MPX-2ds/2d

Release Notes

Version 1.03.2

13.02.2026

Your audio. Our solution.

MPX-2ds/2d Release Notes

Table of Contents

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Version 1.03.2

13.02.2026

Fixed Issues

- Sometimes GPI tunnelling events are not processed (CSM-1557)
- µMPX: Buffer level alarm generates incorrect entries (level: 0) (CSM-1543)
- S2X Tuner: fixed wrong RF level display
- Random and seldom crashes due to SNMP trap sending

Version 1.03.1

30.01.2026

Fixed Issues

- SAT reception from RF-2 partially disrupted / no TS table
- S2X Tuner: S2X DVB standard cannot be selected

Improvements

- SAT tuner: Alarms only trigger when the tuner is actively in use (CSM-1514)

Version 1.03

18.12.2025 – update: 30.01.2026

Known issues

- SAT reception from RF-2 partially disrupted / no TS table

New Functionality

- Added the possibility to decode SCTE-35 splice inserts and to trigger a GPO based on that. The SCTE-35 decoding can be enabled in the TS/Demux input source.

- Added the possibility to limit the bandwidth used by SRT

1.03-rc2

- The possible actions that can be triggered by a GPI are enhanced by two new options, which allow to enable or disable a decoder input source
- First support for new Dual DVB-S/S2/S2X Tuner

Changed Functionality

1.03-rc2

- The PTP functionality did get a complete revision, as it was not working reliably. In the course of these changes we did also revise the way to configure everything related to time and clock handling including NTP and external clock.
  Before this revision several menu options were involved in the configuration. The selection of interfaces, on which PTP should be enabled was done via the "Network Settings / Services" menu. The PTP configuration itself (e. g. domain number and delay mechanism) was done via the "AoIP Settings / External Clock" menu (but only allowing to use the same configuration for all interfaces enabled for PTP). NTP configuration was done via "Network Settings / NTP", time configuration (time zone) via "System Settings / Time". Pretty much scattered all over the place.
  All these configuration options are now consolidated into a single place – the menu "System Settings / Time/Clock":
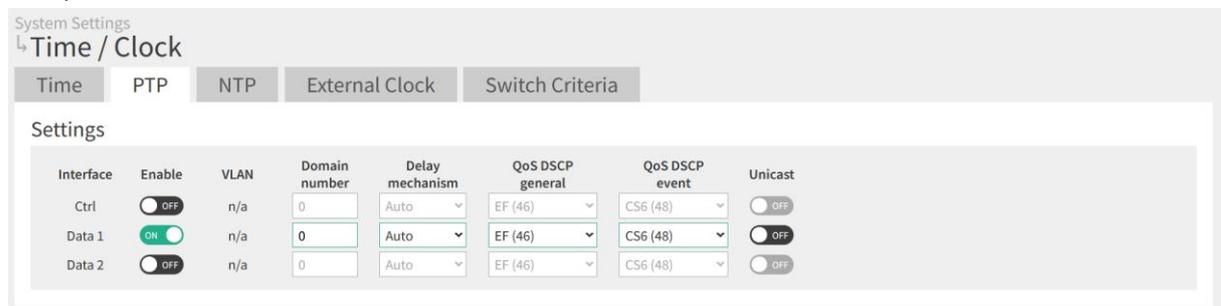
  You still have to configure via the "External Clock" tab (which corresponds to the former "AoIP Settings / External Clock" menu), if and which external clock source (PTP, NTP, 1PPS) should be used for the audio clock synchronization (including optional AES67 outputs).
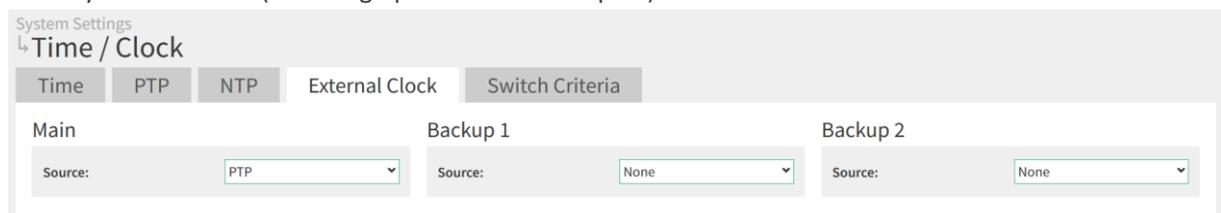
  If a backup for the external clock is configured (e. g. NTP), the switch criteria are now configured via the "Switch Criteria" tab of the "Time / Clock" menu.
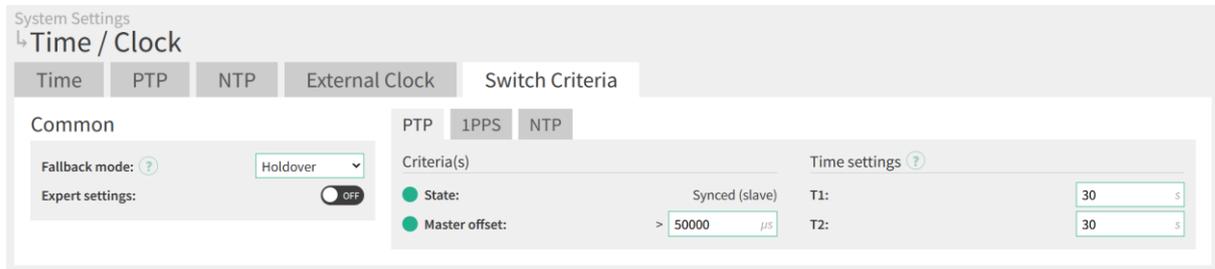
There's also a new "Fallback mode" which allows you to control the behaviour if all configured external clock sources do fail. In "Holdover" mode (the new default) the device will keep the last external clock (no longer regulated) and will not switch to the internal clock, which would always result in a glitch.

- Together with the revision of the time and clock configuration we did also revise the available status information for time / NTP / PTP / external clock. Before the revision the NTP status could be found via the "Status / NTP" menu, whereas the external clock status could be found on the Overview page in a separate tab. PTP status information was only provided via this "External Clock" tab on the Overview page.
The complete status information for all this is now also consolidated into a single place – the menu "Status / Time/Clock":
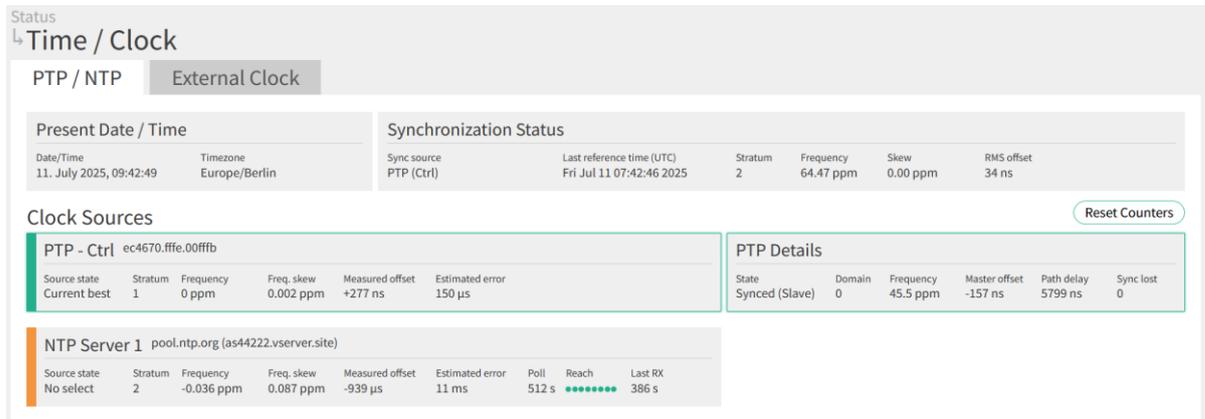
Fixed Issues

- 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 firmware update or a reboot (CSM-1499)
- TS stream info incomplete, or not shown at all on Overview page
- AAC audio decoding from TS may start severely delayed (CSM-1482)
- TS decoding may stop after SAT reception outage in some special cases (but only with the Advanced DVB-S/S2 Single Tuner)
- TS decoding may stop after TS/IP stream input outage in some special cases
- The new Dual DVB-S/S2/S2X Tuner might have a TS bitrate of 0 after a reboot, although the tuner has a lock
- Fix a possible crash when parsing the service table of a transport stream input (CSM-1461)
- RIST requested counter no longer resets after timeout and stream resume
- Frequent buffer restarts when decoding 2x µMPX (excessive CPU load) (CSM-1376)

1.03-rc2

- µMPX: When resetting all counters on the Overview page, brief audio errors sometimes occur (CSM-1335)
- SAT tuner: no private/ancillary data output after reboot (CSM-1305)
- No displayed source on UDP Ancillary Output after select and save (CSM-1305)
- When changing the width of one of the log tables the page got distorted (CSM-1200)
- Icecast Server input source: add/delete didn't show immediate effect
- After e. g. disabling/enabling an AES67 output the stream has an UTC/TAI offset of 37 seconds (CSM-1329)
- The device could crash under special circumstances when loading settings
- SAT tuner: fix wrong status on LCD screen for RF2 input (showing "Not configured")
- Fix the possibility of corrupted core dump files in case of an application crash
- Enhanced robustness of the SAT tuner inputs

Version 1.02.3

24.07.2025

Fixed Issues

- Ancillary data is not processed after a restart (CSM-1305)
- UDP output settings are not displayed correctly (Source is None) (CSM-1305)

Version 1.02.2

25.06.2025

Improvements

Diagnostic report: Extended by further debug information.

Fixed Issues

- µMPX: Disturbed decoding / silence output. Caused by different µMPX buffer values.
  Ext. log shows cyclically: "Realigned µMPX buffer after startup"

Version 1.02.1

19.06.2025

Fixed Issues

- TS/Demux source: Gain may not get applied after reboot.

MPX-2ds/2d Release Notes

Version 1.02

17.06.2025

Improvements

1.02-rc9

- Decoder Main / Backup switching improved

1.02-rc8

- General playout stability improvements

1.02-rc6

- Incorrect configuration of the decoder output type leads to audio errors (example: Audio Mode + µMPX)
- More detailed log entries for possible audio errors

New functionality

- Completely revised dialog that informs about the current status during reboot, firmware update, etc. This dialog is displayed on all browsers that have a connection to the web interface. Regardless of which browser has performed a reboot, for example.
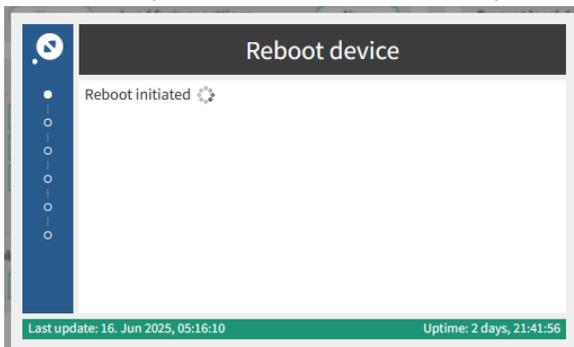
- Added new crash report functionality
  In case the device does crash (performing an unintended reboot) it was until now very difficult to find the root cause for the crash/reboot. In such a case a crash report will now be generated which should be sent to us via the 2wcom Support Center. It will allow us to inspect the reason for the crash/reboot, enabling us to develop a fix for the crash.
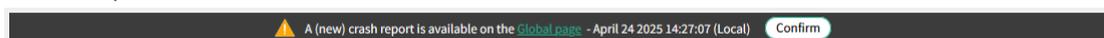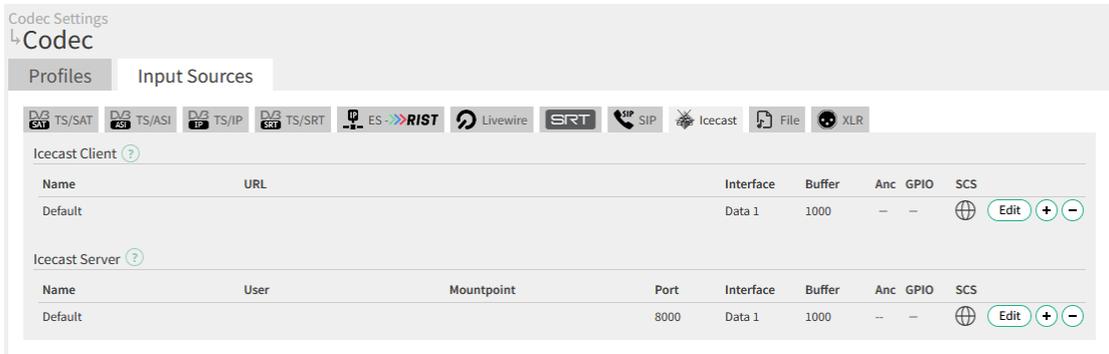

  The crash report can be downloaded via System Settings / Global:

- Added Icecast server as new input source type, allowing to get audio from an Icecast source client connecting to that server instance


- Added PTP QoS DSCP settings
- The PTP delay mechanism to use is now configurable

1.02-rc12

- Added new buttons to the Overview page to reset the counters of all decoders or encoders
- Extended log now tracks NTP reference server changes
- µMPX: Switch criteria "Packet loss (PER)" can now also be evaluated for UDP

1.02-rc11

- µMPX: If more than one µMPX decoder is activated in a decoding chain, the MPX level is now also displayed for the inactive decoder. The prerequisite is that input data is available and the source (µMPX encoder) supports this function.
- µMPX RDS decoding if RDS signal is available
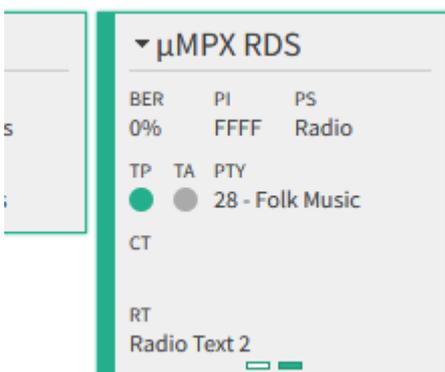
- RDS databridge is now also available for µMPX codec

1.02-rc9

- Ext. log can now be deleted via a button
- GPO: Add "Remote command" to GPO switch source select
- Scheduler: Enables the execution of various actions at defined times. Currently activating/deactivating a decoder source
- Last settings change: The time can be read out as a UNIX timestamp or date
  This is also displayed on the Global page:

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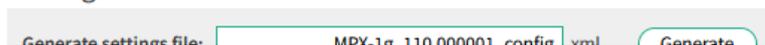
- Input source "File" can be played only once if desired

- 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.

- The LCD screen will now show additionally a SAT status page with reception information:

1.02-rc6

- 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)
- Added the possibility to clear the extended log

1.02-rc1

- Added optional TSL/SSL encryption for Live Listening (to support access via https)
- Added uptime values for RTP/SRT input streams as separate values readable via external APIs

Changed functionality

1.02-rc14

- 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.

1.02-rc12

- 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

- µMPX: If the stream is not actively decoded, all available information about the stream is still displayed

▸ IP					▸ µMPX Standby
-3.7 dBFS Src address 192.168.99.2	Src port 56682	Bitrate 908k	Packets/s 104	Jitter --	Buffer -- Signal lost --
Buffer Err 200 0	Missed --	PER --	MDI --	Timed out 0 Max size 1159	Buffer -- Missed 0 PER 0.0 %

1.02-rc11

- µMPX - Switch criteria: A triggered "No Input Data" criterion now leads to the "Silence detection" monitoring being suspended.
- µMPX: If µMPX generates a dropout message, these now counted on the overview page
- Status values of dual streaming with FEC are now displayed on the overview page

1.02-rc1

- 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

- 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
- Icecast client: enhanced compatibility to streams providing MPEG Layer 2 audio data
- General Icecast client compatibility enhancements
- Changing the input source gain of a file input source was not applied to all instances
- Fixed a problem with the diagnostic report not including all relevant information
- The silence mode available for elementary stream input still had some problems when enabling/disabling it

1.02-rc15

- Stereo decoder: Deemphasis setting is not correct after a device restart
- µMPX: After a restart no µMPX level in the backup stream (partially prevents backup switching)
- Changing the gain of a file input source is not applied to all instances

1.02-rc14

- The silence mode available for elementary stream input sources was not working immediately after enabling it and after a reboot
- Prevent duplicate VLAN IDs on same interface
- 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; since 1.02-rc11)
- PTP for unicast configuration improved

1.02-rc12

- Buffer level alarm not working when uMPX is being decoded

- Alarm for "AES/EBU no signal" not working
- uMPX: Streams with very small data packets (less than 200bytes) not working
- 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
- Bunch of internal improvements and optimizations
- Security enhancements
- 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
- Enhanced stability
- Check minimum SRT passphrase length
- "Jumping" level meters after decoder backup switching on Overview page
- Loading settings might not activate all input source settings

1.02-rc12
- Live Listening: Switching sources / clicking on speakers does not work reliably
- Deleting a VLAN entry is only possible if other settings are also changed
- MPX decoder: fluctuation of the pilot frequency (greater than +-2Hz)
- TS/SRT decoding not possible
- Stereo decoder: Low stereo separation after a short 19kHz sync lost
- uMPX: Rare problem where the decoder could lose packets even though they were received

1.02-rc10
- Fixed possible crash on startup

1.02-rc10
- µMPX ancillary data decoding may drop some bytes
- 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
- 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
- Optional SAT tuner: Switch criteria C/N does not work
- Icecast client: https Icecast streams with credentials may have problems with authentication
- (Limited) support for Internet Explorer 11 was broken

1.02-rc8
- Clearing the extended log did not work (the latest entries reappeared after a few seconds)
- 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

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- Second stream in DualStreaming setup may show missed packets, although they weren't missed

1.02-rc6

- A change of the PCM multichannel offset of an elementary stream input was not taken into account (only after disabling/enabling the input source or rebooting the device)
  1.02-rc5
- Finally fixed the root cause for FPGA underruns and "Restart Audio Buffer" errors

1.02-rc4

- Fixed the root cause for FPGA underruns and "Restart Audio Buffer" errors (in fact not totally – the probability was just reduced significantly)

1.02-rc3

- Fixed PCM clock synchronization (not getting a synchronization lock)
- Increased resilience of PCM decoding (hopefully avoiding often "Restart Audio Buffer" errors)

1.02-rc2

- Eliminated some more circumstances for FPGA underruns (reported in event log)

1.02-rc1

- Fixed audio errors due to FPGA underruns (reported in event log) after a switch between main <-> backup sources has been done
- Enhanced compatibility to µMPX with FEC from StereoTool and Omnia9
- Fixed audio errors in PCM 192 kHz mode in case of missed packets
- Fixed a crash in µMPX decoding when SFN/SPN mode is used
- ProMPEG FEC decoder let main input stream fail, if one FEC port offset is set to 0
- Fix µMPX Dualstreaming
- Dedicated status values for TS/SRT did not work at all
- Fix µMPX decoding from TS in case of PES encapsulation (instead of MPE)

Version 1.01

29.02.2024

Fixed Issues

- Changing the DTE parity setting has no effect
- In some cases, the change to the output source is not adopted
- Changes to the decoder sample rate (optional) are not applied correctly.
- Fixed some problems with the Live Listening feature
- PTP might not work correctly in certain environments
- PTP might not be active after loading a settings file with a corresponding configuration
- Ancillary data sources might not work after loading a settings file with a corresponding configuration
- Changing the SNMPv3 authentication protocol (MD5 <-> SHA1) did not work
- 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 metronome icon on the Overview page (which should signal external clock usage and its state) was not shown appropriately

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1.01-rc4

- Buffer management: Fixed a buffer management issue that was introduced in 1.01-rc2, causing buffers to be played out with a static clock, slowly decreasing or increasing the level until it needs to reset

1.01-rc2

- 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 - Fixed
- UDP ancillary output status did not always work
- When switching from PTP to internal clock, the audio outputs had a short distortion
- 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
- 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
- SNMP: Reading SNMP OID "virtualCommonAudioTsforwardingOutputsMpe2Ipinfo" leads to a crash

1.01-rc1

- GPI Forwarding may have some issues when an input source with GPI Forwarding enabled is used for more than one audio output on different backup levels
- Fixed Icecast client issues with Icecast streams with ContentType audio/mpegurl
- General security and access control improvements
- Fixed some problems with HE-AACv2 and the Icecast Source Client and Icecast Server
- Enhanced compatibility to AAC encoded in short frame mode

New Functionality

- AES67 Outputs (optional): Up to two additional AES67 output streams that can be configured in parallel to the physical outputs.

- Output of the decoded audio directly to the headphone output (requires the Audio Output right)

1.01-rc2

- Added parity configuration for DTE outputs

1.01-rc1

- Added ping and traceroute possibility to 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.

- Settings files can also be uploaded via the Storage page

- Once there are settings files on the internal storage the user can select one of the stored files instead of uploading one and load it via the new consolidated Upload/Activate section, thereby having a sort of preset

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functionality, allowing to quickly switch between different configurations.
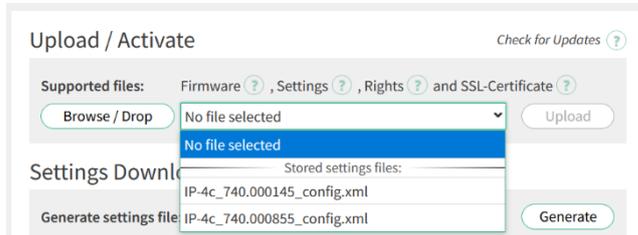
Changed functionality

- File playback did only support files < 2 GB
- Added support for WAV files with BW64/RF64 file format

1.01-rc1

- The different upload sections on the Global web page (Firmware, Settings, Rights, SSL Certificate) were consolidated to just one section allowing to upload every of the four kinds of files

- The manager user can be granted limited access to the Global web page, allowing just to switch between different configurations

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