MoIN
Software

MoIN Datasheet

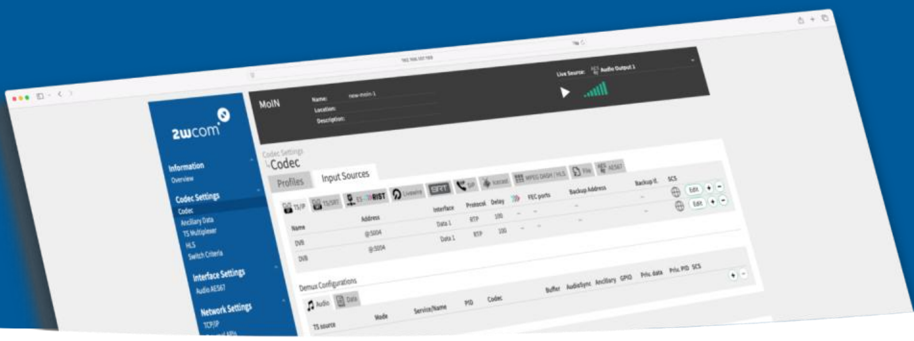
Highlights

- Multi-channel virtual audio over IP encoder/decoder platform
- Supports all common standards, protocols and audio codecs
- Broadcast reliability with modern IT flexibility

Your audio. Our solution.

2wcom

MoIN



MoIN – Multimedia over IP Network – is a turnkey software platform for professional audio contribution and distribution. It seamlessly encodes, decodes and transcodes multiple audio channels in parallel. Designed for broadcasters and media organizations, MoIN can be deployed on dedicated hardware, virtual machines or in the cloud, providing a reliable and scalable solution without the complexity of building and operating your own software platform.

MoIN. The one word that works with everyone.

In northern Germany, "Moin" is the greeting that fits every person at every hour of the day. Our platform is built on the same idea: one system that speaks to every codec, every format and every protocol in your network 24/7 – no matter who is at the other end.

19 audio coding formats

Up to 24 channels per vCPU

Latency starting at 50 ms

Key Benefits

- **Turnkey Solution:** software, operating system, management tools and support in one package – your team works on audio, not on infrastructure.
- **Hybrid by design:** Combination of MoIN and 2wcom hardware products as a comprehensive and reliable solution.
- **One software instead of point solutions:** contribution, STL, distribution, web streaming and DAB+/μMPX in one solution – one operating concept, one support contact.
- **A translator, not just a transporter:** parallel encoding, decoding and transcoding.
- **Comprehensive codec support:** unmatched interoperability with 18 codecs included, even more available as upgrade options.
- **Interoperability down to the clock:** NTP, PTPv2 and sample rate conversion combine sources with different sample rates and clock domains in one channel.
- **Robustness without a lock-in decision:** PRO MPEG FEC, dual streaming, SRT and RIST in one platform – choose the protection per link.

Main Features

Flexible Software Solution

- Software solution for audio-over-IP distribution
- Interoperability with all 2wcom hardware products
- Support of all common audio coding formats for contribution, distribution and streaming
- Multiple deployment methods to ensure the best integration into already existing broadcast environments
- Various features like list view and SIP call manager included
- Upgradeable with software options according to your needs
- Support via SLA available

Control Center

- Single pane of glass (SPOG): unified dashboard for all 2wcom software applications in a single centralized view
- Allows orchestration, monitoring and configuration of all integrated 2wcom software
- Includes network settings, NTP and PTP configuration and license management

Installation Methods

- Installation via cloud-init image with a Proxmox or VMware ESXi server
- Installation as a standalone application on a Linux server using Docker
  Note: The validated reference environment is Ubuntu 24.04 LTS or higher with Docker v29.0 or higher. Other Linux distributions may also be compatible, provided they support the required Docker version.

Versatile Compatibility – Complete Audio Codec Support

- Adaptability for audio contribution in varied network conditions with codecs like Opus, Enhanced aptX (E-aptX) and xHE-AAC.
- Optimized studio-to-transmitter links (STL) with efficient coding formats like MPEG-Layer 2 and AAC-LC; ensure highest fidelity and minimal latency with bit-transparent transmission of digital audio
- Professional audio streaming: Efficient generation of web streams supporting mp3, OGG, Opus and the complete AAC family
- Specialized broadcasting support: DAB+ encoding* and µMPX* as upgrade options

*optional

MoIN

Smooth Integration – Compatible with various Audio over IP Formats

- 2wcom provides robust network connections by combining multiple proven technologies to deliver your audio even in stressful conditions.
- Unicast, multiple unicast, and multicast for maximum network control
- Proven multimedia transport protocols such as UDP and RTP
- Multiplex and de-multiplex from MPEG-TS over IP
- Well-established audio protocols such as AES67, SMPTE ST 2110, Livewire and RAVENNA*
- HLS* or Icecast source client for web streaming

Advanced Streaming Robustness – Unmatched Broadcast Resilience

- PRO MPEG Forward Error Correction (FEC) and dual streaming for resilient, redundant streams
- Secure and reliable streaming over unpredictable networks with Secure Reliable Transport (SRT)* and Reliable Internet Stream Transport (RIST)
- Manage packet size, buffering, and Quality of Service (QoS) for a robust streaming performance
- Multiple redundancy options and source switching for uninterrupted streaming including Dual Streaming and Stream4Sure

Audio Latency Management – Ensuring Precise Synchronization

- Uses Network Time Protocol (NTP) or PTPv2 to synchronize audio input and output across devices
- Handles all audio signals, also based on different clocks
- Sample rate converter (SRC) to combine different connections with different sample rates or different clocks

Smart Management and Seamless Integration

- Well-established APIs and physical control seamlessly integrate into your current infrastructure: Rest API, Ember+, SNMP and NMOS
- Stay informed: Flexibly configurable alarm events and notifications over SNMP

Verified IP Security

- High-level security within open IP infrastructures
- Thoroughly examined by independent audit authorities through whitelist/blacklist penetration tests

Specifications

Formats and Protocols

Audio

Codecs (included)	Linear PCM, G.711, G.722 Opus, Ogg Vorbis MPEG 1/2 Layer 2, 3 MPEG-2/MPEG-4 AAC-LC, MPEG-4 HE-AAC v1 & v2, MPEG-4/MPEG-D xHE-AAC MPEG-4 AAC-LD/ELD/ELDv2 Enhanced aptX (E-aptX), FLAC
Codecs (optional)	µMPX Fraunhofer DAB+ (HE-AAC v2, ETSI TS 102 563) APTmpX
Bit depth	16, 20, 24 bit
Sample rates	16, 22.05, 24, 32, 44.1, 48 kHz (On request: up to 192 kHz)

Streaming

IP protocols	TCP, UDP unicast, multiple unicast & multicast
Audio over IP formats	RTP (RFC 3550, RFC 3551, RFC 3640, RFC 2250) SIP/SAP according to EBU Tech 3326 SMPTE ST 2110 (optional) AES67 based on RAVENNA, Livewire, or Dante (optional) Livewire / RAVENNA (SAP/SDP, RTSP, AES67, PTP, PTPv2) (optional) MPEG-TS
Web streaming	Icecast/Shoutcast Client Icecast Source Client Icecast Server HLS encoder (optional) HLS decoder (optional)
Transmission robustness	SRT, RIST Pro-MPEG FEC #3 release 2 Dual Streaming µMPX FEC Adaptive bitrate switching, source switching concept, management of packet size, buffers, QoS
Latency	Starting at 50 ms depending on the selected codec and buffer settings

Specifications

Synchronization

Internal	free-running
External	PTP, PTPv2, NTP
Decoder synchronization between different devices	< 20 ms using SPN via NTP (optional) SFN passthrough (optional)
Sample rate converter	Asynchronous, any ratio

Data Transmission

Audio streams (in)	up to 512 (more on request), depends on CPU and memory MoIN Solo: up to 8, depends on CPU and memory
Audio streams (out)	Up to 512 (more on request), depends on CPU and memory MoIN Solo: up to 8, depends on CPU and memory
UECP (RDS)	Linked/synchronous to the audio

Control and Monitoring

Interfaces/Protocols (APIs)	Rest API, Ember+, SNMP, NMOS
Configuration	via web interface and APIs
Live Listening	via Icecast
Alarms	SNMP traps, web interface and log messages
RDS/UECP	Realtime decoder

Minimum Server Requirements

Server not included (can be chosen by the customer)

Processor power (min.)	4 CPU cores
Memory	8 GB RAM
SSD	50 GB
Network	1 port with 1 GB

Additional hardware interfaces (on request)

AES/EBU/Analog	Analog/digital audio in/out
Madi	Multi-channel audio digital interface (available as internal or external extension)
Dante	AES67 from Audinate (available as internal and external extension)
GPIO	Relays/Contact closures

Options

Deployment Methods

MoIN is available for 3 different deployment methods. This enables you to choose the deployment that fits your existing infrastructure the best. The table below shows the differences between the three deployment methods: MoIN as a container, MoIN installed with ISO and MoIN as an appliance.

	Container	ISO	Appliance
Hardware	–	–	●
Operating system	–	●	●
Control Center	●	●	●
Setup Wizard	●	●	●
Management UI	●	●	●
Managed updates	●	●	●
Network configuration	●*	●	●

Alternative Integration Approach

If the deployment methods listed above do not fit your requirements and you would like to integrate MoIN into your existing infrastructure, we offer the MoIN Integration Toolkit.

The *MoIN Integration Toolkit* is a modular audio-over-IP software framework that enables developers to integrate professional audio contribution, distribution and streaming directly into their own products and platforms. Based on the proven MoIN technology, it provides modular components, open APIs and integration support for maximum flexibility, automation and control. It is the ideal solution for system integrators, OEMs, infrastructure providers and enterprise customers building their own broadcast and media platforms.

If you have any further questions, please do not hesitate to *contact us*. We are happy to help you find the MoIN deployment that works best for your requirements and infrastructure.

*Requires netplan to be available on the host system.

Options

Editions

Please choose one of the following editions. Even after you purchase a MoIN Edition, you can change the edition according to your requirements. Alternatively, you can customize your MoIN with MoIN Flex (for more information see 3.4 Software Options).

Feature	Basic	Advanced	Tonmeister	FLEX
Audio Channel	1 encoder/ transcoder per channel no decoder redundancy	1 encoder/ transcoder per channel 1 decoder with 3 backup instances per channel	1 encoder/ transcoder per channel 1 decoder with 3 backup instances per channel	1 encoder/ transcoder per channel 1 decoder with 3 backup instances per channel
Audio Codecs*	●	●	●	●
Ancillary Data*	●	●	●	●
Ravenna, AES67, PTP	● (input only)	●	●	●
Icecast Server and Source Client	●	●	●	●
Livewire+	●	●	●	●
Live Listening	●	●	●	●
EBU Tech 3326		●	●	●
SRT/RIST en-/decoder		●	●	●
SPN (Synchronous Playout Network)		●	●	●
HLS en-/decoder		●	●	●
Flexible backup switching		●	●	●
MPEG-2 TS en-/decoder			●	●
MPE			●	●
Stream4Sure			●	●
TS forwarding			●	●
192 kHz			●	●
Mixer			●	●
Loudness Measurement			●	●

*Included in every edition. For a full list of all codecs see Specifications.

Options

Upgrade Options

Additionally to the options connected to the MoIN editions, there are even more upgrade options that can be separately added to your MoIN application. These upgrade options are listed below:

Article no.	Name	Description
VER43123	µMPX encoder –MPX compression	Algorithm to compress the full MPX/composite signal including pilot and RDS to IP. 5 available bitrates: 320 – 800 kbps. Price per encoder.
VER43124	µMPX decoder –MPX decompression	Algorithm to decompress the full MPX/composite signal including pilot and RDS to IP. 5 available bitrates: 320 – 800 kbps. Price for activation as µMPX decoder. Price per decoder.
VER43125	DAB+ encoder license	Fraunhofer Professional DAB+ Codec with EDI/STI-D output. • Local insertion of PAD (DLS, SLS), TA, PTY • Per DAB+ subchannel instance
VER43127	Fraunhofer MuxEnc license	For connection to Fraunhofer DAB ContentServer. With central configuration, PAD insertion and monitoring of the DAB/DAB+ encoders at the multiplexer.
VER43134	THIMEO Stereo Tool sound processing (FM professional)	Sound processing by THIMEO stereo Tool including Declipper and Advanced Dynamics. • Stereo/RDS/RBDS generation • Multiband sound processing with advanced dynamics (AGC, compressors, etc.) • Audio and FM clipper • µMPX encoder

Options

Software Options

For the MoIN Flex edition, please select the options you want to include:

Article no.	Name	Description
VER43115	HLS en-/decoder	HLS streaming protocol en-/decoder functionality for adaptive bitrate streaming.
VER43113	SRT/RIST en-/decoder	SRT functionality for en-/decoder according to SRT standard of the SRT Alliance (including UDP). RIST functionality for en-/decoder according to IETF standard "RIST Simple Profile" and RFC 4585.
VER43112	MPEG-2 TS en-/decoder	EN-/decoding of a MPEG-2 TS (transport stream) according to ISO/IEC 13818-1 or ITU-T Rec. H.222.0.
VER43130	MPE	MPE (Multiprotocol Encapsulation) encoding/decoding.
VER43131	TS forwarding	TS Forwarding enables the forwarding of a complete TS or MPE-forwarding (requires MPE Option)
VER43129	SPN (Synchronous Playout Network)	Output synchronization via NTP time server with an accuracy of 20 ms.
VER43117	Ravenna, AES67, PTP	According to the standard Ravenna of audio over IP interoperability (including AES67, SAP, RTSP, PTP)
VER43114	Icecast Server and Source Client	Audio web stream server (supports various audio codecs such as MP3, Ogg Vorbis and AAC)
VER43116	Stream4Sure – Quad streaming with different coding and quality	Simultaneous transmission/reception of up to four IP streams of different coding and quality and seamless exchange of audio samples in case of failure.
VER43118	EBU Tech 3326	According to the standard of audio over IP interoperability EBU Tech 3326 (including SDP, SIP, SIP phonebook, 2wcom Easy2connect).
VER43119	Livewire+	According to the standard of audio over IP interoperability Livewire+ (including AES67, LWRP, LWCP, Livewire Advertisement).
VER43128	Loudness measurement	Loudness measurement for the audio outputs according to EBU R128 and ITU-R BS.1770/BS.1771. Measures and displays momentary/short-term/integrated loudness, loudness range (LRA) and True Peak (dBTP).
VER43132	Mixer	Integrated audio mixer supporting mixing of up to 5 audio channels.
VER43133	192 kHz	High-resolution audio support up to 192 kHz sample rate.