IP-8e

Release Candidate

Notes

Version 2.17-rc7

02.02.2026

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IP-8e Release Candidate Notes

Version 2.17-rc7

02.02.2026

Fixed Issues

- The ASI input may not lock to an incoming signal
- Settings file may get generated empty or incomplete
- Livewire source: Empty packets not sent

Version 2.17-rc6

26.01.2026

Fixed Issues

- When a sampling rate for an audio input in analog mode is set to a value different than 48kHz, the audio is distorted

Version 2.17-rc5
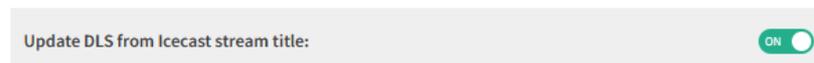
23.01.2026

🎄 Christmas present

- LiveListening is now included free of charge!

New Functionality

- 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:
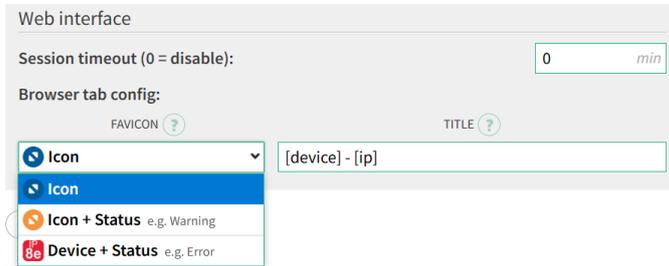

  Additionally the PAD Data configuration for the dynamic DLS update of the corresponding encoder has to be switched to "Input Source".

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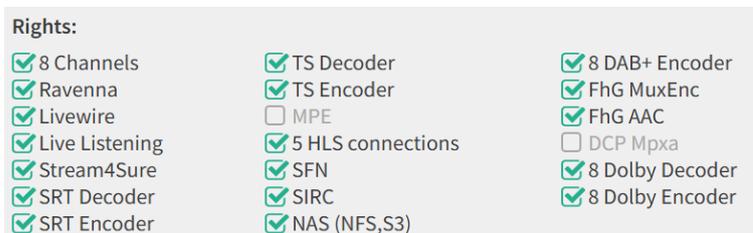- Added a configuration option for the favicon (website/tab icon)

You now have the possibility to show the overall status of the device via the icon color. You might even show the device type via the icon for better differentiation in case of many open browser tabs.
- Added the possibility to limit the bandwidth used by SRT

Changed functionality

- In case of devices with two power supplies, the power LED will flash red/green, if only one power supply is connected to power. This can now be suppressed by enabling only the power supply alarm for the connected power supply and disabling the alarm for the other one. In this case the power LED will show green. Please note, that for this updated behavior you also must update the System Controller firmware to version 1.08 – you will find this firmware via "Check for updates" from within the web interface in the section "Optional Updates".
- The list of present and missing licenses/rights on the System Settings / Global page is now presented in a clearer fashion

Fixed Issues

- 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 firmware 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)

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- 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 via the LCD menu (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.
- PTP: small stability enhancements for the E2E mode
- PTP: added (better) support for the P2P mode
- Fix the possibility of corrupted core dump files in case of an application crash
- 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

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Version 2.17-rc4

25.09.2025

New Functionality

- Optional DAB+ encoder: added the possibility to send the DAB DCP output via SRT

Fixed Issues

- Ember+: Increased buffer sizes to prevent malformed packets when al lot of entries are subscribed (CSM-1308)
- The device could crash under special circumstances when loading settings
- Changing just the input source name did not change its name in the extended log
- The reception of RTCP receiver reports was broken (since 2.17-rc4), thereby breaking the optional RIST functionality
- The Icecast client may get stuck in a redirect, not being able to re-establish the stream decoding (CSM-1358)
- Wrong PTP state in external clock status block
- Optional TS Encoder: fix frame calculation for low bitrate overhead mode
- 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

Version 2.17-rc3

16.07.2025

Fixed Issues

- The revised PTP implementation had a major problem likely selecting the wrong interface (and thereby the wrong PTP hardware clock) internally as the clock source
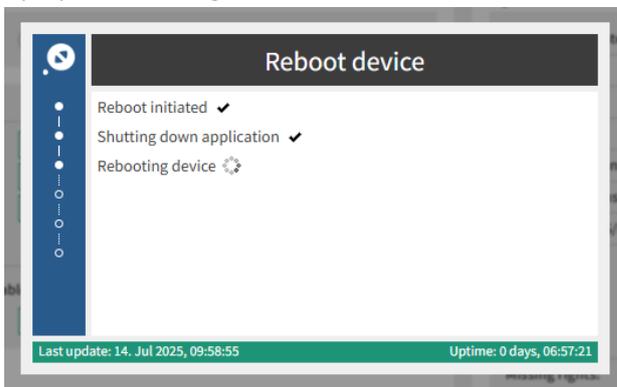
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Version 2.17-rc2

15.07.2025

New Functionality

- 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)
- The rudimentary wait page shown when e. g. doing a firmware update or rebooting the device is replaced by a prettier dialog
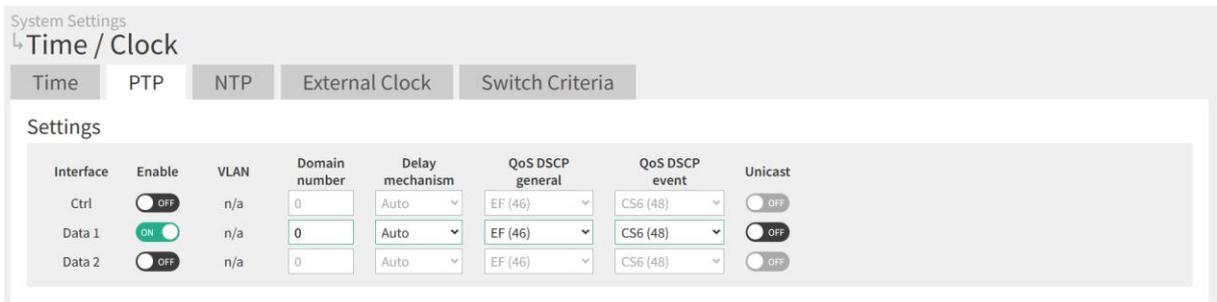

Changed Functionality

- 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.
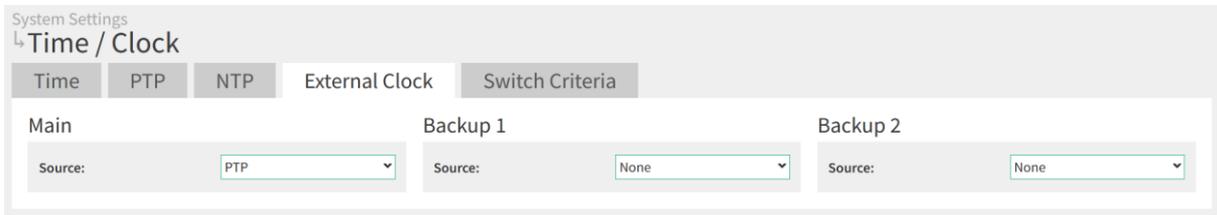
  All these configuration options are now consolidated into a single place – the menu "System Settings / Time/Clock":
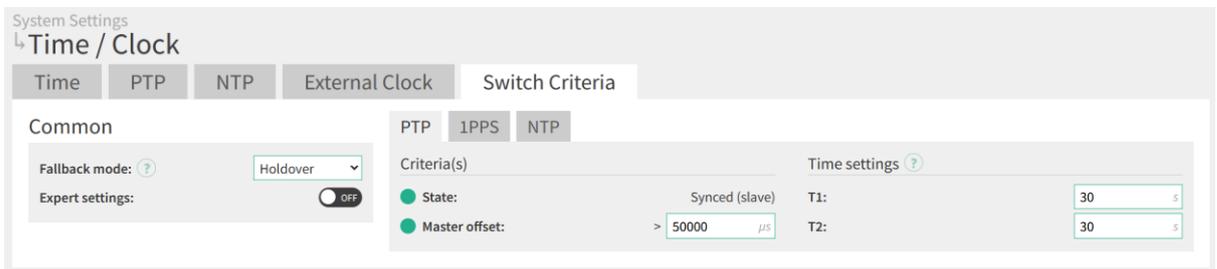

  You still have to configure via the "External Clock" tab (which corresponds to the former "AoIP Settings / External Clock" menu), if and which external clock source (PTP, NTP, 1PPS) should be used for the audio clock synchronization.
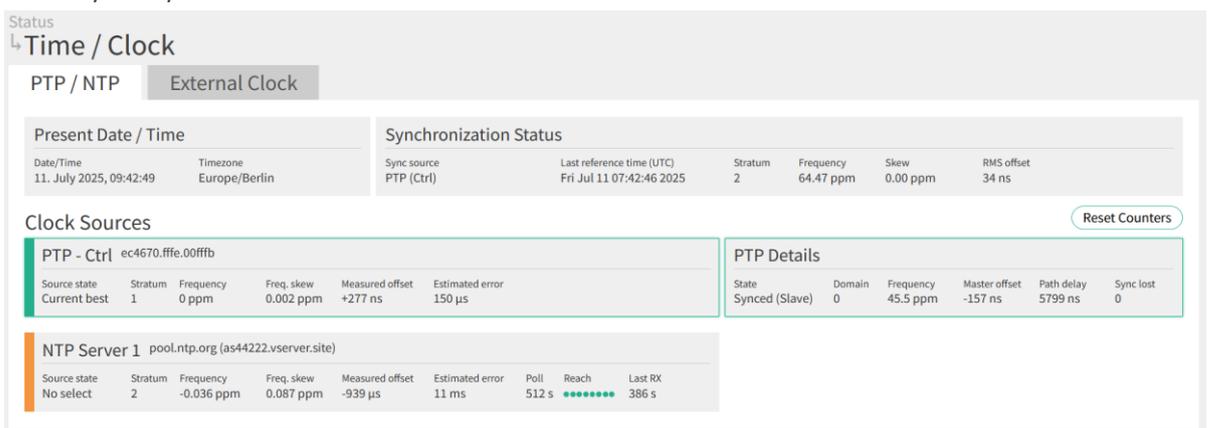
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If a backup for the external clock is configured (e. g. NTP), the switch criteria are now configured via the "Switch Criteria" tab of the "Time / Clock" menu.

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

- ASI output was not running stable
- AAC decoder: in case of bad input (due to e. g. stream interruptions) the audio level may change unintendedly when the decoding resumes (CSM-1074)
- Fix web interface error (showing "no space left on device"), even preventing login to web interface (CSM-1252)

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- 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)
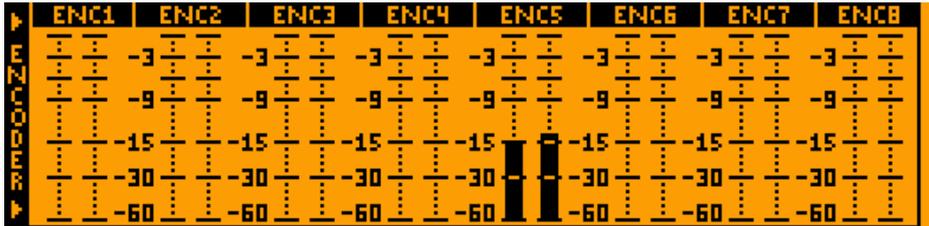
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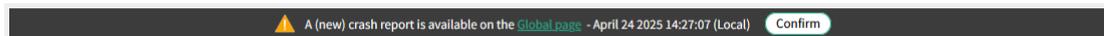
Version 2.17-rc1

09.05.2025

New Functionality

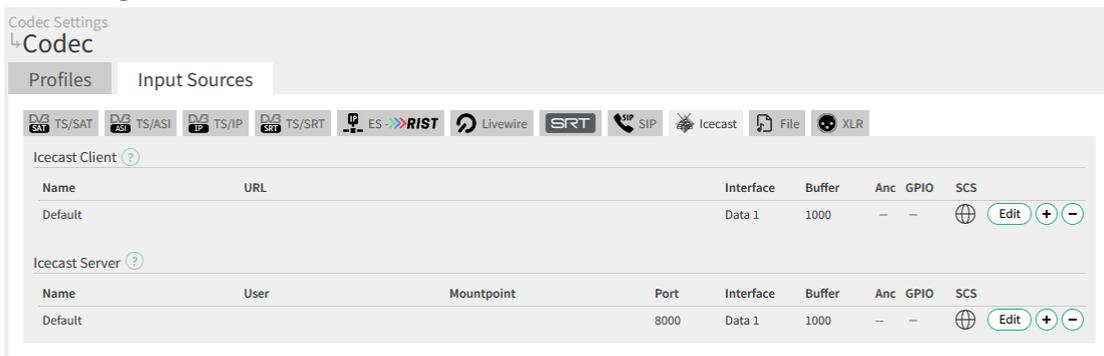- Added LCD screen(s) for the encoder input audio levels

- 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

Fixed Issues

- Headphone output not working with less than max. channel licensed (since 2.16.2)
- 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

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- 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

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