MoIN

Release Notes

Version 2.8.0

31.03.2026

2wcom Systems GmbH – Am Sophienhof 8 – 24941 Flensburg – Germany
+49 461 662830-0 (Fax11) – contact@2wcom.com – www.2wcom.com

Your audio. Our solution.

Table of Contents

Version 2.8.0

31.03.2026

New Functionality

- Added Stereo Tool as an optional feature to process audio and generate MPX signals
- Added a new audio mixer that allows combining up to 5 audio input sources into a single mixed output
- Added loudness measurement based on the EBU R 128 and ITU-R BS.1771 standards, providing standardized loudness monitoring across active audio channels.
- The link offset of the optional AES67 outputs is now configurable to e.g. allow better compatibility to Dante (CSM-1483)
- Optional DAB+ encoder: If the input source for a DAB+ encoder is an Icecast stream, it's now possible to update the DLS dynamically based on the stream title metadata contained in the Icecast stream. This can be enabled in the Icecast input source configuration. Additionally the PAD Data configuration for the dynamic DLS update of the corresponding encoder has to be switched to "Input Source".
- Added a configuration option for the favicon (website/tab icon). You now have the possibility to show the overall status of the device via the icon color, or show the device type via the icon for better differentiation in case of many open browser tabs.
- Added the possibility to decode SCTE-35 splice inserts from a TS/Demux input source.
- Added the possibility to limit the bandwidth used by SRT
- Optional DAB+ encoder: added the possibility to send the DAB DCP output via SRT
- For the optionally available SDP files on the Status/Storage page it is now additionally possible to copy the content of the file to the clipboard (instead of downloading the file)
- Added Icecast server as new input source type, allowing to get audio from an Icecast source client connecting to that server instance

Changed functionality

- Slight rework of the Storage status page
- Easy2Connect: In case of an incoming SIP call and call acceptance mode set to "Manual", the Easy2Connect page will now allow to accept or decline the call. It does now also show information about the caller.
- SIP: Each SIP input source can only be assigned to a single decoder to assure the unambiguous assignment to the audio input used for the back channel. To prevent invalid configurations, multiple assignments of the same SIP input source are marked as faulty and the configuration cannot be saved.
- The list of present and missing licenses/rights on the System Settings / Global page is now presented in a clearer fashion
- Easy2Connect: Show yellow button in case of incoming SIP call and call acceptance mode set to "Manual"
- The rudimentary wait page shown when e.g. doing a software update or rebooting the device is replaced by a prettier dialog

Fixed Issues

- AES67 outputs now correctly display the control interface when streaming data on control is enabled

- Fixed random and seldom crashes in case SNMP traps are activated and sent
- MP2 encoder: rollback to library version from 2022 to possibly fix rare MP2 encoding problems
- Optional MPE Forwarding: fix not working configuration via selection from list
- Optional DAB+ encoder: In case of FhG MuxEnc usage the multiplexer will no longer change the assigned profile of the corresponding encoder, but will only change the configuration internally, thereby not interfering with encoders having the same profile assigned
- Settings file may get generated empty or incomplete
- Optional DAB+ encoder: DAB PAD data download may eat up memory in case of wrong URL (e.g. an audio stream), leading to reboots (CSM-1525)
- Updated SNMP library for better stability
- Extended the allowed range for the input source gain to -72 up to +12dB (instead of -12 up to +6dB)
- Stability improvements
- Fixed possible crash when loading settings
- Security fix according to BSI recommendation
- VLAN not selectable for SNMP Trap Manager destinations (CSM-1476)
- Periodic reboots due to a memory leak in the Pro-MPEG FEC decoder (CSM-1381)
- Fix possible crash report generation after doing a software update or a reboot (CSM-1499)
- AAC audio decoding from TS may start severely delayed (CSM-1482)
- TS decoding may stop after TS/IP stream input outage in some special cases
- Fix a possible crash when parsing the service table of a transport stream input (CSM-1461)
- Audio buffer is running empty in case the audio output is switched to AES67 or Livewire (CSM-1453)
- RIST requested counter no longer resets after timeout and stream resume
- TS/Demux configuration dialog might not open with some special service lists and the device might even crash when trying to change the configuration (CSM-1427)
- TS Encoder: After a reboot the audio payload might be missing in the generated TS output when AAC is used as the audio codec (CSM-1414)
- Fixed wrong (generic) file name in log entry for settings upload/activation
- Fixed possibly empty VLAN select box (CSM-1423)
- “Timed out” counter was not reset on “Reset counters”
- “Missed” counter is no longer reset on stream resume after timed out input stream
- The elementary stream encoder output send delay did not work for high number of packets per second (e.g. with PCM codec) and/or high send delay
- Enhanced compatibility of the Icecast client
- Fixed audio output getting silent due to wrong automatic detection of codec type even if specific codec type is selected (e.g. MP2) in a TS/Demux input source. Even with a specific codec selected the (potentially wrong) automatic could kick in and break decoding.
- Livewire SRC name changes on the device are not changing the advertised name immediately, but only after a reboot (CSM-1385)
- Fix a possible (and theoretically often) crash when loading settings, doing factory settings or changing the audio output name
- Ember+: Increased buffer sizes to prevent malformed packets when a lot of entries are subscribed (CSM-1308)
- Changing just the input source name did not change its name in the extended log
- The Icecast client may get stuck in a redirect, not being able to re-establish the stream decoding (CSM-1358)

- AAC decoder: in case of bad input (due to e.g. stream or reception interruptions) the audio level may change unintendedly when the decoding resumes (CSM-1074)
- The individual gain may not get applied after reboot for TS/Demux inputs (CSM-1256)
- TS/IP input stream may not continue decoding after stream interruption (CSM-1077)
- Improved Icecast client compatibility in case of connection problems
- Encoding Livewire inputs with NTP based SPN lead to decoder audio and sync errors (CSM-1127)
- SIP: if the registrar registration failed (e.g. due to DNS error), the registration was only retried once per hour. Now it is retried after 120 seconds.
- Loading factory settings might not stop/clear all input sources internally which could result in old settings still being used
- Ember+ may answer with “null” on set commands (disturbing e.g. proper operation of VisTool)
- Icecast client may stop receiving data (after redirect to illegal URL) and does not recover automatically
- General Icecast client compatibility enhancements
- Changing the input source gain of a file input source was not applied to all instances
- Optional DAB+ encoder: DCP output stream did not work reliable with FEC activated
- Optional DAB+ encoder: Changed default FIC

Known bugs

- PTP-synchronized AES67 output streams sent through non-PTP-synchronized interfaces use an incorrect base clock, resulting in a 37-second offset
- AES67 Output synchronization is broken for Clock Type NTP and PTP

Version 2.7.0

19.03.2025

New Functionality

- DAB+ Encoder: added some status values of the FhG MuxEnc and the PAD bitrate to external APIs like SNMP
- Extended log now tracks NTP reference server changes
- Added new buttons to the Overview page to reset the counters of all decoders or encoders
- Optional DAB+ encoder: the FhgMuxEnc protocol for the connection to a Fraunhofer DAB Mux is now sending automatically a status for "Audio Silence Detection" together with the audio frames. The Fraunhofer DAB Mux is thereby able to determine if the DAB+ encoder feed is erroneous due to the audio silence detection and can switch internally to a backup source.
- New silence mode for elementary stream inputs: when enabled, empty packets will be inserted in case of missing stream input (like the “Generate Null samples” option for AES/EBU inputs, when no input is connected) This will optionally allow an encoder to generate output from an elementary stream input even if no ES Input is present
- Added the possibility to clear the extended log
- Optional HLS streaming was not working properly with some CDN servers due to cache-control
- Added optional TSL/SSL encryption for Live Listening (to support access via https from within the web interface)
- Added separate status values for the TS Multiplexer outputs readable via external APIs (see e. g. SNMP MIB)
- Added uptime values for RTP/SRT input streams and RTP/SRT encoder outputs as separate values readable via external APIs

Changed functionality

- In case of limited screen width, the new hamburger menu is now more prominent. Additionally, the collapsed menu can be pinned to stay on the screen even in case of limited screen width.
- The PER (packet error/loss rate) for RTP inputs was formerly measured in an interval of 1/10 of the T1 time of the packet loss switch criteria time (where the default is 60s resulting in a default measurement interval of 6s). The measurement does now adapt to the type of incoming stream and its number of packets per second to allow for a resolution of the PER of 0.1% (meaning 1 of 1000 packets). To be able to measure that at least 1000 packets need to be the measurement interval which might result in the PER not updating very frequently.
- The menu of the web interface does now collapse to a hamburger menu in case screen width is limited
- Show Dual Streaming block on Overview even if FEC is enabled
- Icecast server: the maximum number of clients was limited to five. This is increased now to 15 (but only after a “Load factory settings” is done). Alternatively, the maximum number of clients can be changed via

the external APIs (the SNMP node is e. g. cslAudioEncoderOutputsIcecastTimeout, OID 1.3.6.1.4.1.21529.1001.2.3.18.20.319.4096.1.11)
- Consolidated TCP/IP page to only have one “Save” button

Fixed Issues

- The silence mode available for elementary stream input sources was not working immediately after enabling it and after a reboot
- RTCP sender report may get broken after some time (breaking automatic codec detection and transport of the global delay information in case of synchronous playout / SFN)
- Enhanced compatibility to some WAV files
- Decoder stops working with ancillary data decoding enabled after ancillary data is sent (if DTE output is configured)
- PTP for unicast configuration improved
- Enhanced compatibility of the Icecast client in case of FLAC Icecast server content
- SIP connections for AAC improved
- When changing the silence mode configuration available for Livewire input sources, the change was not applied immediately
- For the audio output configuration, the clock source selection was not available when the signal type was set to Analog, although the clock source still affects the output speed control
- RTP input source: when the input stream changes (e. g. due to an encoder restart), RTP reception might not resume, showing an input bitrate of 0, even if the new stream is received by the device (since firmware version 2.16-rc7)
- The decoder buffer level alarm was broken
- Bunch of internal improvements and optimizations
- Security enhancements
- A livewire input source may suddenly start to provide twice as much packets as before when the “Silence mode” is activated (only true for Livewire destinations targeting the encoders)
- FLAC codec did not work in combination with SRT
- SRT in caller mode doesn’t always connect reliable with the SRT listener
- Not currently active TS/Demux backup sources may show wrong ancillary data on Overview page in case a private PID is used as ancillary source
- When changing the ancillary data configuration the ancillary output doesn't work afterwards or the device may even crash
- xHE-AAC encoder: added parameters to control live loudness and DRC
- UDP ancillary input improved (might not work after reboot when DHCP is enabled)
- When changing a TS/Demux source and switching to the Overview page the device could crash with certain transponders

Known bugs

- Broken HLS Input functionality can cause HLS/DASH streams not being decoded at all.
- Ancillary data: UDP Output not working (reception, insertion and decoding are working)
- MPEG TS Multiplex with MPEG Layer II codec might not be decoded by all decoders (buffer underruns can occur). AAC codecs are working.

- Live Listening only works if the https certificate was accepted manually before pressing the play button (eg. By open the Live Listening url in a web browser)
- Load factory settings works but causes a MoIN restart

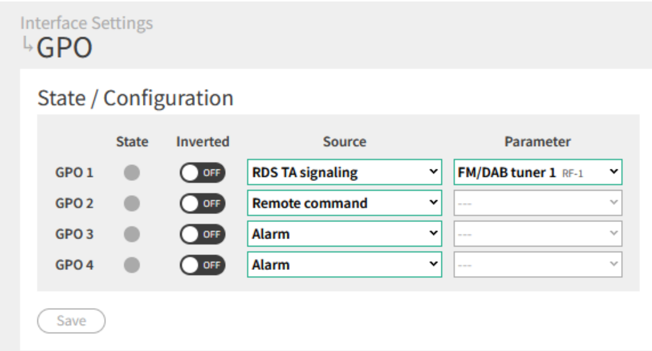
MoIN Release Notes

Version 2.6.3

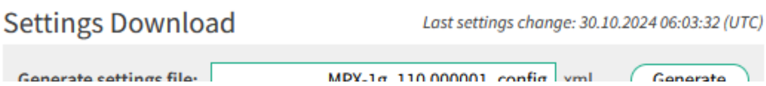
10.12.2024

New Functionality

- Added separate individual status info for AES67 inputs, available via external APIs
- Added a configuration option to file input sources to play only once (no loop)
- Optional FM tuner: added the possibility to signal an active TA via a GPO (new GPO switch source in GPO configuration)



- Scheduler: Enables the execution of various actions at defined times. Currently available actions are activating/deactivating a decoder source
- Last settings change: The time can be read out as a UNIX timestamp or human readable date.
  This is also displayed on the Global page:



Changed functionality

- TS Multiplexer: The “PID removal on bad input” functionality has two different options now. It can not only remove both the PID itself and the signalling via PAT/PMT (what was done until now), but optionally only remove the PID, but still signal it via PAT/PMT.
- DNS settings have to be done in the MCU. The DNS dialog is removed from the TCP/IP page.

Fixed Issues

- Optional DAB+ encoder: DCP output stream did not work reliable with FEC activated
- Automatic codec detection for TS/Demux input sources may not signal correct AAC type
- Fixed possible application crash when loading settings or loading factory settings
- Individual status values for Livewire input sources did not work via external APIs
- TS Multiplexer: "Low bitrate priority" mode did not lead to the lowest possible audio bitrate overhead
- E-aptX decoding inside TS did not work
- Live Listening might be done with wrong audio speed (when switching between different input sources)
- Enhanced compatibility of the optional HLS client/decoder to certain streams

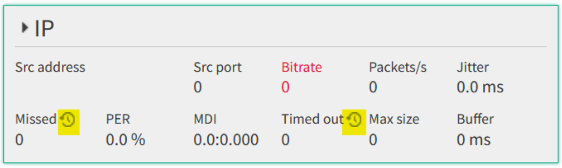
- Decoder: a backup in “Standby” mode isn't activated when an inferior backup is in “Active” mode
- Optional AES67 outputs had a generic name in the SAP announcements – now the audio output name configurable via Codec / Input Sources / Interfaces gets used (if set)
- Optional AES67 outputs: improved startup behavior, inhibiting a possible output burst at startup
- Enhanced stability
- Check minimum SRT passphrase length
- “Jumping” level meters after decoder backup switching on Overview page
- TS/SRT decoding was broken
- SIP: manual connection handling improved for incoming connections
- Live Listening: switching the input source while listening may not work reliable
- Loading settings might not activate all input source settings
- Avoid alarms of inactive channels due to reduced channel count In “Analog/Digital” device mode
- Fix the possibility to get frequent “Encoder – no input data” alarms without “good” messages In between
- Optional DAB+ encoder: Changed default value of DCP output stream data packet spreading to 50% to avoid problems with the receiving multiplexer

Version 2.6.1

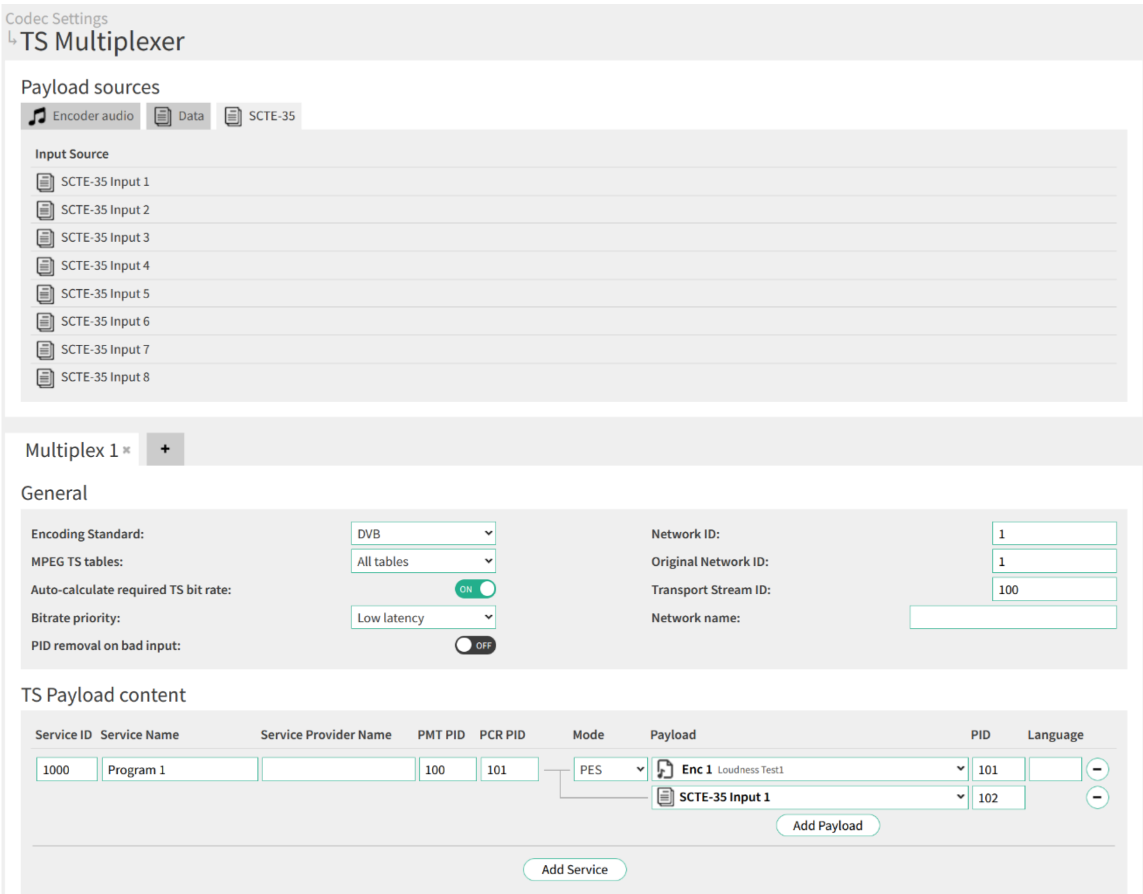
24.10.2024

New Functionality

- Fixed a problem with the AES67 output on device startup (in case a file is used as input source)
- Changed functionality
- Consolidated TCP/IP page to only have one “Save” button
- The event log page does now also refresh automatically when it is initially empty
- Started to add the possibility to get a counter history on the Overview page via a small icon next to the label. It’s more or less a shortcut to the extended log, showing you only the entries for the counter of this input source. Currently it’s added to the “Missed” and “Timed out” counter, others should follow.



- TS Multiplexer: added support for SCTE-35 (Digital Program Insertion Cueing Message)
  There are new SCTE-35 endpoints/inputs available on the TS Multiplexer configuration page:



  Splice inserts must be sent via one of the external APIs. Via SNMP the node is

virtCslAudioEncoderScte35Spliceinsert (OID 1.3.6.1.4.1.21529.1001.2.3.18.466)
The splice information has to be passed in JSON format and should look like this:
{"cue_point": { "event_id": "1073742575", "splice_time": { "year": 2024, "month":9, "day":"6", "hours": "12", "minutes": "17", "seconds": "37", "mseconds": "210"}, "duration": "149.5"} }
(Time information in UTC, more information will follow in the manual)

Changed functionality

- Consolidated TCP/IP page to only have one “Save” button
- The event log page does now also refresh automatically when it is initially empty
- Consolidated TCP/IP page to only have one “Save” button
- The event log page does now also refresh automatically when it is initially empty

Fixed issues

- Fixed a problem with the AES67 output on device startup (in case a file is used as input source)
- The optional ASI input may not detect the TS coming from the ASI signal connected to the ASI input after a reboot of the device
- Fixed high number of DNS requests with NTP synchronization enabled
- Fixed T1/T2 alarm times not editable
- Fixed possible crash when embedding ancillary data in the encoder with certain codecs
- ProMPEG FEC encoder reports wrong FEC matrix when one FEC port offset is set to 0
- ProMPEG FEC decoder let main input stream fail, if one FEC port offset is set to 0
- Dedicated status values for TS/SRT did not work at all
- "Automatic" decoder config did not work when relying on SAP as the source for the codec detection (e. g. for AES67 inputs)
- General playout stability improvements
- Clearing the extended log did not work (the latest entries reappeared after a few seconds)
- X.509 certificate error log entries moved from event log to extended log
- Fix accidentally shown number in encoder info blocks on Overview page
- In case of audio errors, the corresponding event log entry may show the wrong reason for the error
- An audio error may occur when an external clock switch happens
- Second stream in DualStreaming setup may show missed packets, although they weren’t missed
- Added "Remote command" to GPO switch source select
- Fixed a problem on the optional Easy2Connect web page, not showing the phonebook correctly
- Elementary stream output with external clock set to NTP, activated SPN/SFN and DualStreaming with send delay ≠ 0 can lead to jumps in the AES67 output of a decoding MoIN instance, leading to errors on devices receiving the AES67 output
- Icecast client: https Icecast streams with credentials may have problems with authentication
- TS Multiplexer: configuring a new multiplex via the web interface can lead to an invalid configuration
- (Limited) support for Internet Explorer 11 was broken
- Fixed a problem on the Codec page, sometimes not allowing to switch between the “General” and the “Switch Criteria” tabs in the input source configuration dialogs

- A disabled decoder (switched to Off) could still show a green/valid status on the Overview page instead of the grey/inactive status

Version 2.6.0

29.08.2024

New Functionality

- File playback did only support files < 2 GB
- Added support for WAV files with BW64/RF64 file format
- TLS can be enabled / disabled for Live Listening
- Supporting DAB+ encoder (specific license / right required)
- Supporting µMPX encoder (specific license / right required)
- Supporting APTmpX encoder and decoder (specific license / right required)

Fixed issues

- S3 storage can not be started due to a library conflict
- Ancillary data sources might not work after loading a settings file with a corresponding configuration
- Icecast server: removed the additional “bitrate” information from the connection request answer (in addition to the “ice-bitrate” information, introduced in 2.15-rc7), as it breaks compatibility with other Icecast clients, e. g. VLC
- Changing the SNMPv3 authentication protocol (MD5 <-> SHA1) did not work
- Improved initiating/terminating a SIP call via GPI
- After IP address change the SRT decoder (in Listener mode) did not receive any SRT streams
- Corrected the spelling of "Input" in the encoder low level alarm log entries
- Optional DAB+ encoder: input source reassignment for an encoder was not correctly considered when the encoder was also used for a DAB Submux
- Diagnostic Report fixed
- Fixed broken design on TCP/IP page

Known bugs

- Live Listening does not work in the web GUI in https / TLS mode

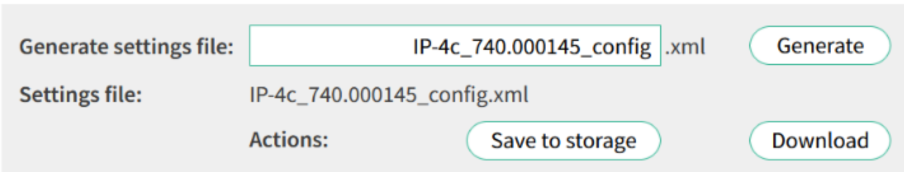
Version 2.5.5

03.06.2024

New Functionality

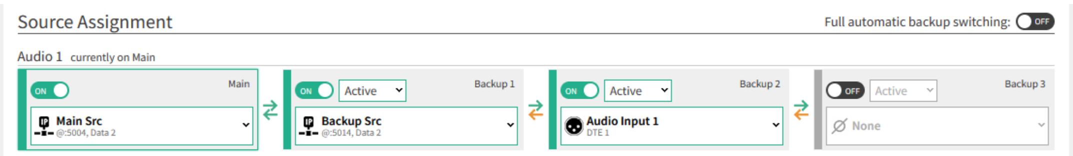
- Added parity configuration for DTE outputs
- Added traceroute possibility to the web interface (Network Settings / TCP/IP / Tools)

- Added event log entries when loading a settings file or loading factory settings
- When generating a settings file on the Global page you can now not just only download the file, but also save it to the internal storage. You can also directly change the name of the settings file



- Added RTCP switch to AES67 audio input configuration
- Added ping possibility to web interface (Network Settings / TCP/IP / Tools)
- Added configuration of SIP call acceptance mode per SIP input source (Automatic (all) / Manual (all) / Automatic (phonebook only) / Manual (phonebook automatic) / Reject) If set to Manual the actual call acceptance is only possible via the external APIs, e. g. Ember+.
- Added encoder input low level detection alarm (in addition to the silence detection alarm)
- Added an event log entry when setting a GPO by GPI tunneling
- Added entries to the extended log in case of CC (continuity count) errors in transport stream decoding
- Added silence detection alarm for encoder inputs
- Added missing VLAN support to SRT input/output configurations

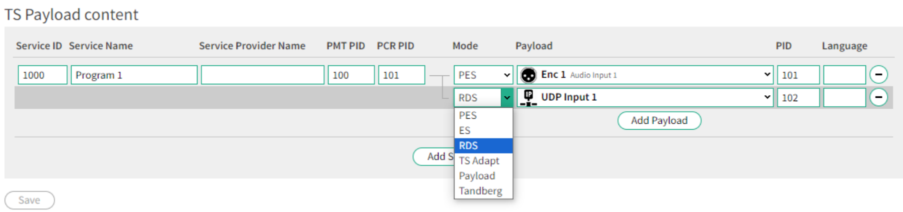
- For security reasons the password hashes of the user accounts are no longer stored as MD5, but as bcrypt hashes. The migration will be done automatically.
- Added the IP interface link status to the IP interface selection in the different configuration dialogs
- Added speed selection (Auto / 1000 Mbit / 100 Mbit) to the TCP/IP interface configuration.
- Sometimes auto negotiation does not work correctly. In this case a manual configuration might help.
- SRT: added source address and port as well as uptime (time since last connection start) to the decoder status information
- Added the possibility to configure individual switch criteria per input source Instead of having one global witch criteria configuration per input source type each input source can now have different switch criteria, e. g. different levels and times for the audio silence detection. To allow this each input source configuration dialog now has an additional tab allowing to enable and configure individual switch criteria.
  This change did also introduce a new visual appearance for the configuration dialogs.
- The fully automatic switching between the different decoder ranks (Main, Backup1-3) in both inferior and superior direction can now be deactivated. It will then allow you to configure, if the backup switching between two decoder ranks should be automatically done in both inferior and superior direction (as it's done by default) or if the automatic is only allow to switch in inferior direction. The switch back to a superior input rank will then not be done automatically but only on user request (e. g. after checking the superior input signal manually).



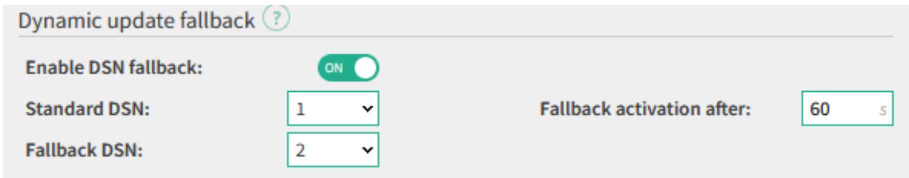
- Added an (optional) HLS/MPEG DASH client for the decoders (no transcoding)
- Added a possibility to configure a delay for the GPI Forwarding/Tunneling (per profile)
- Added AES/EBU CRC Error counter to Overview page
- Added the possibility to access the latest 50 event log entries via external API (e. g. SNMP)
- Added SNMP traps for detected local GPI and remote GPI (GPI Forwarding) changes
- Added the possibility to enable an automatic firmware update check, giving a hint in the web interface, that a new firmware is available without the need to manually check it
- Some minor improvements to logging of SRT drops and connect in extended log

Changed functionality

- The PTP configuration did only allow to configure a single IP interface, providing no backup functionality in case PTP fails on that interface. Now all interfaces configured for streaming data (under Network Settings / Services) will be used for PTP, automatically switching between the interfaces when needed
- New version of the xHE-AAC encoder (04.05.04) with default loudness -24 LUFS
- Updated Fraunhofer libraries to latest versions
- TS Multiplexer: Rework of the private data encapsulation mode configuration. Instead of configuring a global mode the mode can now be configured individually per payload

  As part of this change we also added three more private data encapsulation modes: TS Adaptation, Payload (behind TS Adaptation) and a special Tandberg compatibility mode.
- RDS Databridge (optional): If no RDS update is received for a period of time, the Databridge can switch the RDS encoder to a "Fallback DSN".

- The manager user can be granted limited access to the Global web page, allowing just to switch between different configurations
- Cleanup of SRT / TS/SRT Overview info block fields
- Allow lower SAT transponder frequencies for C band transponders
- Loading settings via the web interface will show a warning when trying to load settings from a different device type. The settings can still be loaded, though, as many settings are of general nature.

Fixed issues

- Fixed communication with NTP service
- Fixed playout error for files and Icecast streams when a NTP service is configured
- When doing “Load factory settings”, the user account passwords were not reset to default
- When loading a settings file, the user account passwords were not taken from the loaded settings file
- The optional low symbol rate SAT tuner did only work with symbol rates down to 100 kSym/s. This limit is now reduced to 64 kSym/s
- The metronome icon on the Overview page (which should signal external clock usage and its state) was not shown appropriately
- The AES67 output may provide packets that are “too early” in certain situations
- The audio buffer configuration did not work with small audio buffer values for the TS/Demux input sources.
- The overall delay was significantly higher compared to our old FlexDSR series for certain audio streams (with more than one audio frame inside one PES frame). This should now be almost on par.

- Elementary stream input sources with multicast addresses were re-initialized when saving although no relevant changes were made
- Fixed audio buffer drift in case of enabled external clock (PTP/1PPS) and enabled sample rate converter
- Web interface: session timeout did no longer work
- SRT input stream will be only restarted if parameters were changed
- Standby input sources were no longer activated when needed
- UDP ancillary output status did not always work
- When switching from PTP to internal clock, the audio outputs had a short distortion
- Changing the sample rate converter configuration of one audio output did disturb all audio outputs
- Web interface: on the Login page the user name field now has the focus on page load to allow immediate typing without the need for the additional click into the field
- Web interface: for elementary stream and TS/IP input sources not applicable configuration options are getting hidden in case of UDP protocol selection
- Icecast client: enhanced compatibility to OGG/FLAC streams
- Icecast server: enhanced compatibility to some clients (expecting a “bitrate” information instead of the “ice-bitrate” information)
- RDS Databridge (optional): the EON TA functionality does now respect the EON PI configuration and will not react on EON Ta from a service with a different PI then the configured one
- Improve robustness of the TS decoder (could hang up completely on bad input data, stopping TS processing completely)
- TS private data decoding may stop after bad weather and disturbed SAT reception
- File playback can be disturbed (since firmware 2.15-rc1) -> the minimum configurable audio buffer has been increased to 100ms
- Fixed Icecast client issues with Icecast streams with ContentType audio/mpegurl
- When switching between different elementary streams (e. g. Ravenna streams) the device may crash and gets unresponsive
- Reverted IGMP rejoin functionality introduced in firmware 2.15-rc2, as it did more harm then being useful in the way it was done
- Optional HLS encoding: fix wrong HLS container formats
- General security and access control improvements
- Again: Improved PTP synchronization of AES67 output, improving AES67 output stability
- PTP did only work with domain 0
- Fixed some problems with HE-AACv2 and the Icecast Source Client and Icecast Server
- Enhanced compatibility to AAC encoded in short frame mode
- When receiving Multicast streams, an IGMP rejoin is done if no streaming data is received for some time
- Improved AES67 output PTP synchronization in case of PTP errors/problems
- Changing just the audio buffer level of an input source is not (always) applied
- Fix the possibility that the TS decoder does not provide the service list (showing 0 services)
- After settings updates (loading a settings file or loading factory settings) input source changes may not be applied
- TS/Demux: the audio buffer is automatically adjusted to the PES frame length if the configured value is too low to avoid audio buffer underruns

- DTE output configuration with Private Data MPE demux can disturb other DTE outputs
- Under high CPU load IP packets may get lost and reported as missed
- Under high CPU load the audio output may get distorted (event log showing FPGA underruns)
- AAC decoding may encounter problems in case of SIP connections to some other vendors (problem with short frames)
- Enhanced SIP connection compatibility via mobile connections (e. g. LTE)
- The AES67 output may still not be synchronous to PTP
- High CPU load in case of incomplete TCP connections (Icecast/HLS)
- Optional HLS decoder: Improved compatibility
- Web interface: the page heading may show an incorrect title when some menu entries are disabled
- Icecast streams with FLAC codec did not work
- AES67 inputs/outputs allow invalid 32bit sample width
- AES67 input/output is always shown with 24bit sample width instead of the configured one
- Long Icecast URLs containing the ‘&’ character failed to be imported via the settings file
- The audio output can get distorted when the audio outputs have the digital reference input enabled and a decoder input source is providing audio with a sample rate different than the one from the audio reference input (can happen e. g. with a SIP input source and an incoming call with e. g. G.711/722 as the codec)
- The AES67 output may not be synchronous to PTP
- Fixed potential high amount of audio errors due to FPGA audio buffer underruns
- Fixed visual problems with the TS Demux service select update/refresh
- SAT tuner alarms were swapped between RF1 and RF2
- Ancillary data from Icecast input sources may not be put out on device startup
- The audio error counter may not increase for very small buffer underruns
- AES/EBU audio output improved for 192kHz input streams
- TS Encoder: audio decoding may drift due to a problem with the PTS timestamps
- TS Encoder: fix a possible mismatch of TS stream type and audio descriptors in case of AAC
- Device could hang up when the audio file belonging to a file input source is not present (e. g. after loading a settings file)
- Fixed a high CPU load problem when many AES67 receive streams are configured
- MPEG TS decoder: enhanced audio decoding compatibility of RTP streams inside MPE
- Synchronized Playout did not work correctly without SFN license
- Ember+ improvement for read-only items (in the virt subfolder)
- Some special characters in the name fields of the input sources etc. could mess up the display of the defined settings in the web interface
- SAT tuner status may get unavailable (e. g. after disabling/enabling a SAT input source)
- Extended log entries for missed RTP packets may have empty stream names, if the packet was lost in a redundant (DualStreaming) or FEC stream
- Fix possible crashes with AES67 inputs (due to new extended logging)
- External clock configuration - a reconfiguration from an already configured and valid clock source (e.g. from PTP to PPS) was not handled correctly
- SNMP: virtCslAudioDecoderStatusAudioDecoderstate (1.3.6.1.4.1.21529.1001.35.2.3.4.37.3.4096.1.7) did not always provide the correct state

- Optional decoder ancillary data is now also shown on the Overview page if the input source is not the currently active one
- The Icecast Server encoder output did not work with HE-AAC(v2)
- With the optional low symbol rate SAT Tuner the device still could fail to re-establish tuner lock and the audio outputs stay silent if reception gets interrupted due to e. g. bad weather conditions
- SNMP monitoring status was not reset to “disabled” if all input sources are getting disabled
- SNMP monitoring status / Warning LED / Relay were not reset (e. g. for the “Audio – No Input Data” alarm) if all input sources are getting disabled
- The optional SAT Tuner alarms are no longer triggered if the tuner is inactive (unused)
- Alarm "No Input Data" did not check if PES/MPE data is present in case of TS/Demux input source
- Fixed "good" check for ES input source switch criteria "No input data" (T1 time was not considered correctly)
- Fixed "good" check for file input sources (were always bad since latest firmware when “No Input Data” switch criteria was activated)
- Fix AAC-ELD(v2) ES decoding not working when Decoder type is explicitly set accordingly
- Fix local audio level display for an active SIP connection
- FEC: Fix audio frames being unnecessarily recovered in case of 1x4 matrix
- Cyclic EON-PS and EON-PTY did not work with optional FM Tuner and RDS Databridge
- Optional TS MPE Encoder: enhanced compatibility to certain 3rd party decoders

Known bugs

- Live Listening does not work in the web GUI and VLC player
- S3 storage can not be started due to a library conflict

Version 2.5.3

21.12.2023

New Functionality

- SRT multicaller function added.
- Implemented https usage for MoIN rest API.
- Added an option to disable the RTCP output in parallel to an RTP output stream
- Added a configurable silence mode to Livewire input sources for the encoders, inserting empty packets in case of missing Livewire stream input, thereby having continuous encoder output
- Added a new status value providing the delta time between the two streams in RTP DualStreaming setup
- Added configuration option to possible Mono Downmix for the audio outputs, allowing to select the Mono output being either a mix from left and right channel or only the left or right channel
- Decoder backup switching events do generate now event log entries (finally) with the cause of the backup switching (in case of inferior backup level activation) and the backup level which was activated. A corresponding SNMP trap will also be sent out.
- The event log web page will now update automatically when new events arise while having the page open
- First steps to more advanced / extended logging: For some events, which were previously only counted (like missed packets) there's now the possibility to check the point in time when these events occur. There's a new "Extended Log" tab on the Log web page, which shows these events. Currently we added events for RTP Rx start, RTP missed packets, RTP unrecovered packets, RTP Rx timeout, SIP register/connect/disconnect/ declined/error and SRT connect/disconnect. Like the counters this extended log is volatile, meaning it is cleared after a reboot. If syslog is enabled, these events will however also be sent out to syslog, making them externally persistent.
- Added optional support for NFS storage (for audio files; in addition to the internal storage)
- Added optional support for AWS S3 storage (for HLS Push encoding)
- Added interface (and VLAN) selection to the SNMP trap manager configuration
- HLS encoder: HLS container format is now configurable
- Added alarm/monitoring for the optional ASI input
- Added silence detection alarm for encoder inputs
- Added missing VLAN support to SRT input/output configurations
- For security reasons the password hashes of the user accounts are no longer stored as MD5, but as bcrypt hashes. The migration will be done automatically.

Fixed issues

- Fixed SNMP settings, where multiple MoIN containers can cause a high CPU load if multiple VLANs are set up.
- VLANS changes are stored correctly now.
- Fixed rest API links + browser access.
- Fixed MoIN reboot issue, when multiple MoIN containers are restarted at the same time.
- RTP dual streaming improved

- Livewire SRC and DST mapping/numbering was wrong when less than the max. number of channels is licensed, delivering no or wrong audio data
- HLS encoder: small bugfix for FLAC and ISOBMFF
- Sample frequency detection for AES/EBU audio input channels higher than 1 improved in case the samplerate converter at the audio input is switched off (resulting in audio distortions)
- RTP receiver information in case of dual streaming improved
- Improved stability (jitter) of AES67 output
- Fix possible crash when changing the sample rate of an audio input
- RIST receiver improved for low latency transmissions, avoiding sometimes quite aggressive retransmission requests
- RIST receiver improved for dual streams with encoder send delay
- Switching VLAN activation without modification doesn't work
- Control services are not correctly setup in VLAN environment
- When a sampling rate for an audio input in analog mode is set to a value different than 48kHz, the decoded audio is distorted
- File input source from NFS source does not perform auto-reconnect on XML settings import
- Digital interface output level can be displayed greater than 0 dBFS when a gain value > 0 is configured and the audio level is high
- Fixed shown name of radio input source during Drag&Drop
- Drag&Drop of TS Data Demux input sources did not always work Changing an active gateway address for a VLAN requires a reboot to become valid
- VLAN modifications could influence the main interface, too
- SAT tuner: if reception gets interrupted due to e. g. bad weather conditions, the device still could fail to re-establish TS decoding and the audio outputs stay silent
- NFS handling improved
- HLS server push improved for S3 storage
- HLS encoder: Fixed issues with xHE audio on stream start with Safari and iTunes
- Fixed Livewire routing protocol LWRP not working after changing the IP address of the device
- MPE decoder: MPE ancillary data output was not working
- RIST encoder retransmission improvements in case of SFN
- Fixed possible timestamp display problems in new Extended Log
- VLAN output streams will be now refreshed after config changes to VLAN parameters
- Decoder input sources with backup policy "Active when needed" might not get deactivated again when activating superior source
- MPEG TS Decoder: optional decoding of 192kHz PCM in PES mode without external clock did not work
- Ancillary data output for SRT input sources may not work when having configured "Audio Output X" as input source for the Ancillary Output
- The new “Auto Refresh” of the event and extended log web page sometimes didn’t work
- Livewire stream names with more than 15 characters could crash the system if the Livewire Routing Protocol LWRP is active
- SNMP get for virtual IP address nodes did always return 0.0.0.0 (e. g. virtCslIpcfgtempCtrlIp, OID 1.3.6.1.4.1.21529.1001.35.2.42.43.1)

- Encoder did not react on SRC on/off in audio input config (potentially changing input sample rate)
- RIST receiver improved for RTP streams, where the encoder has RIST disabled (could lead to audio buffer not building up)
- RIST didn't work in dual streaming setup for the redundant line
- RTP: Overall packet lost counter could be too large in rare cases
- HLS encoder: fixed a problem with some HLS clients reporting faulty HLS segments (especially with the AAC codecs)
- Fixed compatibility issues when MM01 is the audio encoder and RTP packet fragmentation is activated via "RTP max payload"
- MPEG/TS decoder: enabled the possibility to decode audio streams not announced via PAT/PMT automatically without the need to set the codec type manually
- MPEG TS decoder: fixed compatibility of private data TS decoding if the PID is not announced via PAT/PMT
- MPEG/TS decoder: enhanced audio decoding compatibility of RTP streams inside MPE
- Added support to transcode TS/Demux input sources with private data in Pipe mode
- The new "Auto Refresh" of the event and extended log web page sometimes didn't work
- Livewire stream names with more than 15 characters could crash the system if the Livewire Routing Protocol LWRP is active
- SNMP get for virtual IP address nodes did always return 0.0.0.0 (e. g. virtCslIpcfgtempCtrlIp, OID 1.3.6.1.4.1.21529.1001.35.2.42.43.1)
- Encoder did not react on SRC on/off in audio input config (potentially changing input sample rate)
- RIST receiver improved for RTP streams, where the encoder has RIST disabled (could lead to audio buffer not building up)
- RIST didn't work in dual streaming setup for the redundant line
- RTP: Overall packet lost counter could be too large in rare cases
- HLS encoder: fixed a problem with some HLS clients reporting faulty HLS segments (especially with the AAC codecs)
- Fixed compatibility issues when MM01 is the audio encoder and RTP packet fragmentation is activated via "RTP max payload"
- MPEG/TS decoder: enabled the possibility to decode audio streams not announced via PAT/PMT automatically without the need to set the codec type manually
- MPEG TS decoder: fixed compatibility of private data TS decoding if the PID is not announced via PAT/PMT
- MPEG/TS decoder: enhanced audio decoding compatibility of RTP streams inside MPE
- Added support to transcode TS/Demux input sources with private data in Pipe mode
- The web interface can get unresponsive on the Overview page, when an input source description is longer than one line and changes its content regularly (e. g. in case of file playlists)
- The AES67 output may not be synchronous to PTP
- Individual switch criteria changes of an input source were not applied immediately
- Wrong individual switch criteria for the file input sources
- Ancillary data from Icecast input sources may not be put out on device startup
- The audio error counter may not increase for very small buffer underruns
- PTP clock synchronization for AES67 output improved
- DNS configuration per interface did not work via LCD menu

Known bugs

- n/a

Version 2.5.2

25.07.2023

New Functionality

- n/a

Fixed issues

- Improved NET-SNMP communication that could cause errors and CPU load
- Fixed agentx master agent error by increasing app queue stack from 256 to 2048

Known bugs

- n/a

Version 2.5.1

07.07.2023

New Functionality

- n/a

Fixed issues

- Fixed high CPU load after MoIN start / reboot.
- Improved MoIN transcoding performance.

Known bugs

- n/a

Version 2.5.0

03.07.2023

New Functionality

- Decoder backup switching events do generate now event log entries (finally) with the cause of the backup switching (in case of inferior backup level activation) and the backup level which was activated. A corresponding SNMP trap will also be sent out.
- The event log web page will now update automatically when new events arise while having the page open
- First steps to more advanced / extended logging: For some events, which were previously only counted (like missed packets) there's now the possibility to check the point in time when these events occur. There's a new "Extended Log" tab on the Log web page, which shows these events. Currently we added events for RTP Rx start, RTP missed packets, RTP unrecovered packets, RTP Rx timeout, SIP register/connect/disconnect/ declined/error and SRT connect/disconnect. Like the counters this extended log is volatile, meaning it is cleared after a reboot. If syslog is enabled, these events will however also be sent out to syslog, making them externally persistent.
- Added optional support for NFS storage (for audio files; in addition to the internal storage)
- Added optional support for AWS S3 storage (for HLS Push encoding)
- Added interface (and VLAN) selection to the SNMP trap manager configuration
- HLS encoder: HLS container format is now configurable
- Added alarm/monitoring for the optional ASI input

Changed functionality

- The Icecast Server did still answer with "ICY 200 OK" to a connection request and only for certain user agents / browsers with "HTTP/1.0 200 OK". For some time now the ICY answer is however deprecated and should no longer be used, so we do now always answer to a connection request with the HTTP answer, thereby hopefully improving the general compatibility of the Icecast server to certain clients.

Fixed issues

- Fixed a bug that RIST connections can cause reordered packages
- Fixed bug that only one MoIN was accessible via SNMP by using Linux sockets instead of global port
- Fixed MoIN reboot via web GUI so that the MoIN does not hang on waiting page
- Fixed links for external APIs
- Encoder did not react on SRC on/off in audio input config (potentially changing input sample rate)
- RIST receiver improved for RTP streams, where the encoder has RIST disabled (could lead to audio buffer not building up)
- RIST didn't work in dual streaming setup for the redundant line
- RTP: Overall packet lost counter could be too large in rare cases
- HLS encoder: fixed a problem with some HLS clients reporting faulty HLS segments (especially with the AAC codecs)

- Fixed compatibility issues when MM01 is the audio encoder and RTP packet fragmentation is activated via “RTP max payload”
- MPEG/TS decoder: enabled the possibility to decode audio streams not announced via PAT/PMT automatically without the need to set the codec type manually
- MPEG TS decoder: fixed compatibility of private data TS decoding if the PID is not announced via PAT/PMT
- MPEG/TS decoder: enhanced audio decoding compatibility of RTP streams inside MPE
- Added support to transcode TS/Demux input sources with private data in Pipe mode
- Changing just the LNB config for a SAT input source was not applied
- Optional Live Listening feature did not work with Safari browser
- Fixed ancillary data descriptor in MPEG TS encoding for better compatibility
- A configured source IP for SSM (Source Specific Multicast) could not get deleted
- When switching between different web interface menu items the page will now always scroll back to top
- DTE baudrate changes were not applied immediately, but only after a reboot – Fixed
- Ancillary data was not shown on the Overview page for XLR audio input sources
- NFS handling improved
- HLS server push improved for S3 storage
- HLS encoder: Fixed issues with xHE audio on stream start with Safari and iTunes
- Fixed Livewire routing protocol LWRP not working after changing the IP address of the device
- MPE decoder: MPE ancillary data output was not working
- RIST encoder retransmission improvements in case of SFN
- Fixed possible timestamp display problems in new Extended Log
- VLAN output streams will be now refreshed after config changes to VLAN parameters
- Decoder input sources with backup policy "Active when needed" might not get deactivated again when activating superior source
- MPEG TS Decoder: optional decoding of 192kHz PCM in PES mode without external clock did not work
- Ancillary data output for SRT input sources may not work when having configured "Audio Output X" as input source for the Ancillary Output
- The new “Auto Refresh” of the event and extended log web page sometimes didn’t work
- Livewire stream names with more than 15 characters could crash the system if the Livewire Routing Protocol LWRP is active
- SNMP get for virtual IP address nodes did always return 0.0.0.0 (e. g. virtCslIpcfgtempCtrlIp, OID 1.3.6.1.4.1.21529.1001.35.2.42.43.1)
- Improved stability (jitter) of AES67 output
- Fix possible crash when changing the sample rate of an audio input
- RIST receiver improved for low latency transmissions, avoiding sometimes quite aggressive retransmission requests
- RIST receiver improved for dual streams with encoder send delay
- Switching VLAN activation without modification doesn't work
- Control services are not correctly setup in VLAN environment
- When a sampling rate for an audio input in analog mode is set to a value different than 48kHz, the decoded audio is distorted
- File input source from NFS source does not perform auto-reconnect on XML settings import

- Fixed shown name of radio input source during Drag&Drop
- Drag&Drop of TS Data Demux input sources did not always work
- RTP receiver information in case of dual streaming improved

Known bugs

- n/a

Version 2.4.0

19.04.2023

! Only released as moin-container image. No full release planned.

moin-container:2.4.0

The moin-container is the part of the software that contains most of the audio over IP code base and the user can start multiple instances of this container. It is mentioned separately in this document to differentiate between the software orchestration and the actual audio over IP software.

New Functionality

- NFS Storage can now be setup on page “Storage” and can be used for file playout
- AWS S3 Storage can be mounted and used for HLS PUSH
- Log Entries when switching between main and backups

4999	2023-04-19 11:25:29	Informational	Audio 1 Output Silence Detection (left:-8, right:-8)
4998	2023-04-19 11:25:28	Warning	Output 1 - Backup 2 activated
4997	2023-04-19 11:25:28	Warning	Output 1 - Main source failed: ES - no input data

- Added interface (and VLAN) selection to the SNMP trap manager configuration
- HLS encoder: HLS container format is now configurable

Changed functionality

- The Icecast Server did still answer with “ICY 200 OK” to a connection request and only for certain user agents / browsers with “HTTP/1.0 200 OK”. For some time now the ICY answer is however deprecated and should no longer be used, so we do now always answer to a connection request with the HTTP answer, thereby hopefully improving the general compatibility of the Icecast server to certain clients.

Fixed issues

- Encoder did not react on SRC on/off in audio input config (potentially changing input sample rate)
- RIST receiver improved for RTP streams, where the encoder has RIST disabled (could lead to audio buffer not building up)
- RTP: Overall packet lost counter could be too large in rare cases
- HLS encoder: fixed a problem with some HLS clients reporting faulty HLS segments (especially with the AAC codecs)
- Fixed compatibility issues when MM01 is the audio encoder and RTP packet fragmentation is activated via “RTP max payload”
- MPEG/TS decoder: enabled the possibility to decode audio streams not announced via PAT/PMT automatically without the need to set the codec type manually
- MPEG TS decoder: fixed compatibility of private data TS decoding if the PID is not announced via PAT/PMT
- MPEG/TS decoder: enhanced audio decoding compatibility of RTP streams inside MPE
- Added support to transcode TS/Demux input sources with private data in Pipe mode
- Changing just the LNB config for a SAT input source was not applied

- Optional Live Listening feature did not work with Safari browser
- Fixed ancillary data descriptor in MPEG TS encoding for better compatibility
- A configured source IP for SSM (Source Specific Multicast) could not get deleted
- When switching between different web interface menu items the page will now always scroll back to top
- DTE baudrate changes were not applied immediately, but only after a reboot – Fixed

Known bugs

- Page “System Settings – Ancillary Data” is blank and should be removed.
- The WebUI crashes completely when the NFS connection runs into a timeout due to a not accessible NFS server.
- In inconsistent conditions when a moin-container subscribes two times to the same multicast or if the multicast is sent two times on the same address and port, the container can get unstable due to heavy tracing of the error.
- xHE-AAC: Ancillary data and GPIO Forwarding does not work.

Version 2.3.0

08.12.2022

moin-container:2.3.0

The moin-container is the part of the software that contains most of the audio over IP code base and the user can start multiple instances of this container. It is mentioned separately in this document to differentiate between the software orchestration and the actual audio over IP software.

New Functionality

- Added gain configuration to all input sources to e. g. allow alignment of main and backup sources
- Added playlist support (m3u, m3u8, pls) to the file input source
- Added VLAN support to the Icecast client input source
- Major Livewire integration enhancements:
- Added optional Livewire Sources for the audio inputs, providing its physical XLR audio inputs as Livewire audio streams and optional Livewire Sources for the audio outputs (instead of AES67 streams), allowing to have a Livewire audio stream instead of physical XLR as audio output.
  An IP-4c with 4 channels will announce 8 Livewire Sources, where SRC1 – SRC4 will reflect the audio inputs and SRC5 – SRC8 will reflect the audio outputs
- The Livewire input sources for the Encoder section will now be a fixed number of Livewire Destinations (as much as encoders are available), thereby allowing the configuration via LWRP (corresponding to the fixed number of Livewire Destinations for the Audio Decoder section).
  An IP-4c with 4 channels will now announce 24 Livewire Destinations, where DST1 – DST16 will reflect the possible Livewire input source for Audio1/Main, Audio1/Backup1 and so on and DST17 – DST24 will reflect the possible Livewire input sources for Encoder 1 to 8.
- GPIO Tunneling has now the option to tunnel the Livewire GPO state instead of the physical GPI state.
- Added Livewire level meters for Sources and Destinations (enhance compatibility to e. g. Pathfinder)
- Added support for the Livewire GPIO snake mode, allowing to link the GPOs of the IP-4c to the GPIs of a different Livewire device.
- Added two virtual Livewire GPI ports (GPI 3 and 4), which will reflect the state of GPO port 1 and 2. This will allow other Livewire devices to register for GPI changes via snake mode and thereby follow a GPO change on the IP-4c (e. g. via GPIO Tunneling), reflected as a virtual GPI change. The virtual GPI ports will thereby provide a GPO pass-through to other Livewire devices.
- Allow decoding of not announced TS audio services (with missing DVB tables)
  The codec has to be set manually ("Automatic" will not work)
- Added VLAN interface status

Changed functionality

- Improved the file upload via the Storage page
  The upload limit was increased to 100 MB and the error handling was improved.

Fixed issues

- Fixed a bug in Icecast transcoding where some streams had audio interruptions.
- Fixed display error on overview page when a specific number of encoders are created (e.g. 25 or 27)
- Fixed a possible crash with NTP synchronization and “Bind to interface” option enabled
- Fixed an issue with syslog messages stop working after a reboot
- Fixed a possible crash on the Overview page, when RTP with dual streaming, VLAN and multicast is enabled
- Fix GPIO Tunneling info not shown for encoder if no ancillary data source is selected
- Fix xHE-AAC problems with low bitrates
- Icecast input source handling improved for faulty meta data from some Icecast servers
- IGMP binding improved for RTP multicast (in case of interfaces with identical addresses)
- Livewire: increased general compatibility with Pathfinder
- Livewire: changes done to Livewire Destinations via LWRP were not applied
- SIP: calls could not be cancelled during connection establishment
- SIP: redial improved
- SIP/SDP: (EBUACIP:VERSION) information moved, enhance compatibility to Unify OpenScape SBC
- NMOS: improved compatibility with STAGED mode (SDP parameters)
- AES67 output compatible with DHD (silent mode)
- AES67 outputs loose the PTP synchronization after reconfiguration the AES67 output parameters
- AES67 output: improve stability of empty packets mode
- Fixed MPEG TS signaling for MPEG2-AAC
- Improved transcoding of MPEG TS with ancillary data
- Fixed an issue with the ancillary output not working and providing no data when a TS ancillary data or a private data input source is configured and only assigned to one of the ancillary outputs without using the same TS source in one of the decoders (or maybe encoders)

Known bugs

- n/a

Version 2.1.0

24.08.2022

New Functionality

- Added VLAN Support for AES67 Inputs/Outputs
- Adding VLANs to running moin-containers

Fixed Issues

- Offline installation is now working
- Installations without DHCP are now working
- Fixed a bug that caused moin-containers to have no license even if a correct license was installed
- The NTP synchronization was having big jumps every hour that also resulted in audio errors each hour, this has been fixed by moin-container version 2.1.0
- Fixed a bug that resulted in a crashing ntp service every 12 hours
- The MCU network page sometimes changed the order of interfaces or left out some interfaces. This has been fixed by moin-mcu version 2.1.0
- Uploading new images to page Maintenance was not possible, this has been fixed by a new installation script
- Fixed too low memory allocations for moin-containers that could result in occasional crashes shown as an out of memory crash

Known bugs

- For more than 16 channels (encoders + decoders) the web interface starts to get slow and is not responding well enough
  <u>Workaround</u>: use multiple instances each with up to 16 channels
- MN-63: Unstable results of MoIN MCU Networking page. Hitting the save button can result in a miscommunication not setting the VLANs correctly.
  <u>Workaround</u>: click save 2-3 times
- MN-62: Removing network connection and re-applying it can cause the decoder to lock on Backup 1, which is not recovering from that situation automatically. The problem is caused when the sequence number rolls over
  <u>Workaround</u>: an active RTCP connection (RTP Port + 1) will mitigate the issue, because the decoder recovers quickly from that wrong behavior when it detects a problem with the IP stream via RTCP
- MN-61: Installations with management interfaces of netmasks with 25 bit or more do not work
- MN-60: CPU and memory limits are not using the number of encoders/decoders, this can result in misbehaving moin-containers when too many decoders are used simultaneously
- MN-59: Policy based routing is not working for interfaces with VLANs, that causes problems when an interface with a VLAN tries to send IP data to a destination outside of its subnet that can’t be reached by the systems default gateway

moin-container:2.1.0

The moin-container is the part of the software that contains most of the audio over IP code base and the user can start multiple instances of this container. It is mentioned separately in this document to differentiate between the software orchestration and the actual audio over IP software.

New Functionality

- Completely revised ancillary data handling for full flexibility
- The fixed linkage between ancillary inputs/outputs and corresponding audio inputs/outputs is entirely removed. Instead a variable number of UDP ancillary inputs/outputs can be configured besides the available DTE inputs/outputs.
- When configuring the Encoders on the Codec page, each configured audio input source can be accompanied by one of the ancillary input sources to be encoded together using the given profile. In that way the ancillary input sources are no longer limited to the physical XLR audio input sources, but can now be used together with any audio input source, e. g. Livewire, AES67 or Icecast. In addition there's a special "Pipe" ancillary input selection for the Encoder. Using the Pipe mode the ancillary data of the audio input source (including possible GPI Forwarding) will be preserved and transcoded together with the audio. Moreover there's a new "Ancillary Output" tab on the Codec page allowing to configure freely the input sources to use for the available ancillary outputs (DTE and UDP). When using one of the "Audio Output" input sources (that's the default), the ancillary data coming from the input sources configured on the Decoder tab including the backup switching will get used.
- Ancillary data and GPI Forwarding state available on Overview page
  If ancillary data or GPI Forwarding information is contained in an input source, the Audio info blocks on the Overview page will allow to inspect that data. This is especially useful in a transcoding setup, where the ancillary data would otherwise not be available for examination.
- TS private data output
  It's now possible to configure private data only TS Demux input sources (without the need to configure an audio PID). These "data only" sources can be used on the new "Ancillary Output" tab.
- Optional audio PID removal from TS Multiplexer in case of audio input loss (to e. g. trigger external backup)
- Added NMOS support (Networked Media Open Specifications, https://specs.amwa.tv/nmos/)
- The NMOS support can be enabled on the new "External APIs" page, which summarizes the configuration of the external APIs SNMP, Ember+ and NMOS
- AES67 inputs/outputs can now optionally replace the physical XLR inputs/outputs, e.g. for SIP
- The optional AES67 input stream can also be a Multichannel input stream – it's possible to select the channel(s) to use from it
- Added audio output buffer level alarm (to get e. g. a fast relay alarm before the audio buffer runs empty and trigger thereby an external backup)
- Added audio error count alarm (to get an event log entry in case the audio error counter increases)
- Added new ancillary input and output alarms for the dedicated inputs and outputs
- Added ancillary data decoding support to Icecast input sources
- Added the possibility to bind the optional syslog output to a certain interface
- Added optional support for ASI input (available with the satellite tuner or standalone)
- Enhanced optional MPE encoding capability of TS Multiplexer to support RTP, too

- Added xHE-AAC support to optional HLS encoder
- In case external clock synchronization is active, the transcoding elementary stream output of an unsynchronized RTP stream used as encoder input source can now be synchronized to the external clock (activate the “Synchronous Playout” switch in the RTP elementary stream output settings)

Changed functionality

- RIST: the additional bandwidth used by the RIST encoder can be limited now
- Increased compatibility to not RFC2250 standard compliant MPEG audio streams (e. g. from Telos iPort)
- SAP is now also available with the Livewire license and no longer limited to the Ravenna license

Fixed issues

- A memory leak caused a reboot of the device after a few hours up to a few days in case PTP clock synchronization was enabled
- TS/Demux: fetching the service list could sometimes fail to get all service names
- Livewire source select was wrong in Livewire input source configuration dialog
- Livewire was always able to set GPO without respecting switch source setting
- Changing the manual decoder config of a TS/Demux source did not trigger a reconfiguration
- Fixed a crash when changing the TS/IP input source used by a Demux source in MPE mode
- Audio Output as TS input source did not show its levels in the payload info block on the TS Multiplexer overview page
- When only the port of an elementary stream output was changed, the SAP/SDP announcement wasn’t Updated
- MPEG TS encoder: improved compatibility to certain decoders (PTS offset reduced)
- MPEG TS decoder: private data encapsulated as ES (without PES) wasn’t decoded
- SAP information wasn’t updated on external clock configuration change
- SIP: Support of mobile VoIP devices which don't handle SIP correctly
- SIP: status information for SNMP improved and enhanced
- SIP: Invalid SIP number shown in log if SIP call failed
- SIP: RE-INVITES improved (https://datatracker.ietf.org/doc/html/rfc3581#section-3 is now implemented
- correctly)
- The optional SIRC Data Channel was included in the TS only every 2nd time on TS Multiplexer changes

Known bugs

- Codec page: some PHP error messages will get shown on the TS/IP and TS/SRT input source tabs, when the TS_Decoder license is missing