Broadcast 3.0

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.
**What is it?**

Virtual module (app) for the V__matrix eco-system providing format conversion between SD, HD and 4K/UHD formats with audio embedding/de-embedding, frame-sync, HDR to SDR color space conversion and color correction.

**What does it do?**

The vm_udx app provides four independent paths of format conversion between SD, HD and 4K/UHD for IP and/or SDI signals. Conversion between SD and HD formats use one path while conversion to/from 4K uses four paths. Each path provides video framesync and audio sample rate conversion as well as audio delay functionality. Every path also has full audio embedding/de-embedding capabilities with audio gain and shuffling. Broadcast quality RGB and YUV color correction is provided for every processing path.

With the +HDR option the vm_udx app gets 4 instances of SDR<->HDR color space conversion using 3D LUTs. A large selection of LUTs developed by the BBC especially for live production are included and users are able to upload their own custom LUTs as well. The included LUTs allow for conversion between SDR and HDR in HLG and PQ.

Fundamentally designed with IP networking in mind vm_udx natively supports both ST2022-6 and ST2110-20 IP video as well as ST2110-30/RAVENNA/AES67 IP audio streams. Conversion between IP video and IP audio standards is also possible, e.g. ST2022 to ST2110. To ensure high availability ST2022-7 seamless protection switching (SPS) is natively supported. With the available io_bnc rear-plates vm_udx allows for legacy connection to SD-, HD- and 4K-SDI. Both single-link 12G-SDI as well as quad-link (2SI) is supported as is the ability to convert between single-link and quad-link.

**NEW: +4UDX license option**

An additional four independent paths can be added, bringing the total amount of Up/Down/Cross processing per C100 up to eight.

**KEY FEATURES**

- 4 instances of Up/DownCross conversion between SD/HDX (1 instance when converting to/from 4K)
- Additional 4 instances available with the +4UDX option
- HDR<->SDR conversion with the +HDR option
- Audio mono matrix with full audio embedding/de-embedding and shuffling between SDI/IP, IP/IP and SDI/SDI with SMPTE2110-30/RAVENNA/AES67
- Includes framesync and RGB/YUV color correction
- 4K 12G-SDI single-link inputs/outputs when combined with the io_bnc rear-plates
- Built-in programming, configuration and streaming telemetry capabilities
- Simple management and control through Lawo vsmStudio makes operation imperceptible from a traditional baseband environment while maintaining all of the benefits of an IP system

**NEW FEATURES**

- +HDR Option
  - The new +HDR option adds professional quality High Dynamic Range (HDR) to Standard Dynamic Range (SDR) conversion using 3D LUTs for both HLG and PQ formats to the vm_udx app.

**NEW: +HDR Option**

The new +HDR option adds professional quality High Dynamic Range (HDR) to Standard Dynamic Range (SDR) conversion using 3D LUTs for both HLG and PQ formats to the vm_udx app.
What is it?
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.

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.

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”

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/HEVC decoding for any Lawo LiveView™ enabled product.

Infinitely expandable. If you need to add more heads or more sources, just add another C100 with a vm_dmv app installed...
What is it?
Ultra low-noise frame for V__matrix C100 processing blade.

What does it do?
While the V__matrix C100 core processing unit was designed for data center and equipment room environments, there are applications where one might want to install some processing cores in noise-sensitive places such as control rooms or audio booths. The new V__matrix Silent Frame addresses this type of applications.

With two slots for standard C100 cores and any accompanying rear-plates the Silent Frame is so quiet that it might even impress an audio engineer. The unit's large, low-spinning fans are temperature controlled and provide the same front to rear cooling air flow as standard V__matrix frames – but whisper-silent.

As the V__matrix Silent Frame houses the same C100 processing blades like all other V__matrix frames, it has immediate access to any of the V__matrix apps already available including vm_streaming, vm_dmv and vm_udx.

The current line-up of V__matrix Virtual Modules includes the following apps and options:

- Whisper-quiet IP Streaming, Processing & Multi-viewing
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.

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 applications, e.g., virtualized playout channels. smartDASH supports solutions based on AWS, Google, VM Networks and others.

NEW: smartDS
smartDASH is a complementary data storage solution to smartDASH, allowing smartDASH users to log and store months of alarm and data logs for a historical record of everything from network bandwidth to signal loss. smartDASH’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.

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
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, A52

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.
**VSM**

Virtual Studio Manager

**What is it?**
IP Broadcast Control and Workflow Solution

**What does it do?**
VSM (Virtual Studio Manager) is a vendor independent Broadcast Control System and custom workflow solution that runs on an IP backbone and integrates easily with the majority of the most popular broadcast equipment on the market. These include IP edge devices and network infrastructures as well as traditional video routers, video switchers, audio routers, audio consoles, multiviewers, intercoms, modular equipment and other third-party devices. Equipment from different manufacturers can be seamlessly “glued” together, giving unmatched recall and logic control possibilities on top of a scalable TCP/IP backbone with a strong redundancy concept. Operators can control their production facility intuitively through highly customizable touchscreen-optimized software panels and a wide range of hardware LCD button panels, giving them the freedom of individual workflows. Advanced features such as dynamic resource management (pooling), Tally management (vsmTally), Boxing, dynamic tieline management, virtual devices and lots more set the benchmark for IP broadcast control systems.

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**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 GUIs
- Users can easily deploy their individual workflows and production setups

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**NEW FEATURES**

- 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 adaptations. You will get the job done faster, and changing layouts whenever necessary becomes a breeze.
- 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.
- 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.

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Get the complete VSM brochure here.
**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.

**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?
mc² 96 Grand Production Console with increased fader count

What does it do?
The new Xtra Fader Version of Lawo’s mc² 96 Grand Production Console features an increased fader count in the Central Control Section. 16 instead of eight faders in the Sweet Spot provide direct access to the double number of channels in the ideal listening position. Audio engineers enjoy an enhanced freedom for a more convenient and flexible workflow in fine-tuning their audio production settings. The mc² 96 console’s Xtra Fader Version is dedicated to demanding studio and OB truck applications with a need for maximum number of faders on a small footprint. The mc² 96 Xtra Fader Version allows to apply a 112-fader frame that fits into a mere 2,350 millimetres (92.52 inches) width for the installation in standard-sized OB vans (truck crosswise).

Get the complete mc² 96 brochure here.
mc²56 Dual Fader

A Global Standard
Re-defined

What is it?
Audio production console for mobile production, broadcast studies, performing arts and recording

What does it do?
This 3rd generation of the mc²56 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.

BIG FADER COUNT, SMALL FOOTPRINT: THIRD-GENERATION LAWO mc²56 PRODUCTION CONSOLE INTRODUCES DUAL FADER BAY

The dual-fader operating surface increases the console’s fader count while keeping its size small. With this new option, the mc²56 becomes the only audio mixing console in its category that can host up to 144 faders within such a compact footprint. The dual fader option also incorporates Lawo’s revolutionary LiveView™ thumbnail previews of associated video streams directly in the fader strips, enhancing user accuracy in fast-paced production situations.

+++ NEW FEATURES +++

Lawo mc²56 Production Console Intros Dual Fader Bay

The newest version of the mc²56 incorporates several groundbreaking features from Lawo’s mc²96 flagship console without sacrificing the identity of its predecessors – retaining virtues like compact size, flexibility and versatile design for applications ranging from broadcast trucks and studios to live performance and recording.

For optimized performance within IP video production environments, there is full support for native SMPTE ST2110, AES67/RAVENNA and DANTE®, while Lawo’s revolutionary LiveView™ feature enables thumbnail previews of video streams directly in the fader labeling displays. Best-in-class performance in networking applications has been taken to the next level with the addition of unique capabilities such as IP-Share™ gain compensation and DSCA™ Dynamic Surface-to-Core Allocation. All of this and more, simply reinforces this console’s place as the number one choice within complex IP-based production infrastructures.

KEY FEATURES

- Perfect for outside broadcast vehicle dimensions: 64 faders fit across most standard installations.
- 21.5” super-precise full HD touch-screens, color-TPF’s 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
- Automated mixing assistants incl. Automix, Audio-follow-Video, Downmix, AMBIT Upmix and KICK 2.0
- 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

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.

The UHD Core features 1,024 Lawo-grade DSP channels on 1RU (512 channels in 96kHz mode) IP network processor based on open standards (ST2110-30/-31, AES67, RAVENNA)

Full redundancy: SPS stream redundancy (ST2022-7) with 8x 10GBase capable independent SFP network interfaces plus hardware redundancy via hot-spare redundancy unit

Sub-millisecond network latency via special high-performance RAVENNA profile

DSP resources shareable amongst up to four consoles

Scalable DSP performance via licensing system

Designed for mc² 56 and mc² 96 consoles

Futureproof, software-defined hardware – this is just the beginning...

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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.
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.

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 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" / 1RU (A__madi6), 19" / 3 RU (A__digital64 & A__stage48)
- 4 RU (A__stage64 & 80)
- Redundant internal PSU (100–240 Volts)
- Low noise design

+++ NEW PRODUCTS +++
New A__line Audio-over-IP Nodes
A__madi6 and A__digital64 join Lawo’s A__line series of WAN-capable Audio-over-IP nodes.

A__madi6 bundles three independent MADI-IP bridges onto a single RU, powered by an A__line default redundant power supply. Each bridge features two bi-directional MADI ports on SFP and two Dual Media Ethernet ports for streaming and control.

A__digital64 node supports 32 AES inputs, including Sample Rate Conversion and 32 AES Out in a 3RU footprint. An additional redundant pair of MADI ports and two Dual Media streaming ports with support for WAN-compatible ST 2022-7 class C seamless protection switching complete the audio IO. With ST2110-10 compliant PTP clocking support, additional wordlock IO, GPIO and a dedicated management port, the A__digital64 integrates seamlessly into both hybrid and IP-centric broadcast installations.
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__madi6, 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 Magiplan, 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
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.

Get the complete ruby brochure here.

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.
Power Core MAX

AES67 Mixing Engine and I/O Gateway

**What is it?**
Power Core MAX is the AES67 mixing engine that can support up to four independent Lawo mixing surfaces. It leverages the capabilities of Power Core, the world’s most power-packed DSP mixing engine, to pack more capability, flexibility and value into 1 RU than ever before.

**What does it do?**
Power Core may be the most potent audio signal processor ever made for broadcast. So powerful, in fact, that a single Power Core can support as many as 96 faders, 96 DSP channels, 80 summing busses and much more. The new Power Core MAX lets you take advantage of these amazing resources to support multiple mixers – two, three, or even four mixers (depending upon intended use) – a unique ability perfectly suited to today's multi-studio radio facilities.

All of Power Core's advanced features are available: EQ, de-essing, dynamics, AutoMix, even delay synchronization. Used with Lawo's mixing controllers, like the award-winning ruby control surface, or VisTool virtual mixer, it's a powerful mixing and routing engine.

And, 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. Standard front-panel I/O includes dual-redundant SFP Ethernet (128 bi-directional AES67 streams), and 4 dual-redundant MADI ports (128 audio channels) — perfect for native MADI-to-AES67 AoIP conversion.

Even more I/O can quickly be added via 8 rear-panel expansion slots: analog, AES3, MADI and DANTE® I/O modules are available. ST2110-30 compliance assures seamless workflow in combined radio & TV facilities, while aut_switc_hing, dual-redundant power and ST2022-7 Seamless Protection Switching help ensure continuous uptime.

In addition to the new MAX package, Power Core customers can choose from three other license packages, tailored to a variety of operational needs and price points.

**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
- Up to 32 instances of AutoMix and / or De-Essing

**ST2110-30/31 compliance assures seamless workflow in combined radio & TV facilities, while aut_switc_hing, dual-redundant power and ST2022-7 Seamless Protection Switching help ensure continuous uptime.

Get the complete Power Core information as part of the ruby brochure here.

+++ NEW FEATURES +++

- MAX package supports two, three or even four mixing surfaces per Power Core
- 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.

**SPECS**

- 19” / 1RU
- 1960 x 1960 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)
- 2x 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 Out, 2x MADI, 2x DANTE interface cards, or unique Studio I/O card with 2x Mic in, 2x HP out, 2x speaker out.

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Get the complete Power Core information as part of the ruby brochure here.
AES67 Stream Monitor

**What is it?**
The world’s first true AES67 monitoring & inspection tool.

**What does it do?**
Modern broadcasting facilities have embraced the AES67 standard for AoIP networking. Lawo AES67 Stream Monitor keeps you informed of the state of your most critical program streams with real-time confidence metering, LUFS metering, silence sense, audio level alerts and more – perfect for system diagnostics or Master Control monitoring.

AES67 Stream Monitor displays information at-a-glance for up to 16 streams containing multiple channels of audio. Average and peak bargraph meters for each channel are onscreen constantly, accompanied by a level readout in Loudness Units, and visual alarm indications for under-level, over-level and error states.

Clicking on any stream display populates multiple information tabs, enabling users to see statistics for stream jitter, packet loss and data errors, view loudness over time, and, if dual network connections (ST2022-7) are present, compare statistics from both NICs. A detailed SDP information tab allows SDP data to be copied and pasted into other system applications. You can even audition streams using your PC’s built-in audio card. AES67 Stream Monitor is the AoIP diagnostic tool you’ve been waiting for.

**KEY FEATURES**
- Monitors 16 critical AES67 streams with up to 128 total audio channels per instance.
- Tracks loudness compliance and stream health over time.
- Under- and over-level loudness alarms, with user-definable alert thresholds per stream.
- Downloadable file-based error report.
- ST2022-7 compatibility allows monitoring dual-redundant network connections.
- Works on Windows® 10 PCs.
- VMWare compatible for multi-instance deployment on virtual machines.
- Essential tool for system installers, integrators, facility engineers.

**SPECS**
- Runs on 64-bit Windows® 10 operating system.
- 16:9 monitor with full-HD screen resolution or better required.
- 1x Gigabit LAN connection required.
- ST2022-7 monitoring requires second Gigabit NIC.
What is it?
A toolkit for building custom tablet-based graphical control panels.

What does it do?
VisTool Solo is a lightweight GUI-builder software that lets you create graphical software control panels for producers, studio guests, hosts and operators. VisTool Solo turns Windows™ tablets into powerful touchscreen talent panels.

VisTool Solo provides a vector-based GUI toolkit filled with buttons, meters, switches, faders, timers and other objects you can use to build touch-sensitive control panels. No more hardwired button panels needed — VisTool Solo gives talent a responsive, intuitive control environment that helps shows run smoother.

Imagine the possibilities: custom panels for hosts, guests, producers and news stations, with the tools they need most. Headphone source selection. Mic on/off buttons. Programmable event timers. Countdown clocks. Even virtual faders for peripheral audio devices. Engineers can use VisTool Solo too — it’s perfect for remote control of routing crosspoints, loudness meters, silence alarms and more.

VisTool Solo is portable, too: with a Wi-Fi connection, talent can take controls with them to wherever they work. No wires to run or cables to snag — just easy, intuitive control.

KEY FEATURES
- Works on common Windows-based touchscreen tablets
- Design custom, single-page multi-touch control screens
- Use supplied vector graphic library, or import your own custom graphics
- Easily connects to AES67 studio network via LAN or Wi-Fi
- Build personal control pages tailored to talent’s unique workflows
- Perfect for building confidence displays, meter panels or routing controllers

SPECS
- Runs on Windows tablets meeting these specs:
  - Intel Atom® x5 4-core CPU with benchmark of 1,250 points minimum (based on mobile GPU ratings from www.cpubenchmark.net)
  - 16:9 full-HD screen resolution or better recommended
  - 64-bit Windows® 10 operating system
  - 4GB RAM or greater
  - 64GB storage or greater
  - Network connection (LAN or Wi-Fi)
### PRODUCTS for BR

**PRODUCTS for BROADCAST³.0**

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<th>CATEGORY</th>
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| **RADIO CONSOLES** | - Crystal Radio Broadcast Console
- Ruby Radio Broadcast Console
- Sapphire Radio Broadcast Console |
| **AUDIO PRODUCTION CONSOLES** | - mc² 36 Audio-Line Audio Production Console
- mc² 56 Audio Production Console
- mc² 96 Grand Production Console |
| **IP VIDEO INFRASTRUCTURE** | - V-matrix Seamless Signaled IP Routing & Processing Platform
- V-matrix vm_streaming 4K/HD Gateway & AV Processing App
- V-matrix vm_udx 4K Up / Down / Cross Conversion App |
| **BROADCAST CONTROL & IP ORCHESTRATION** | - VSM The IP Broadcast Control System
- VsmSOUL Broadcast Orchestration and Unification Layer
- VsmPANEL Customizable Software Control Panel |
| **AUDIO PROCESSING, ROUTING & I/O** | - Power Core Ultra-high Density Network DSP Engine
- Nova17 Modular Audio Node with Onboard DSP
- Nova25 Modular Audio Node with Onboard DSP |
| **IP AUDIO AGGREGATION & PROCESSING** | - Nova37 Modular Audio Node with Onboard DSP
- Nova73 Compact Module [MC] Audio Router and Console Core
- Nova73HD High-Density Modular Audio Router and Console Core |
| **MULTIVIEWERS** | - V-matrix: vm_mv 4K / HDR Multiviewer App
- V-matrix: vm_dmv 4K / HDR Distributed Multiviewer App
- V-matrix: vm_streaming 4K / HDR Gateway & AV Processing App
- V-matrix: vm_udx 4K Up / Down / Cross Conversion App |
| **BROADCAST NETWORK MONITORING** | - smartDASH Broadcast Network Monitoring System
- smartSCOPE Packet Inspector Media & Network Analyzer |
| **RADIO SOFTWARE** | - VisTool PC-based GUI Building Software
- R-LAY VPE PC-based Virtual Audio DSP Engine
- R-LAY VSC PC-based Virtual Sound Card |
| **IP AUDIO I/O** | - DALLIS Modular IO System
- mc² Compact IO Audio Interface
- A-IHE With Replaceable Audio I/O Interfaces |
| **VIDEO PROCESSING** | - V-matrix Seamless Signaled IP Routing & Processing Platform
- V-matrix vm_streaming 4K/HD Gateway & AV Processing App
- V-matrix vm_udx 4K Up / Down / Cross Conversion App |
| **COMMENTARY** | - LCU IP-based Commentary Unit
- LCC Studio Commentary Control Software |

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