IP-4c Release Candidate Notes

Version 2.17-rc12

24.10.2025

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

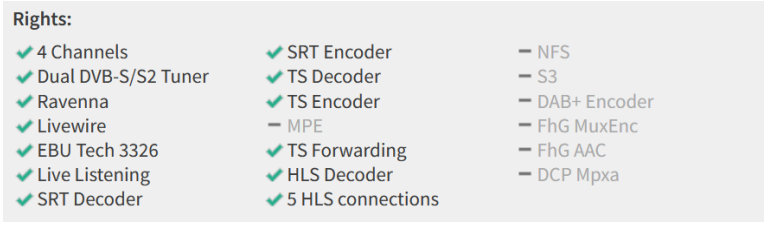
Version 2.17-rc12 3
Version 2.17-rc11 3
Version 2.17-rc10 4
Version 2.17-rc9 4
Version 2.17-rc8 5
Version 2.17-rc7 5
Version 2.17-rc6 5
Version 2.17-rc5 6
Version 2.17-rc4 6
Version 2.17-rc3 6
Version 2.17-rc2 7
Version 2.17-rc1 10

Version 2.17-rc12

24.10.2025

Changed functionality

- The list of present and missing licenses/rights on the System Settings / Global page is now presented in a clearer fashion



Fixed Issues

- Audio input set to AES67: audio buffer value of the AES67 input source was not applied when assigned to a decoder
- "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.

Version 2.17-rc11

06.10.2025

Fixed Issues

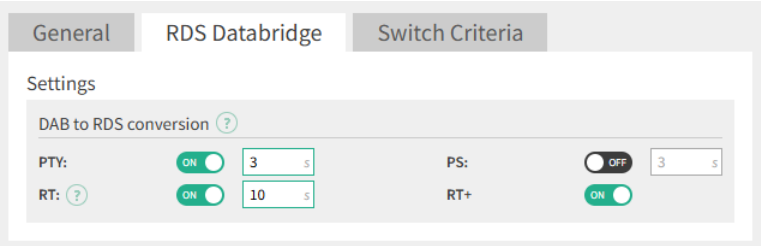
- The sample rate converter (SRC) was accidentally removed from the 4th audio input/output in the FPGA firmware (since 2.17-rc10)

Version 2.17-rc10

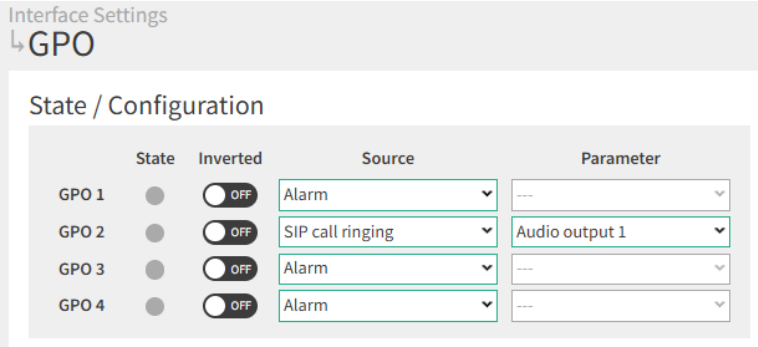
01.10.2025

New Functionality

- DAB tuner (optional): Added an RDS Databridge as UECP ancillary data provider from the DAB PAD data



- Added the possibility to trigger a GPO to signal a ringing SIP call in manual call acceptance mode



Fixed Issues

- PTP: small stability enhancements for the E2E mode
- PTP: added (better) support for the P2P mode
- Enhanced robustness of the optional SAT tuner inputs
- 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

Version 2.17-rc9

25.09.2025

Fixed Issues

- Ember+: Increased buffer sizes to prevent malformed packets when al lot of entries are subscribed (CSM-1308)
- Optional SAT tuner: fix wrong status on LCD screen for RF2 input (showing "Not configured")

- 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 audio outputs could have output errors (signal outages) when reset counters is activated
- 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)

Version 2.17-rc8

04.09.2025

Fixed Issues

- NMOS: Fix SDP information (manifest) still containing redundant stream information though redundancy is switched off
- NMOS: Use the correct clock information in the sender manifests (SDP)
- NMOS: Use the same AES67 output names as in SAP/SDP
- NMOS: Removed leftovers of the example code

Version 2.17-rc7

21.08.2025

Fixed Issues

- NMOS: Fix startup behaviour/configuration of the AES67 inputs

Version 2.17-rc6

19.08.2025

Fixed Issues

- Further enhancements to configuration changes done via NMOS (CSM-1329)

Version 2.17-rc5

15.08.2025

Fixed Issues

- After e. g. disabling/enabling an AES67 output the stream has an UTC/TAI offset of 37 seconds (CSM-1329)
- NMOS configuration enhancements
- After changing some AES67 output settings via NMOS they might be deactivated afterwards though they should still be enabled (CSM-1329)
- Further improvements to PTP synchronization stability

Version 2.17-rc4

12.08.2025

New Functionality

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

Fixed Issues

- The AES67 outputs could behave erratically when NMOS is enabled (CSM-1329)
- SIP call signalling via GPO only worked for audio output 1 (CSM-1313)
- The AES67 inputs only worked for a few seconds with "RTCP enabled" set to Off
- Wrong PTP state in external clock status block
- Optional TS Encoder: fix frame calculation for low bitrate overhead mode
- Optional 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
- Overview: optional FM tuner shows wrong RDS values if the source is in standby

Version 2.17-rc3

16.07.2025

Fixed Issues

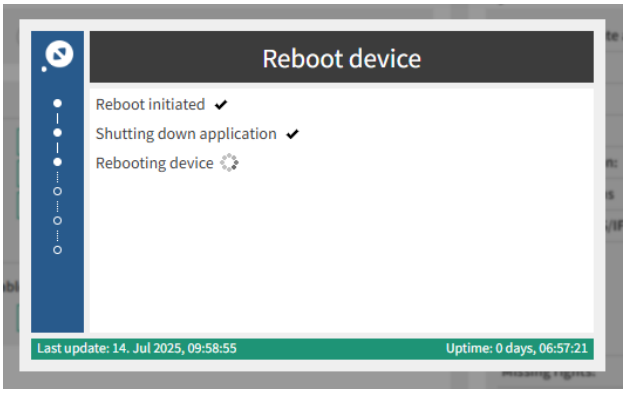
- The revised PTP implementation had a major problem likely selecting the wrong interface (and thereby the wrong PTP hardware clock) internally as the clock source

Version 2.17-rc2

15.07.2025

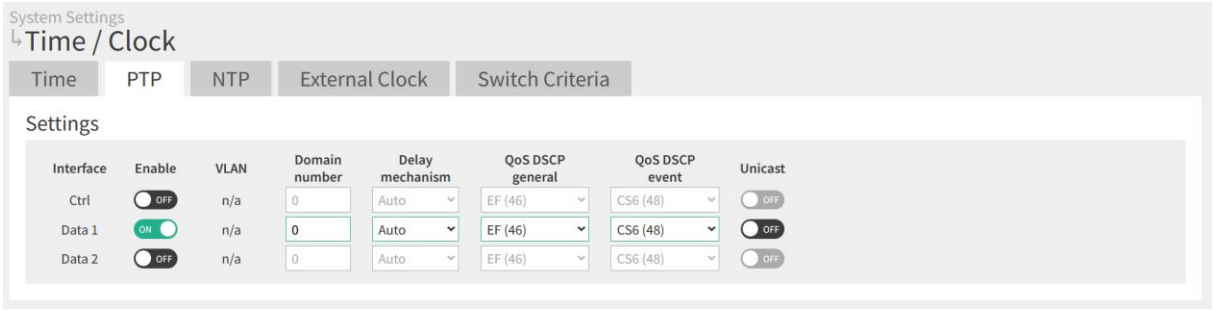
New Functionality

- 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
- Optional SAT tuner: added a second C/N alarm
- 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

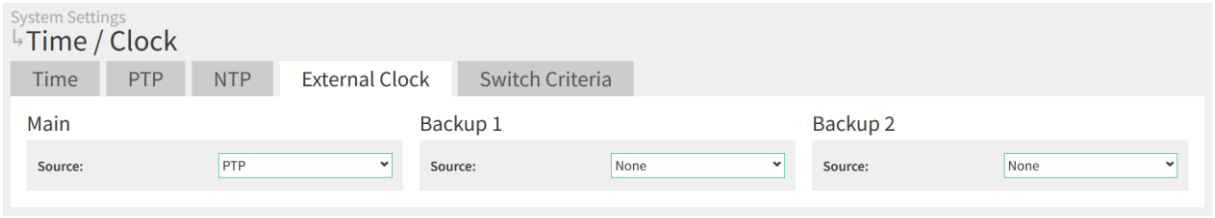


Changed Functionality

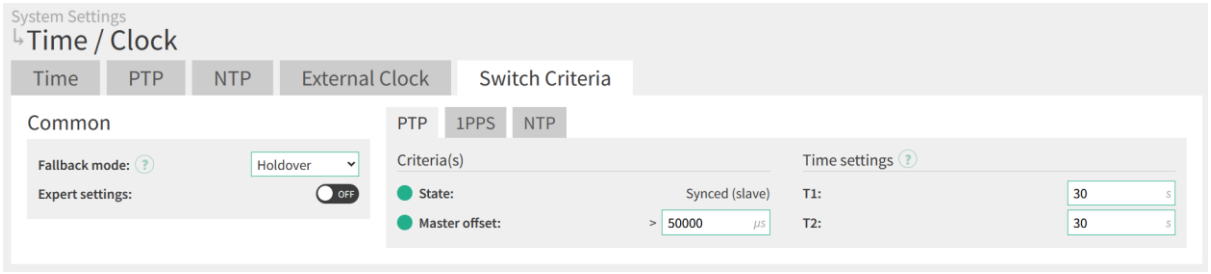
- The PTP functionality did get a complete revision, as it was not working reliably. In the course of these changes we did also revise the way to configure everything related to time and clock handling including NTP and external clock.
  Before this revision several menu options were involved in the 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 (including optional AES67 outputs).



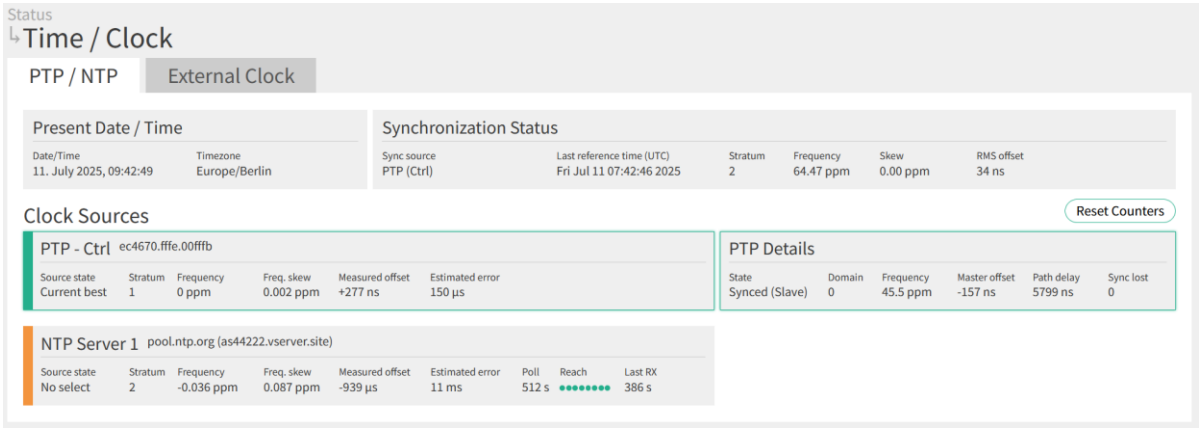
If a backup for the external clock is configured (e. g. NTP), the switch criteria are now configured via the “Switch Criteria” tab of the “Time / Clock” menu.



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":



- If an audio output is configured to be AES67 (instead of AES/EBU or Analog), the AES67 output stream can now be enabled/disabled via the web interface (before the AES67 output stream was always enabled when the audio output was switched to AES67 and couldn't be disabled via the web interface)

Fixed Issues

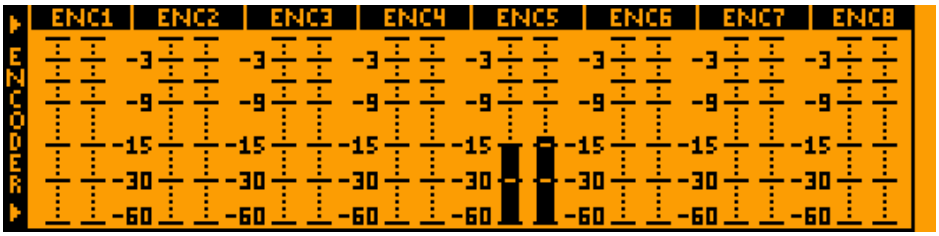
- AAC decoder: in case of bad input (due to e. g. stream or SAT reception interruptions) the audio level may change unintendedly when the decoding resumes (CSM-1074)
- Fix web interface error (showing "no space left on device"), even preventing login to web interface (CSM-1252)
- The individual gain 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)
- Fix high AES67 output jitter
- Fix AES67 output time jumps when decoder is stopped/started
- 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.

Version 2.17-rc1

09.05.2025

New Functionality

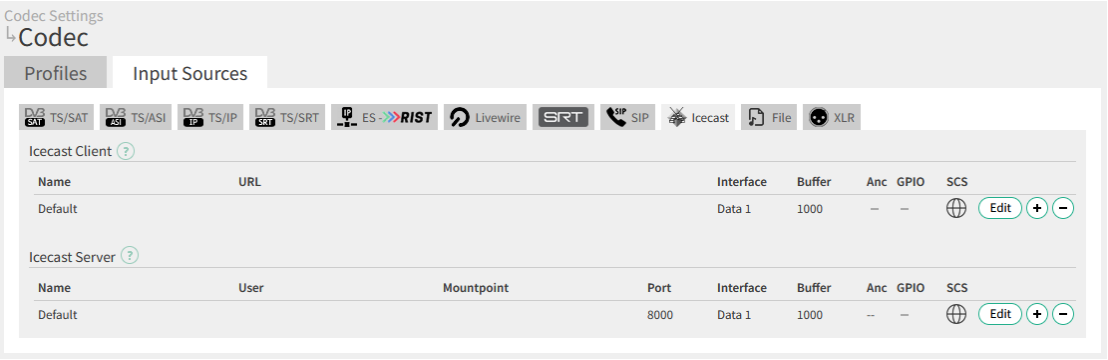
- Added a LCD screen for the encoder input audio levels



- 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

Changed Functionality

- Easy2Connect: Show yellow button in case of incoming SIP call and call acceptance mode set to “Manual”

Fixed Issues

- Headphone output not working with less than max. channel licensed (since 2.16.1/2.16/2.16-rc7)
- 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
- Synchronous playout SPN based on NTP may be broken (off by some seconds) when the XLR inputs are used as the input source
- 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