Welcome to Broadcast 3.0
NETWORK. AUDIO. VIDEO. CONTROL.

Lawo compact 2019
Broadcast 3.0 is based on the cornerstones of IP transport, software-defined processing, orchestration and seamless control of network resources, and automated workflows. This 3rd generation of broadcast infrastructure solutions raises production capabilities to a new level, enabling more efficient utilization of resources and smarter content creation.
Lawo’s vm_dmv is a virtual module app (VM) for Lawo’s V__matrix IP routing & processing platform, complementing the existing vm_mv16-4, vm_mv18-4 and vm_mv24-4 multiviewer line-up. It turns the V__matrix C100 software-defined processing blade into a distributed, infinitely expandable, true IP multiviewer.

What does it do?
The vm_dmv is based on a distributed architecture where multiple modules network together. These modules could be hosted together in the same V__matrix frame, in different frames or even at different geographical locations. Basically anywhere as long as they are networked together via IP.

Every vm_dmv has an input stage capable of receiving up to 24 sources of any combination of 4K/3G/HD/SD which is limited only by the physical (up to 18 SDI inputs) or network (2x 40GbE) I/O. These sources are downscaled by the vm_dmv and returned to the network as Lawo LiveView™ IP streams.

In parallel to the input stage, every vm_dmv also features an output stage capable of creating up to eight 3G mosaics (or two 4K mosaics) with up to 64 sources/PIPs each (128 each in 4K). The output stage compiles a mosaic from the appropriate LiveView™ streams needed, automatically taking into account the size the user requests for the PIPs. The output stage can both use LiveView™ streams that it has generated by its own input stage or by subscribing to other LiveView™ streams from the network.

As any vm_dmv can use any LiveView™ stream from any other vm_dmv on the network it scales linearly with each vm_dmv app that is added to the network which results in an “infinitely” expandable and distributed multiviewer.

The result is not only the world’s first infinitely expandable multiviewer, it’s also a solution that significantly reduces rack-space, weight and power-consumption. In addition, Lawo’s V__matrix platform is already renowned for its software-defined functionality where C100 blades can be changed at run-time by loading different virtual modules.

NEW: LiveView™ Decoding Engine
This server-based solution enables the V__matrix vm_dmv multiviewer to show both production (ST2110/ST2022-6) and transmission (MPEG/OTT) formats on the same screen for complete end-to-end visibility. The LiveView™ Decoding Engine integrates full frame rate MPEG-2/4/10/HEVC decoding for any Lawo LiveView™ enabled product.

KEY FEATURES

- Virtual Module app (VM) for V__matrix C100 core processing blade
- Distributed multiviewer architecture with unlimited inputs and heads
- Full support of IP and SDI sources in 4K, 3G, HD and SD
- Lawo LiveView™ support
- Support of embedded and discrete audio
- Pixel perfect mosaics with ultra-low latency
- Intuitive drag&drop layout configuration with Lawo’s “theWALL”

World’s 1st Infinitely Expandable True IP Multiviewer
smartDASH

System Monitoring And Realtime Telemetry

What is it?
The Lawo System Monitoring And Realtime Telemetry Dashboard (aka smartDASH) is a vendor-agnostic enterprise software suite designed to provide full network and media visibility across an all IP, all SDI or hybrid WAN/LAN broadcast infrastructure.

What does it do?
Strategically positioned to bridge the gap between IT and Video engineering, smartDASH straddles both sides of the operation to provide a comprehensive view of what the network is doing and how the media streams flowing across the network are behaving. Based on a UNIX OS, this software defined networking solution incorporates a powerful and robust database to document and rapidly search any aspect of the operation, from a simple cable ID number to seeking the journey of a multicast across a transnational multi-hop WAN. Additionally, by leveraging a vast library of hardware communication protocols, the system automatically interrogates live and dormant path connections to create the most intuitive and data rich presentation layers of a COTS-hybrid infrastructure. With its award winning deep packet inspection microservices, smartDASH supports monitoring and decoding a wide range of media formats, from low bitrate OTT/ABR streams to uncompressed ST2110 studio production flows, in addition to characterizing the packet pacing off the delivery network. This unique approach brings a deeper dimension of operational visibility by unifying network telemetry and mixed media flow into a single glass view. smartDASH users have a zero-footprint installation and can be deployed on-prem, private or public cloud with HTTP accessibility from a browser or mobile device. This vendor-agnostic state-of-the-art software solution is an essential component to the conscious operation of any IP-based broadcast installation.

KEY FEATURES
- Scalable enterprise software providing data from every corner of your network
- Effortlessly document all aspects of your media network and infrastructure
- Keep track of device inventory, including warm and cold spares
- Account for CAPEX and OPEX KPIs to manage total cost of ownership
- Visualize connectivity and bandwidth usage with precision
- Discover and trace live media flows from origination to destination
- Convert network and media data points into actionable intelligence
- Advanced telemetry agents provide in-service and out-of-service alarm generation and reporting
- Real-time communication and telemetry from COTS platforms to third party purpose build equipment
- High performance deep packet inspection of mixed media flows
- Comprehensive and unified presentation layer dashboards
- Improve your revenue through awareness, speed of data

NEW: Auto-Topology Microservice
This new feature automatically discovers, registers and documents the IP network topology by leveraging OPENCONFIG, a vendor-neutral model-driven communication protocol, to present device status in real time. If it's on the network, the Auto-Topology’s inter/intra-facility discovery functionality will find it, analyze it and present it in smartDASH’s topology view with clarity.

NEW: PTP Monitoring Microservice
This new feature detects, classifies and analyzes PTP at the network layer, and acts as a referenced, locked receiver for essential timing measurements for video, audio and data streams. New features include auto detection of PTP, type modes, histograms, slave ports and lock state.

NEW: Cloud-enabled
smartDASH can now run on virtual machines and in the cloud for running network device analysis of cloud-based application, e.g. virtualized playout channels. smartDASH supports solutions based on AWS, Google, VM Networks and others.

NEW: smartDS
smartDS is a complementary data storage solution to smartDASH, allowing smartDASH users to log and store months of alarming data logs for a full record of everything from network bandwidth to signal loss. smartDS’s reports feature can access and analyze the smartDS data to create rich data reports for operations, engineering and management tiers within the organization. The smartDS software can run on a dedicated hardware server or in virtualized or cloud-based environments.

Get the complete SMART brochure here.
smartSCOPE
Deep Packet Media Inspection & Network Analyzer

What is it?
smartSCOPE is a media-agnostic, high density 24/7 analysis platform for IP flows in live production and delivery networks.

What does it do?
IP flows can frequently suffer from packet loss, jitter, encoding impairments and transport violations to name a few. Having the right monitoring and visualization technologies in place to proactively identify impairments and service disruption is imperative to the success of the operation. The smartSCOPE incorporates the essential analysis and decoding processing necessary to alert the operation tiers of service and packet transport related impairments for ST2110/ST2022 production flows, linear MPEG services and over-the-top adaptive bitrate (OTT/ABR) defined streams. At the heart of the smartSCOPE, packets are processed for service compliance with also the unique ability to analyze the conditions of the delivery network. This dual focused analysis approach provides a clear demarcation between video delivery and video processing thereby drastically reducing the mean time to repair (MTTR) and eliminating finger pointing between transport and processing departments.

+++ NEW FEATURES +++

NEW: Support of live production formats
In addition to existing transmission format support, smartSCOPE now also supports SMPTE ST2110-20/21/30/40 and ST2022-6/7 live production video formats. As a result, smartSCOPE allows integrated monitoring of both compressed and uncompressed streams in high capacity hybrid systems. This new approach to signal and network probing provides full confidence and precision analysis for operations and engineering over managed and unmanaged networks. Scalable to over 100 services in a single instance, it supplies a much-needed solution to broadcasters leveraging COTS server architectures.

KEY FEATURES
- Acquires, decodes and analyzes media flows across multiple high capacity IP interfaces
- Supported compressed formats: MPEG-2 TS, H.264, HEVC
- Supported production formats: ST2022-6/7, ST2110-20/30/40, PTP
- Supported OTT formats: HLS, HDS, RTMP, DASH
- Supported audio formats: AES67, PCM, MP1-L1, AAC, HE-AAC, A-52

SPECS
- Stand-alone application or seamlessly integrated with smartDASH
- Available on HW / SW / VM / cloud platforms
- 2x 10GbE /40GbE

Get the complete SMART brochure here.
Virtual Studio Manager

What is it?
IP Broadcast Control System

What does it do?
VSM (Virtual Studio Manager) is a hardware independent Broadcast Control System that runs on an IP backbone and integrates easily with the majority of the most popular broadcast equipment on the market. These include video routers, video switchers, audio routers, audio consoles, multiviewers, intercoms, modular equipment and many third-party devices. By talking native protocols where possible, equipment from different manufacturers can be seamlessly “glued” together, giving unmatched recall and logic control possibilities system-wide. Control interfaces in the form of a wide range of hardware LCD button panels and software panel clients allow simplified operation from highly custom designed configurable GUIs. With a modern TCP/IP backbone and strong redundancy concepts, VSM can control complete broadcast facilities, thus giving the users the freedom to create their own workflows and production setups.

### KEY FEATURES

- Control System to be used in all areas of broadcasting
- Integrates with the majority of the most popular equipment
- Based on a IP backbone using standard IT hardware
- A single control interface for numerous devices
- Third-party hardware manufacturer-independent
- Dynamic router tieline management that includes transparent Tally logic
- Strong redundancy architecture designed for 24/7 non-stop operation
- Control complete broadcast facilities with LCD button panels or custom designed configurable GUls
- Users can easily deploy their individual workflows and production setups

+++ NEW FEATURES +++

NEW: HTML5 vsmWebPanel
The touch-operation optimized vsmWebPanel turns mobile devices into powerful and flexible VSM control panels, enabling full control of the broadcast infrastructure from basically everywhere.

NEW: HTML 5 UI for vsmGadgetServer
Make use of HTML5-capable web-browsers to configure a broad range of control connections of vsmGadgetServer from any host and any device without the need for installing and maintaining specific software.

NEW: Ember+ Gateway in vsmStudio
Provide any set of parameters to other instances of vsmStudio using Ember+ and start building your own VSM control cloud. Share essential parameters of your installation and access them anywhere: facility-wide, country-wide, world-wide - the network is the limit.

NEW: Panel Inheritance in vsmStudio
Experience the new efficiency of panel creation. With panel inheritance, panel layouts and functionalities are derived within a panel hierarchy. Panel editing is concentrating on master panels and individual adaptions. You will get the job done faster, and changing layouts whenever necessary becomes a breeze.
vsmSOUL

What is it?
Lawo’s vsmSOUL Seamless Orchestration and Unification Layer is adding an overarching orchestration service for IP-based production environments to the VSM control system.

What does it do?
vsmSOUL is aware of, and handles, information from all system components. It manages the generation and routing of audio and video streams in any multi-vendor IP setup, and is compatible across individual interfaces and technical solutions. vsmSOUL provides a single point of control for any network size and any network topology, seamlessly integrated into vsmStudio and vsmGadgetServer.

vsmSOUL provides the central service for stream routing and resource management across single-switch, spine-leaf, or mesh network infrastructures. Through vsmStudio, it provides a unified northbound matrix representation of the network towards an overall control system. Using standardized or vendor-specific APIs, vsmSOUL accesses switches and network components, including encoding and decoding devices, cameras, multiviewers, processors, switchers, consoles, etc., to directly control the generation, registration, routing and monitoring of streams. It follows industry specifications like NMOS to utilize flows. In addition, proprietary interfaces are used to achieve a more sophisticated control over edge devices.

KEY BENEFITS
- Vendor neutrality for network nodes and IT switches
- Designed for multi-vendor employment
- Unified northbound matrix representation of the network through vsmStudio
- Capable of Hitless Merge
- Sophisticated system redundancy
- Broadest third-party control capabilities in combination with VSM
- Highest operational UI flexibility using VSM hardware and software panels
- No workflow changes for the operator

SPECS
- Northbound abstraction of the network through vsmStudio
- Switch-API support southbound, with access to multicast routing and native switch functionality
- Full Layer 3 compatibility
- Agnostic to various switching mechanisms. Supported switching modes: Patching, Make-before-break, Break-before-make...
- Compatible with NMOS IS-04, IS-05, SMPTE 2022-6/-7, 2110, AES67, RAVENNA
- Well known user interface for configuration and operation

Get the complete VSM brochure here.
**What is it?**
Audio production console for mobile production, broadcast studies, performing arts and recording.

**What does it do?**
This 3rd generation of the mc² 56 represents the next step in the evolution of a console that has dominated the audio production industry with hundreds of units in operation around the world. Designed to deliver unrivaled innovation, it provides not just pure and simple access to ultimate performance – it’s a global standard redefined.

### KEY FEATURES
- Perfect for outside broadcast vehicle dimensions: 64 faders fit across most standard installations.
- 21.5” super-precise full HD touch-screens, color-TFTs in channel strips and touch-sensitive color-coded encoders.
- Extended free controls with direct access to four parameters in addition to gain.
- Dust-proof long-life high-performance faders.
- Designed for multi-user operation.
- LiveView™ Video Labels.
- IP-Share™ Gain Compensation.
- DSCA™ Dynamic Surface to Core Allocation.
- Parallel “New York” Compression.
- Superb tools for surround and 3D / immersive sound mixing.
- Integrated Loudness Metering.
- Two customizable user panels incl. option for RTW TM7/TM9 Goniometer.
- Comprehensive Local I/O.
- Energy-saving low-noise design.

### SPECS
- Frames with 16 to 144 faders.
- Up to 8,192 x 8,192 crosspoints.
- Up to 1,024 DSP channels.
- Up to 192 summing buses.
- Up to 128 aux buses.
- 44.1 – 96 kHz operation.
- Designed for IP-based infrastructures with native support for all relevant IP standards: SMPTE ST2110, AES67, RAVENNA and DANTE®.

Get the complete mc²56 brochure here.
Ultra-high Density IP DSP Engine for mc² Consoles

What is it?
The UHD Core is a network-based, software-defined audio DSP engine with unparalleled processing density elevating mc² 56 and mc² 96 consoles to the next dimension.

What does it do?
Utilizing the IP network as an extension of the console core’s backplane, Lawo’s UHD Core can be located anywhere on the network. Its ultra-high processing density with >1,000 mc² fully featured DSP Channels can either be utilized by a single mc² console for coping with even the most challenging productions or be shared amongst up to four consoles. Due to a flexible licensing model the UHD Core is ideal for both mobile applications and facility use. For mobile productions the scalable DSP performance with temporary licenses is a great way to turn CAPEX into OPEX, whilst in facility applications the possibility of resource pooling and flexible allocation of DSP resources to multiple consoles can significantly increase the utilization of the audio infrastructure investments.

The UHD Core features low-noise cooling and is set to fulfill highest demands in production quality and reliability. Eight independent 10/1 GbE capable network interfaces enable the use of redundant networks via ST2022-7 seamless protection switching (SPS). For management the unit provides two redundant RJ45 1GbE management ports. In addition, full hardware redundancy is achieved by a 2nd hot spare unit which permanently mirrors all settings.

The system latency is comparable with conventional architectures connected via the backplane: a special high-performance RAVENNA profile provides network roundtrip latency in the sub-millisecond range whilst the processing power and speed of the UHD core outperforms the processing latency of multiple DSP devices.

As the UHD Core’s functionality is defined by its software, it’s a future-proof investment with a feature-set that is designed to expand.

Get the complete A__UHD Core brochure here.
IP Audio I/O & DSP Node for Remote Production

**What is it?**
The Power Core RP is a fully featured Remote Production solution for mc² audio consoles with integrated modular I/O, DSP and IP streaming capabilities.

**What does it do?**
Power Core RP combines modular audio I/O and high-density DSP processing functionality into a WAN-capable IP node. The unit’s ST2022-7 network redundancy and its Class-C jitter/network latency robustness eliminate the need for dedicated WAN gateways from 3rd party suppliers and thus reduces the set-up complexity of remote productions, plus it reduces potential single points of failure. The unit’s comprehensive audio connectivity includes two redundant 1GbE SFP ports for AoIP, one MADI port (with a 2nd port for redundancy), plus eight modular I/O slots which can be filled with a mix of Mic, Line and AES3 cards. Also a studio card with Mic/Line in/out and two headphone amplifiers is available.

Power Core RP’s DSP capabilities include 64 fully featured processing channels and provide low-latency on-site monitor and IFB mixing. mc² consoles at home have full control of all relevant channel parameters (gain, fader, mute, EQ, dynamics, Aux Send Level…) of the DSP node at the remote location. A touch-screen optimized control GUI provides additional control for on-site and remote operation.

The Power Core RP is standards-based and natively supports ST2110-30/-31, AES67 and RAVENNA AoIP networking. The unit is the perfect mate for Lawo’s video contribution solution V_remote4, which provides WAN-capable bi-directional signal transport for 4K/3G/HD/SD-SDI video signals plus a comprehensive set of onboard video processing features.

**SPECS**
- 19” / 1RU
- Software-defined hardware
- Standards-based ST2110 / AES67 / RAVENNA AoIP networking
- ST2022-7 Network Redundancy with SPS
- Modular I/O concept with 8 slots for Mic, Line and AES3 connectivity

**KEY FEATURES**
- Native IP Remote Production Node with on-board WAN capability
- Low-latency on-site audio processing for monitoring & IFB mixing
- Up to 64 processing channels
- Full remote control of all relevant channel parameters (Gain, Fader, Mute, EQ, Dynamics, Aux Send Level…) from mc² consoles
- Touch-screen optimized control GUI for additional on-site and remote operation

Get the complete Power Core RP information here.
**What is it?**

A__stage AoIP nodes are compact devices capable of streaming uncompressed broadcast-quality audio to Layer 3 networks in real time via both WAN and LAN connections.

**What does it do?**

The A__stage nodes effortlessly convert audio – mic, line-level, AES3 and even digitally-encoded baseband MADI – to audio-over-IP streaming traffic. Like any A__line device, the nodes use open-standard SMPTE 2110-30/31, AES67 and RAVENNA protocols to transport uncompressed audio in real-time on Layer 3 IP networks; IP audio streaming is managed using either the open-source Ember+ control API, or standards-based RAVENNA advertising and discovery.

A__line nodes sync to both PTPv2 and wordclock reference, and can even convert between them. Two redundant network interfaces utilize ST2022-7 Seamless Protection Switching (SPS) using two discrete network paths to ensure error-free stream delivery. With ample receive buffer capacity to meet ST2022-7 class C, redundant path differentials of up to 150ms are supported for WAN applications.

Additionally, the A__stage boxes provide true flexibility through a non-blocking routing matrix that allows any input to be routed to any output. In networked infrastructures they simplify level control by supplying ppm metering for all Analog and AES3 interfaces. The units integrate tightly with Lawo’s VSM IP Broadcast Control System and are designed to serve as IP audio stageboxes for mc² consoles, audio extensions for the V__matrix ecosystem, or as stand-alone IP audio gateways.

**KEY FEATURES**

- **WAN-capable Audio-over-IP nodes**
- **MIC, LINE, AES3, dual-redundant MADI & GPIO interfacing**
- **Discrete Class A Mic preamps**
- **ST2110-30/31, AES67 and RAVENNA support**
- **ST2022-7 class C streaming redundancy**
- **PTP / WordClock Sync and Sync format conversion**
- **Seamless integration with Lawo’s VSM IP Broadcast Control System and mc² audio production consoles**

**SPECS**

- **19" / 3 RU (A__stage48); 4 RU (A__stage64 & 80)**
- **Redundant internal PSU (100–240 Volts)**
- **Low noise design**

Get the complete A__line brochure here.
The Audio Solution for Automated News Production

What is it?
Universal Networked Audio Engine for Automated Production

What does it do?
The mc² Micro Core is a versatile standalone audio processing core, ideal for setting up automated news production workflows and similar applications. It’s basically an mc² in a box, allowing for audio production without an audio console. The mc² Micro Core provides 192 DSP channels and a routing matrix of 512x512 channels. It is connected to the outside world via four SMPTE2110 / RAVENNA/AES67 ports – if MADI connectivity is needed, this can be readily accomplished with A__madi4, Lawo’s bidirectional MADI-to-RAVENNA interface. The mc² Micro Core is designed to be controlled by all major automation systems. In addition, Lawo’s VisTool GUI building software provides remote control via fully customizable user interfaces incl. metering. The unit’s automated mixing assistants like AutoMix allow for fully automated audio mixing e.g. of news shows or off-tube commentary applications. Supported protocols and control systems include Lawo VSM, Evertz Magnum, GV Ignite, Ross Overdrive, Vizrt Mosart, Imagine Magplian, BFE KSC, Pharos, among others. The processing core provides extensive diagnostic tools for remote maintenance, including log files and http access, just like any mc² console. The compact 19”/3RU device uses inaudible, low-spinning fans for cooling.

KEY FEATURES
- Full mc² feature set on smallest footprint
- Renowned Lawo mc²-series processing quality
- Wide range of I/O devices available
- Automated Mixing Assistants
- Customized User Interfaces
- Compatible with all major automation systems
- SMPTE2110-30/AES67/RAVENNA Audio-over-IP connectivity

SPECS
- Dimensions (H x W x D): 133 mm (3 RU) x 483 mm (19") x 456 mm (18")
- Weight: 12.3 kg (27.1 lb)

Station Automation System

Customizable Touch Screen GUI
Radio never looked so good.

What is it?
The streamlined, intuitive AoIP mixing console optimized for today’s monitor-centric studio workflows.

What does it do?
Ruby pairs physical mixing controls with an intuitive, context-sensitive GUI that gives instant access to more advanced functions when needed. With familiar faders and switches close at hand and the power of advanced DSP tools and routing tools only a screen’s tap away, ruby gives your talent the tools they need to create easily, naturally, effortlessly — in the way that suits them best.

Today’s studio tasks are more computer-oriented than ever, and screens are everywhere. Operators are also busier than ever, with no margin for error. Ruby is fine-tuned to match the pace of today’s radio workplace, with physical and virtual controls that complement each other naturally. Multi-touch enabled, context-sensitive information displays let operators adjust settings quickly and easily; then instantly dock them to free screen space for other production tools. You can even create your own customized screens with amazingly powerful VisTool Unlimited GUI-builder software (optional). Motorized faders assume preset positions silently, while advanced automated functions, like AutoMix hands-free mixing and one-touch AutoGain mic gain control, leave talent free to create instead of fussing with faders.

ruby was designed from the ground up as a standards-based mixing console. There are hundreds of AES67 / RAVENNA and MADI channels built into its Power Core mixing engine. It’s also ST2110-30 compliant, for seamless interoperability in combined radio & TV facilities. And ST2022-7 Seamless Protection Switching protects against network interruptions. ruby is the ideal console for today — and for the future.

KEY FEATURES
- Flexible, intuitive design lets operators choose between physical and on-screen controls
- Windows™-based VisTool GUI builder lets you design custom multi-touch control screens
- Single-frame or split-frame, flush or counter-top mountings
- Standards-based AES67 / RAVENNA IP-Audio Networking, ST2022-7 network redundancy
- EZConfig setup wizard helps speed installation
- Dual-mode SmartSnap snapshots switch quickly between On-air and Production modes
- Stereo, mono and 5.1 mix outputs
- AutoMix hands-free mixing rides gain on multiple channels automatically
- One-button AutoGain optimizes microphone levels while talent talks

SPECS
- 4-fader to 16-fader frame sizes can combine for consoles of up to 60 faders
- Smooth 100mm motorized faders
- 96 input channels with full Lawo DSP capabilities
- Snapshots support up to 120 virtual faders
- 80 summing busses
- Works with Power Core AoIP mixing engine & modular I/O system

Get the complete ruby brochure here.
Power Core

AES67 Mixing Engine and I/O Gateway

What is it?
Power Core is a DSP mixing engine and modular I/O system for Lawo radio consoles. Thanks to its audio signal density, ability to handle diverse audio types and expandable audio capacity, broadcasters have also found Power Core to be the ideal gateway between legacy audio formats and standards-based AES67 IP media networks.

What does it do?
Power Core may be the most powerful audio signal processor ever made for broadcast. So powerful that as many as 96 channels of DSP input processing can be unlocked to use for anything from EQ to de-essing, from dynamics to delay synchronization. Used with Lawo’s ruby radio console, it’s a powerful mixing and routing engine. By itself, Power Core is an “über-node”, capable of ingestting massive amounts of audio for your AES67 AoIP network — with ST2110-30 compliance to assure seamless workflow in combined radio & TV facilities. In its most advanced configuration, Power Core’s massive DSP capabilities can be unlocked to apply audio correction to vast numbers of signals plant-wide. A variety of feature packs are available to tailor Power Core to any performance or budget requirement.

Power Core’s standard front-panel I/O includes SFP Ethernet ports capable of handling 128 bi-directional AES67 streams, and 4 ports dual-redundant MADI ports (128 channels of audio) — perfect for native MADI-to-AES67 AoIP conversion. Although this represents a staggering amount of standard I/O, more can quickly be added via the 8 rear-panel expansion slots, which accommodate a variety of analog, AES3, MADI and DANTE® I/O modules, all equipped with high-density DB-25 connectors. Autoswitching, dual-redundant power is standard, as is ST2022-7 Seamless Protection Switching, to ensure 24/7 uptime.

KEY FEATURES
- Standards-based AES67 / RAVENNA IP-Audio networking, ST2022-7 network redundancy
- AES67 and MADI I/O standard, expandable with multiple analog and digital I/O options
- 80 summing buses, configurable as Program, Record, Aux, Group, Mix-Minus (clean feed) or General Purpose. A full DSP channel with EQ, Dynamics and Delay functions may be applied to any of these buses (up to 16 stereo or 32 mono buses)
- 96 Signal Processing input channels, each with gain, signal presence indicator, direct out, Insert, fader, Aux send with Pre / Post switching, pan / balance, AutoGain for each Mic input
- Dynamics suite with gate, expander, compressor, limiter
- Up to 32 instances of AutoMix and / or De-Essing
- 4 individual AutoMix groups available allow creation of multiple independent mixes

SPECS
- 19" / 1RU
- Up to 1,960 x 1,960 routing matrix
- 4x MADI (each 64 channels I/O) with SFP cages (MADI ports 1/2 and 3/4 can be grouped as dual-redundant interfaces)
- 4x AES67 / RAVENNA with SFP cages (each 64 channels I/O, incl. redundancy)
- Eight expansion slots for additional 8x Mic/Line In, 8x Line In, 8x Line Out, 4x AES3 I/O, 2x MADI, 2x DANTE interface cards, or unique Studio I/O card with 2x Mic in, 2x HP out, 2x speaker out

NEW:
- DANTE® I/O Card with sample rate conversion
- Dual-redundant DANTE ports with 64 channels
- ST2110-30/-31 compliance for audio interchange compatibility with video networks
- ST2022-7 Seamless Protection Switching (SPS) provides network link redundancy for mission-critical applications

Get the complete Power Core information as part of the ruby brochure here.
**Virtual Radio Mixer**

**What is it?**
R\*LAY is a Virtual Radio Mixer. Using technology from the IT world, we’ve created a professional radio mixing console that runs in a virtualized PC environment, with an intuitive, multitouch-enabled graphical interface.

**What does it do?**
With R\*LAY, you can build an entire broadcast studio on a single touchscreen PC. R\*LAY isn’t just a soft mixer: it’s a virtual mixing environment. Its modular approach lets you choose the tools you need from a variety of readily available broadcast software – remote codecs, VoIP phone systems, sophisticated audio processors, virtual patch bays, streaming encoders and more – by leveraging the immense power of today’s COTS computing platforms. Host a full-featured 8-fader Virtual Mixing Console on the same PC that hosts your playout system and other studio tools — or install R\*LAY on a multi-touch laptop and pair it with a portable OnAir 4 audio interface to create a complete OB kit that fits in a backpack. A 4-fader version is also available; perfect for production stations or journalist kits.

R\*LAY’s clean, intuitive multi-touch interface gives talent a familiar, console-like mixing interface that accommodates all of today’s diverse audio types: AES67 / RAVENNA streams, analog or digital inputs, and PC software with WDM or ASIO interfaces. You’ll find it’s perfect for news bullpens, editing suites, remote broadcasts, backup facilities or disaster recovery. And with SMPTE ST2022-7 Seamless Protection Switching to help ensure network stream delivery, R\*LAY is more than just a touchscreen mixer — it’s enterprise-class virtual mixing software.

**KEY FEATURES**
- AES67 / RAVENNA compliant
- ST2022-7 Network Redundancy
- Runs on standard touchscreen computers
- Supports native PC audio (ASIO, WASAPI, WDM, MME)
- Mix-minus (clean feed) with Talkback
- 4-fader & 8-fader versions
- Use PC audio or pair with Lawo A__OnAir4 stereo interface for professional quality mic, line & digital I/O
- GPIO via open-source Ember+ control protocol
- On-screen control of user shortcuts, mixer snapshots, GPIO functions and more

**SPECS**
- Intel Core-i Series Quad Core CPU or better; 4 GB RAM; 2 GB available hard drive space; Gigabit LAN network interface; Windows 7 or 8 (32 or 64 Bit), or higher

Get the complete R\*LAY VRX+ Bundle brochure here.
**PRODUCTS for BROADCAST**

### RADIO CONSOLES
- **radio** Radio Broadcast Console
- **sapphire** Radio Broadcast Console
- **ruby** Radio Broadcast Console
- **crystal** Radio Broadcast Console

### AUDIO CONSOLES
- **mc2 96** Grand Production Console
- **mc2 Compact I/O** Audio Stagebox
- **mc2 56** Audio Production Console
- **mc2 36** Audio One Audio Production Console
- **mc2 56** Audio Production Console
- **mc2 96** Grand Production Console

### AUDIO PROCESSING, ROUTING & I/O
- **Power Core** High-density modular IP DSP Audio Node
- **Compact Engine** remoteLink Audio Node with Onboard DSP
- **Novo 17** Modular Audio Node with Onboard DSP

### AUDIO PRODUCTION CONSOLES
- **mc2 96** Audio One Audio Production Console
- **mc2 56** Audio Production Console
- **mc2 36** Audio Production Console
- **mc2 56** Audio Production Console
- **mc2 96** Grand Production Console

### IP AUDIO AGGREGATION & PROCESSING
- **Nova 33** Modulo IP/DSP Audio Node & Console Core
- **Nova 33 compact** Modulo IP/DSP Audio Node & Console Core
- **Nova 73** Modulo IP/DSP Audio Node & Console Core
- **A.UHD Core** Ultra-High Density Network DSP Engine
- **mc2 MicroCore** IP Audio I/O & DSP Node for Automated Production

### IP AUDIO I/O
- **mca2** Modular Audio System
- **mca2 Compact I/O** Audio Interface
- **A.IW** Audio Interface Baseband Interface

### VIDEO PROCESSING
- **V/*matrix** System-defined IP Routing & Processing Platform
- **V/*matrix cmx66 alike** System-defined IP Routing & Processing Platform
- **V/*matrix cmx77 alike** System-defined IP Routing & Processing Platform
- **V/*matrix cmx77 alike 24-4** System-defined IP Routing & Processing Platform
- **V/*matrix cmx77 alike 256-8** System-defined IP Routing & Processing Platform

### MULTIVIEWERS
- **V/*matrix: vm_dmv** 4K / HDR Distributed Multiviewer App
- **V/*matrix: vms_dmv** 4K / HDR Multiviewer App
- **V/*matrix: vm_streaming** 4K / HDR Gateway and A/V Processing App
- **V/*matrix: vm_udx** 4K Up / Down / Cross Conversion App

### BROADCAST NETWORK MONITORING
- **smartDASH** Broadcast Network Monitoring System
- **smartSCOPE** Broadcast Network Monitoring System

### BROADCAST CONTROL & IP ORCHESTRAION
- **vsm** VSM The IP Broadcast Control System
- **vsmSOU** VsmSOU Video Orchestration and Unification Layer
- **vsmFALL** VsmFALL Multi-Studio Tally Management System

### COMMENTARY
- **LCU** IP-based Commentary Unit
- **LCC** Live Commentary Control Software

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Seamless control and monitoring of Lawo and 3rd party devices:
- Cisco & Arista networks
- IP nodes
- IP cameras
- Baseband equipment