frequencies, achieving a more natural response. The ADL 700 gives you the best of both.

Selectable microphone impedance.

Lowering or raising the ADL 700 mic-input impedance can create subtle coloring and filtering effects, enabling you to get a wider variety of tonalities without using the EQ.

We spared no expense.

The hand-built ADL 700 employs three military-grade vacuum tubes, operating with ±300V power rails for maximum headroom and superb tone. The special transformer design also ensures low-noise operation, with maximum common-mode rejection.

We use only the finest components right down to the last polypropylene film capacitor, including switched attenuators, analog VU and LED meters, and custom-designed, proprietary transformers. There are no op-amps or ICs in the signal path.

The ADL 700 is not inexpensive. But after you've heard it, we think that you're going to be pleasantly surprised at just how good a value it is.

Give the ADL 700 a serious listening test at your PreSonus dealer today.

- High-voltage, Class A, dual-transformer design
- One hand-selected dual-triode 12AT7 and two hand-selected 6922 vacuum tubes
- >73 dB Gain
- Selectable microphone-input impedance
- Switched Gain and variable fine Trim controls
- Microphone, instrument, and line inputs with Input Select
- Ultra low noise (–100 dB S/N ratio)
- Variable high-pass filter, 48V phantom, –20 dB pad, polarity invert
- Fully variable FET compressor/limiter with attack, release, threshold, ratio, makeup gain, and stereo link
- Four-band, semi-parametric equalizer
- Dual-mode analog VU metering (output and gain reduction)

1. High-Pass Filter
2. Source/ Input Impedance Select
3. Trim
4. Gain
5. Threshold
6. Attack
7. Ratio
8. Release
9. Makeup Gain
10. Low Band EQ Frequency
11. Low Band EQ Gain
12. Low-Mid Band EQ Frequency
13. Low-Mid Band EQ Gain
14. High-Mid Band EQ Frequency
15. High-Mid Band EQ Gain
16. High Band EQ Frequency
17. High Band EQ Gain
18. Level
19. Instrument Input
20. Polarity
21. Phantom Power
22. High Gain
23. Compressor Bypass
24. Meter -6 dB
25. Gain-Reduction
26. LF Peak
27. EQ Bypass
28. EQ/Compressor Reverse
29. HF Peak
30. Output
31. Comp Link
32. Line Input
33. Mic Input

StudioLive 32.4.2AI Digital Mixer

- **32 mic/line inputs** with high-headroom Class A XMAX™ mic preamplifier
- **4 subgroups**
- **Stereo/mono main out**
- **14 auxiliary mixes**
- **48-in/34-out FireWire digital recording interface** (24-bit/44.1 kHz and 48 kHz)
- **32 channel strips**
  - Trim control with –20 to +20 dBV line/-15 to +65 dBu mic gain range (80 dB!)
  - +48V phantom power switch for condenser microphones
  - FireWire Input Select
  - 100 mm precision faders
  - Lighted Solo and Mute buttons
  - Access to Fat Channel functions
  - 15-LED ladder metering + clip LED
  - Analog ¼" insert (rear panel)
- **4 subgroup buses, each with:**
  - Solo
  - Mute
  - Access to Fat Channel functions (except high-pass filter and phase reverse)
- **14 aux sends, each with**
  - Solo
  - Pre/post-fader send
  - Output-level control
  - Access to Fat Channel functions (except phase reverse)
  - Mix and Mix/Pan Fat Channel metering
  - Available sources: 32 input channels, Aux A and B, Tape Input, Talkback
- **4 internal effects sends, each with:**
  - Mute
  - Pre/post-fader send
  - Output-level control
  - Access to Fat Channel (except phase reverse)
  - Effects-send Select for Fat Channel metering
  - Mix button for aux-bus mixing and Fat Channel metering
- **Master section**
  - Aux Input A and B
  - Level Control and Select (Fat Channel metering) switch
  - Access to all Fat Channel functions (except phase reverse)
  - Talkback System
    - Mic Level control
    - Output Select (Aux 1-2, 3-6, 7-10, 11-14, Main)
    - Talk button
    - Rear-panel XLR mic input with level control and continuous 48V phantom power
  - 2 Track In
    - Level control
    - Tape Input to Mains button
    - Digital source on/off
  - Solo Bus
    - Cue Mix volume control
    - PFL/AFL and Solo In Place (SIP) buttons
  - Monitor Bus
    - Headphone-output level control
    - Control-room monitor-level control
    - Solo Bus to Monitor button
    - Tape Input to Monitor button
    - Main L/R Digital Return to Monitor button
    - Main Mix to Monitor button
- **Fat Channel with rotary encoders:**
  - Pan with dedicated 15-LED display
  - Stereo link for input channels, aux buses, and subgroups
  - Polarity Invert (main channels only)
  - High-pass filter: 6 dB/oct., sweepable from Off to 1 kHz (main channels and aux's only)
  - 4-band, fully parametric equalizer
    - Low EQ: sweepable from 36 Hz to 465 Hz, ±15 dB, switchable shelf or peaking
    - Low Mid EQ: sweepable from 90 Hz to 1.2 kHz, ±15 dB, variable Q 0.1 to 4.0
    - High Mid EQ: sweepable from 380 Hz to 5 kHz, ±15 dB, variable Q 0.1 to 4.0
    - High EQ: sweepable from 1.4 kHz to 18 kHz, ±15 dB, switchable shelf or peaking
    - Master EQ On/Off button
  - Gate: Threshold: 0 to -84 dB, Attack: 0.02 to 500 ms, Release: 0.05 to 2 sec. Bandpass Key Filter: 40 Hz to 16 kHz, second-order resonant band-pass filter Q (0.7) with Key Listen function
  - Compressor: Threshold,-56 to 0 dB; Ratio 1:1 to 14:1, LIM = ∞:1; Attack 0.2 to 150 ms; Release 2.5 to 900 ms; Makeup Gain 0 to 28 dB; Soft Knee switch; Auto Mode with 10 ms Attack and 150 ms Release
  - Limiter: variable Threshold -28 dBfs to 0, ∞:1 Ratio
  - Output Assign: 4 subgroups and main
  - All settings can be copied among channels and saved as user presets.
  - 50 channel-strip presets for drums, bass, guitars, keyboards, and vocals
- **Alt EQ/Dyn button -** A/B two complete sets of Fat Channel EQ and dynamics-processor settings
- **4 internal digital effects-processors** 2 reverbs and 2 delays with customizable presets
- **16 31-band graphic equalizers**
  - Main outputs, aux outputs
- **6 mute groups** with All On / All Off switch
- **Scene store and recall**
  - Global Scene Storage: all current StudioLive settings
    - Up to 80 at a time
  - Automatic Global AutoStore
  - Individual channel-strip Scene storage
    - Up to 99 at a time, plus…
    - 50 user-editable factory presets for instruments and vocal
  - 6 Quick Scene buttons—speed dial for Scenes
  - Customizable naming (for example, *"Saturday Gig" or "Main Worship Service"*)
  - Lockout mode to keep inspired amateurs from changing your settings
- **Metering/displays**
  - 32 x 16- LED Fat Channel matrix:
    - Pre-dynamics/pre-fader input
    - Post-dynamics/post-fader output
    - Gain reduction
    - Aux 1-14 and FX A-D output
    - Fader-position recall
  - 8 x 15-LED main meter bank
    - Selected channel level
    - Selected channel gain reduction
    - Sub buses 1 to 4
    - Main stereo outputs
  - 15-LED horizontal Pan/Balance display
  - 64 x 194 LCD matrix
  - Effects parameters
    - Scene creation, storage, and recall
    - System menus
  - 2-digit Channel Selected display
- **Input/output**
  - 32 main inputs, each with XLR mic, ¼" TRS line, and ¼" inserts
  - 2 ¼" TRS stereo (L/R) aux inputs
  - 1 XLR talkback-mic input with phantom power and level control
  - Unbalanced RCA stereo (L/R) tape inputs and outputs
  - XLR stereo (L/R) main outputs with level control
  - ¼" TRS stereo (L/R) main outputs
  - XLR mono output with level control
  - ¼" TRS stereo (L/R) control-room outputs
  - ¼" TRS headphone output
  - 4 ¼" subgroup outputs
  - 14 ¼" aux outputs
  - 32 pre-insert, balanced direct outputs, Ch. 1-8, 9-16, 17-24, 25-32 (DB25 sockets)
  - S/PDIF digital out
  - 2 FireWire S800 (IEEE 1394b) ports
  - Dedicated USB 2.0 port with included Wi-Fi LAN adapter for direct wireless control
  - Optional Dante-plus-dual-FireWire-S800 or Thunderbolt-plus-dual-FireWire-S800 I/O cards
- **Digitalia**
  - High-definition analog-to-digital converters (118 dB dynamic range)
  - Unlimited-headroom, 32-bit floating point, digital mixing and effects processing
  - Internal sample frequency 44.1 kHz and 48 kHz
- **Physical**
  - Rugged, non-flex steel chassis
  - 100 mm long-throw faders
  - Military-grade, quick-touch buttons
  - BNC 12V lamp socket
  - Optional dust covers available

StudioLive 32.4.2AI Included Software

- **Universal Control** control-panel application for Mac® OS X and Windows®
- **Studio One™ Artist 2.5**
  - PreSonus' revolutionary digital-audio workstation with unlimited track count and 4+ GB of third-party resources..
- **Capture™ 2.0**
  - 32-bit, 96 kHz multitrack recording application (unlimited input channels and stereo stream from StudioLive) for Mac® and Windows®
  - Record Now button starts recording with one mouse click
  - Prerecord captures audio up to one minute before you press Record
  - Auto-Save and automatic session recovery
  - Essential editing suite (copy, cut, paste, splice, resize)
  - Soundcheck mode for quick virtual soundchecks
  - Session Lock averts accidental keyboard access
  - Peak LED-style meter bridge with clip indicators
  - Big Meter mode makes monitor a huge meter bridge
  - Marker placement and recall
  - Export between markers
  - Stereo Playback mode for use with computer soundcards
  - Full transport control
  - Import/export individual .wav, .aiff or OpenTL
- **Virtual StudioLive™-AI**
  - Bidirectional control of commonly used mixer parameters
  - Easy drag-and-drop workflow
  - Drag presets directly to channels
  - Drag parts of presets directly to components in the Fat Channel
  - Adjust the Fat Channel gate, compressor, and EQ plus the graphic EQ and effects
  - Quickly drop entire scenes to the mixer for instant recall of all channel, effects, and graphic EQ settings
  - Load effects quickly by simply dragging presets into the GUI
  - Integrated Smaart Measurement Technology™ with easy-to-use wizards for audio analysis, system optimization, and feedback suppression
  - Makes StudioLive as easy to use as Studio One
  - Use the mouse to quickly assign channels to multiple buses, mute, solo, etc.
- **Studio One® Artist 2.5 Computer system requirements**
  - Windows
    - Studio One: Windows XP, Vista, Windows 7 or 8 (32- or 64-bit)
    - Capture: Windows 7 or 8 (32- or 64-bit)
    - Intel Core 2 Duo or Intel Xeon® processor or better) (Intel Core 2 Quad or AMD Athlon™ X4 or better recommended)
    - 2 GB RAM (4 GB or more recommended)
  - Mac
    - Mac OS X 10.6.8 or later
    - Intel Core Duo processor (Intel Core 2 Duo or Intel Xeon® processor or better recommended)
    - 2 GB RAM (4 GB or more recommended)
  - Windows and Mac Systems
    - FireWire S800 (IEEE 1394b) port
    - Internet connection recommended
    - DVD-ROM drive
    - Internal or external 7200 RPM storage drive recommended
    - Monitor with 1280x768 resolution
- **Virtual StudioLive-AI Computer system requirements**
  - Windows
    - Windows 7 or 8 (32- or 64-bit)
    - Intel Core Duo or AMD Athlon X2 processor (Intel Core 2 Quad or AMD Athlon X4 or better recommended)
    - 2 GB RAM (4 GB or more recommended)
  - Mac
    - Mac OS X 10.7.2 or later
    - Intel Core Duo processor (Intel Core 2 Duo or Core i3 processor or better recommended)
    - 2 GB RAM (4 GB or more recommended)
  - Windows and Mac Systems
    - FireWire S800 (IEEE 1394b) port
    - Internet connection recommended
    - DVD-ROM drive
    - Internal or external 7200 RPM storage drive recommended
    - Monitor with 1024x768 resolution
- **SL Remote-AI Computer system requirements**
  - Apple iPad
    - Apple iOS® 5.1 or later
    - Apple iPad® 1 or later
- **QMix-AI Computer system requirements**
  - Apple iPad
    - Apple iOS® 5.1 or later
    - Apple iPhone® 3GS or later or iPod touch®

PRM1

- Type: Black electret condenser
- Element: Pressure, FET preamplifier
- Polar Pattern: Omnidirectional
- Connections: Polarity Pin 2 output positive voltage (relative to Pin 3) when diaphragm receives positive pressure (moves inward)
- Connector:3 pin XLR Male
- Performance
  - Frequency response: 20 Hz to 20 kHz
  - Sensitivity (@ 1 kHz open-circuit voltage) -37 dB/Pa (14 mV/Pa) 1 Pa=94 dB SPL
  - Rated impedance: 200Ω
  - Minimum load impedance: 1,000Ω
  - EIN (A-weighted): 26 dB
  - Max. SPL (1 kΩ load): 132 dB SPL (THD ≤ 1%, 1 kHz)
  - Dynamic range (1 kΩ load): 106 dB
  - S/N ratio: 70 dB
- Power
  - Phantom power: 9-52V
  - Current consumption: 2 mA
- Physical
  - Length: 7.56" (192 mm)
  - Width: 0.83" (21 mm)
  - Weight: 5.3 oz (150g)
  - Environmental conditions: 10-50˚ C, humidity 0-95%

Capture 2.0

- **Capture™ 2.0 Computer system requirements**
  - Mac
    - Mac OS® X 10.6.8 or later
    - Intel® Core™ Duo processor (Intel Core 2 Duo or Intel Xeon® processor or better recommended)
    - 2 GB RAM (4 GB recommended)
  - Windows
    - Windows® 7 x64/x86 SP1, Windows 8 x64/x86
    - Intel Core Duo or AMD Athlon® X2 processor (Intel Core 2 Quad or AMD Athlon X4 or better recommended)
    - 2 GB RAM (4 GB recommended)
  - Mac and Windows Systems
    - Internet connection (needed for activation)
    - DVD-ROM drive
    - Monitor with 1280 x 768 resolution
    - 20 GB hard-drive space

StudioLive AI Active Integration PA Systems

- **StudioLive 312AI**
  - Transducers
    - (1) 12-inch, ferrite, low-frequency transducer
    - (1) coaxial 8-inch, ferrite midrange driver
    - (1) 1.75-inch titanium compression driver
  - Horn Dispersion: 60˚ x 90˚
  - Power: 2,000W, Class D triamplification
    - 500W HF
    - 500W MR/LF
    - 1,000W LF
  - Inputs / Outputs
    - (1) combo XLR/ balanced ¼" TRS line-level input with attenuation control
    - (1) balanced XLR microphone input
    - (1) balanced XLR mix output
    - Ethercon (control only)
    - USB for use with included WiFi LAN adapter
  - Dimensions (W x D x H): 16.22" (412 mm) x 19.42"(493.3 mm) x 23.80" (604.5 mm)
  - Weight: 62 lbs./28.12 kg
  - AC power requirements: 100-240 VAC, 50/60 Hz
  - Optional Dante Ethernet network card
  - Accessories
    - Pull-tested Sub Pole
    - M10 Kit (four M10 eye bolts)
- **StudioLive 315AI**
  - Transducers
    - (1) 15-inch, ferrite, low-frequency transducer
    - (1) coaxial 8-inch, ferrite midrange driver
    - (1) 1.75-inch titanium compression driver
  - Horn Dispersion: 60˚ x 90˚
  - Power: 2,000W, Class D triamplification
    - 500W HF
    - 500W MR/LF
    - 1,000W LF

SPECIFICATIONS

- **StudioLive 315A (continued)**
  - Inputs / Outputs
    - (1) combo XLR/ balanced ¼" TRS line-level input with attenuation control
    - (1) balanced XLR microphone input
    - (1) balanced XLR mix output
    - USB for use with included WiFi LAN adapter
    - Ethercon (control only)
  - Dimensions (W x D x H): 24" (609.6 mm) x 21.53" (546.9 mm) x 26.42" (671.1 mm)
  - Weight: 71.6 lbs. (32.48 kg)
  - AC power requirements: 100-240 VAC, 50/60 Hz
  - Optional Dante Ethernet network card
  - Accessories
    - Pull-tested Sub Pole
    - M10 Kit (four M10 eye bolts)
    - Protective cover

- **StudioLive 328AI**
  - Transducers
    - (2) 8-inch, ferrite, low-frequency transducers
    - (1) coaxial 8-inch, ferrite midrange driver
    - (1) 1.75-inch titanium compression driver
  - Horn Dispersion: 60˚ x 90˚
  - Power: 2,000W, Class D triamplification
    - 500W HF
    - 500W MR/LF
    - 2 x 500W LF
  - Inputs / Outputs
    - (1) combo XLR/ balanced ¼" TRS line-level input with attenuation control
    - (1) balanced XLR microphone input
    - (1) balanced XLR mix output
    - USB for use with included WiFi LAN adapter
    - Ethercon (control only)
  - Dimensions (W x D x H): 15" (381 mm) x 15.68" (398.27 mm) x 29.13" (739.9 mm)
  - Weight: 56 lbs. (25.4 kg)
  - AC power requirements: 100-240 VAC, 50/60 Hz
  - Optional Dante Ethernet network card
  - Accessories
    - Pull-tested Sub Pole,
    - M10 Kit (four M10 eye bolts)
    - Protective cover

- **StudioLive 18sAI**
  - Transducer
    - (1) 18-inch, ferrite transducer
  - Power
    - 1,000W, Class D amplifier
  - Inputs / Outputs
    - (2) combo, balanced XLR-¼" TRS, stereo line inputs
    - (2) balanced XLR, switchable stereo/summed-mono line outputs
    - USB for use with included WiFi LAN adapter
    - Ethercon (control only)
  - Dimensions (W x D x H): 25.98" (659.9 mm) x 24" (609.6 mm) x 21.85" (555 mm)
  - Weight: 94 lbs. (42.64 kg)
  - AC power requirements: 100-240 VAC, 50/60 Hz
  - Optional Dante Ethernet network card
  - Accessories
    - Mountable Sub Dolly
    - Pull-tested Sub Pole
    - StudioLive AI Product Pack (includes Sub Dolly, Sub Pole)
    - Protective cover

- **SL Room Control**
  Computer system requirements
  - Mac
    - Mac® OS X 10.7.2 or later
    - Intel Core Duo processor (Intel Core 2 Duo or Core i3 or better recommended)
    - 2 GB RAM (4 GB or more recommended)
  - Windows
    - Windows® 7 x64/x86 SP1, Windows 8 x64/x86
    - Intel® Core™ Duo or AMD Athlon™ X2 processor (Intel Core 2 Duo or AMD Athlon X4 or better recommended)
    - 2 GB RAM (4 GB or more recommended)
  - Mac and Windows Systems
    - Monitor with 1024x768 resolution
  - Apple iPad
    - Apple iOS 5.1 or later
    - Apple iPad 1 or later

Studio One 2.5

- **Studio One 2.5**
  Computer system requirements
  - Mac
    - Mac OS® X 10.6.8 or later
    - Intel® Core™ Duo processor (Intel Core 2 Duo or Intel Xeon® processor or better recommended)
    - 2 GB RAM (4 GB recommended)
  - Windows
    - Windows® 7 x64/x86 SP1, Windows 8 x64/x86
    - Intel Core Duo or AMD Athlon® X2 processor (Intel Core 2 Quad or AMD Athlon X4 or better recommended)
    - 2 GB RAM (4 GB recommended)
  - Mac and Windows Systems
    - Internet connection (needed for activation)
    - DVD-ROM drive
    - Monitor with 1280 x 768 resolution
    - 20 GB hard-drive space

Sceptre™

- **Sceptre™ S6 / S8**
  - Transducers
    - (1) coaxial 6.5" (S6) or 8" (S8) midrange driver
    - 1-inch (25 mm) horn-loaded HF transducer
  - Power
    - 200W RMS, Class D biamplification
  - Inputs / Outputs
    - (1) balanced XLR line-level input
    - (1) balanced ¼" TRS line-level input
  - 102 dB maximum continuous SPL
  - Frequency response
    - S6 = 53 Hz - 22 kHz
    - S8 = 35 Hz - 22 kHz
  - Dimensions (W x D x H)
    - S6 = 9" (22.6 mm) x 9.6" (243.8 mm) x 12" (304.8 mm)
    - S8 = 11" (279.4 mm) x 10.5" (266.7 mm) x 15" (381 mm)
  - Weight
    - S6 18 lbs. (8.16 kg)
    - S8 23.3 lbs. (10.57 kg)
  - AC power requirements: 100-240 VAC, 50/60 Hz

Eris™

- **Eris™ E5 / E8**
  - Transducers
    - (1) 5.25" (E5) or 8" (E8) Kevlar low-frequency transducer
    - (1) 1-inch (25 mm) ultra-low-mass, silk-dome, high-frequency transducer
  - Power
    - 70W (E5) or 130W (E8) Class AB biamplification
  - Inputs / Outputs
    - (1) balanced XLR line-level input
    - (1) balanced ¼-inch TRS line-level input
    - (1) unbalanced RCA line-level input
  - 102/105 dB maximum continuous SPL
  - Frequency response
    - E5 = 53 Hz - 22 kHz
    - E8 = 35 Hz - 22 kHz
  - Dimensions (W x D x H)
    - E5 7" (178 mm) x 7.68" (195 mm) x 10.24" (260 mm)
    - E8 9.84" (250 mm) x 11.77" (299 mm) x 15.12" (384 mm)
  - Weight
    - E5 10.2 lbs (4.63 kg)
    - E8 22.2 lbs (10.07 kg)
  - AC power requirements: 100-240 VAC, 50/60 Hz

ADL 700

- **ADL 700**
  - (1) 12AT7 and (2) 6922 vacuum tubes
  - 300V preamplifier rails
  - Input / Output
    - (1) unbalanced ¼" instrument input (front panel)
    - (1) transformer-balanced XLR line input
    - (1) transformer-balanced XLR mic input
    - (1) transformer-balanced XLR line output
  - Frequency response: 10 Hz to 45 kHz, ±1 dB
  - S/N ratio: -100 dB
  - Gain: >73 dB
  - Dimensions (WxDxH): 19" (482.6 mm) x 17" (431.8 mm) x 3.5" (88.9 mm)
  - Weight: 22.75 lb. (10.32 kg)
  - AC power requirements: 100-240 VAC, 50/60 Hz

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