SAT-4da

Release Notes

Version 1.06.2

18.12.2025

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

SAT-4da Release Candidate Notes

Version 1.06.2

18.12.2025

🎄 Christmas present

- LiveListening is now included free of charge!

Fixed Issues

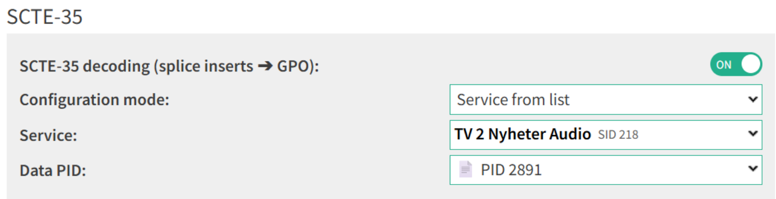
- Security fix according to BSI recommendation
- VLAN not selectable for SNMP Trap Manager destinations (CSM-1476)
- Periodic reboots due to a memory leak in the Pro-MPEG FEC decoder (CSM-1381, since firmware 2.16-rc7)
- Fix possible crash report generation after doing a firmware update or a reboot (CSM-1499)
- TS stream info incomplete, or not shown at all on Overview page (this bug was introduced in 2.17.2)
- AAC audio decoding from TS may start severely delayed (CSM-1482)

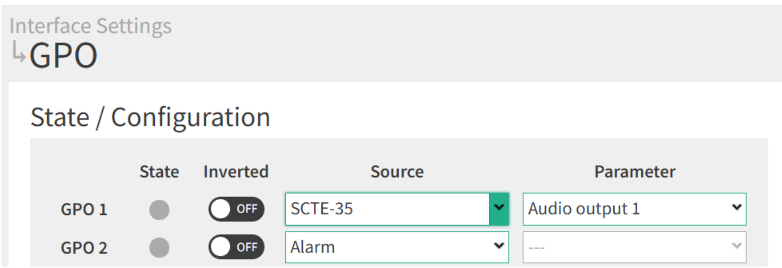
Version 1.06.1

02.12.2025

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.





Fixed Issues

- 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

SAT-4da Release Candidate Notes

Version 1.06

18.11.2025 – identical to 1.06-rc10

New Functionality

1.06-rc6

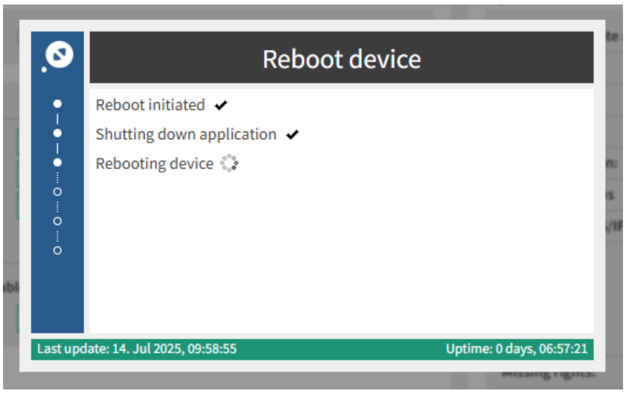
- Added support for new Dual DVB-S/S2/S2X Tuner
- The optional TS Forwarding does now also allow to forward the signal from the ASI input via IP (in addition to the TS from one of the two SAT RF inputs)

1.06-rc5

- The optional TS MPE Forwarding can now be used in combination with all available TS input sources (including TS/IP) and is no longer limited to the TS from the satellite tuner
- Added Ancillary Data Output alarms

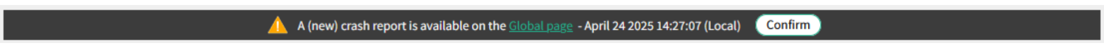
1.06-rc2

- 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
- 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



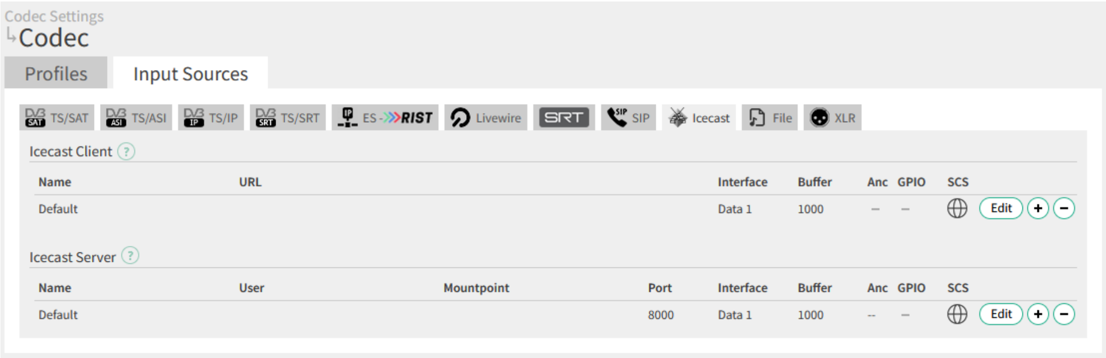
1.06-rc1

- 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

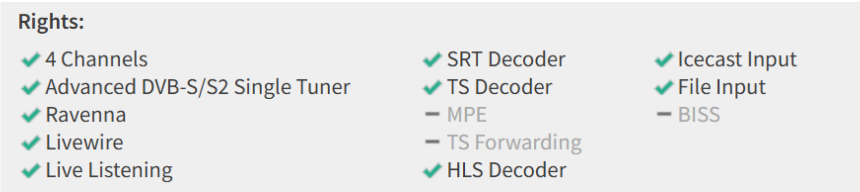


- Added PTP QoS DSCP settings

Changed functionality

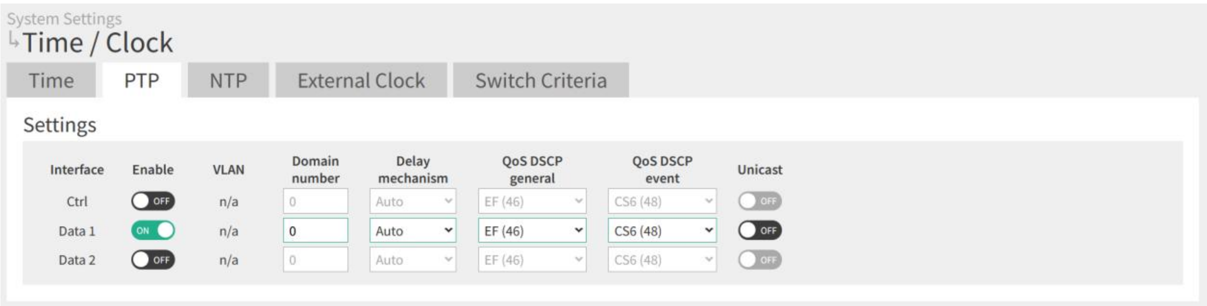
1.06-rc7

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

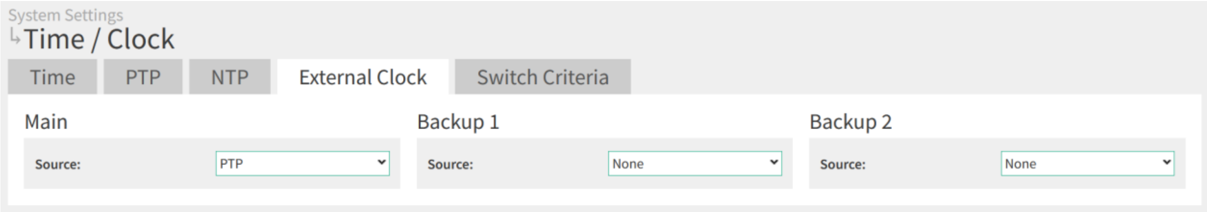


1.06-rc2

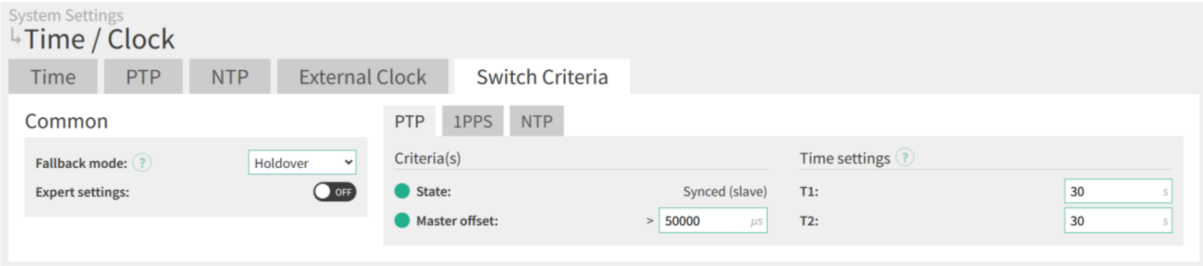
- The PTP functionality did get a complete revision, as it was not working reliably. In the course of these changes we did also revise the way to configure everything related to time and clock handling including NTP and external clock.
  Before this revision several menu options were involved in the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

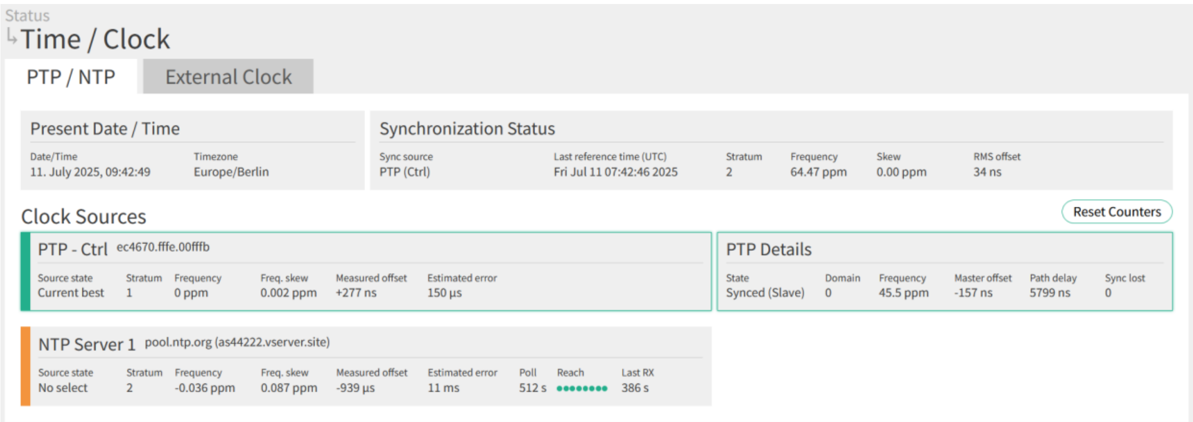


If a backup for the external clock is configured (e. g. NTP), the switch criteria are now configured via the “Switch Criteria” tab of the “Time / Clock” menu.



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”:



- 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

1.06-rc10

- 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)

1.06-rc9

- The new Dual DVB-S/S2/S2X Tuner might have a TS bitrate of 0 after a reboot, although the tuner has a lock
- RIST requested counter no longer resets after timeout and stream resume

1.06-rc8

- 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)
- Fixed wrong (generic) file name in log entry for settings upload/activation
- Fixed possibly empty VLAN select box (CSM-1423)

1.06-rc7

- "Timed out" counter was not reset on "Reset counters"
- "Missed" counter is no longer reset on stream resume after timed out input stream
- 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.

1.06-rc6

- PTP: small stability enhancements for the E2E mode
- PTP: added (better) support for the P2P mode
- Fixed a problem with the RF2 input in combination with high symbol rates
- 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

1.06-rc5

- Fix a problem with the RF2 satellite tuner input in combination with high symbol rates not correctly passing the TS stream to the TS decoder (CSM-1345)
- Fix wrong status on LCD screen for RF2 input (showing "Not configured")
- 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 audio outputs could have output errors (signal outages) when reset counters is activated
- The Icecast client may get stuck in a redirect, not being able to re-establish the stream decoding (CSM-1358)

1.06-rc4

- After e. g. disabling/enabling an AES67 output the stream has an UTC/TAI offset of 37 seconds

- Further improvements to PTP synchronization stability
- Wrong PTP state in external clock status block
- No private/ancillary data output after reboot
- No displayed source on UDP Ancillary Output after select and save
- When changing the width of one of the log tables the page got distorted
- 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

1.06-rc3

- 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

1.06-rc2

- 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

1.06-rc1

- Headphone output not working with less than max. channel licensed (since 1.05/1.05-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
- General Icecast client compatibility enhancements
- Changing the input source gain of a file input source was not applied to all instances

SAT-4da Release Candidate Notes

Version 1.05

13.03.2025 – identical to 1.05-rc10

New Functionality

1.05-rc10

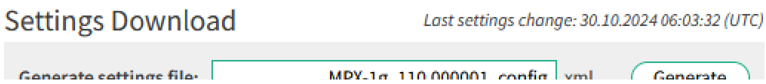
- The PTP delay mechanism to use is now configurable

1.05-rc7

- Added a new button to the Overview page to reset the counters of all decoders

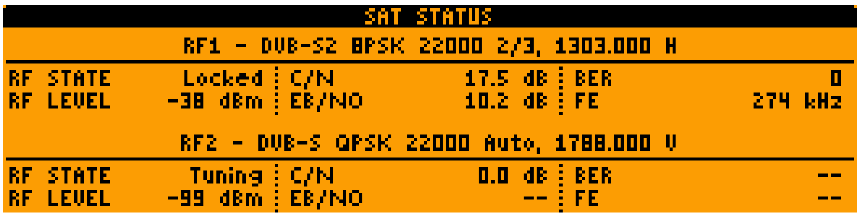
1.05-rc6

- Added a configuration option to file input sources to play only once (no loop)
- 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:

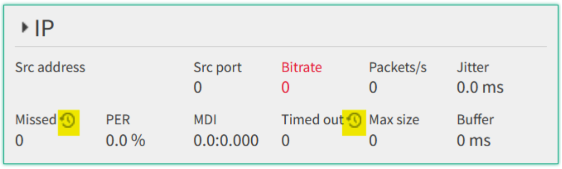


1.05-rc5

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

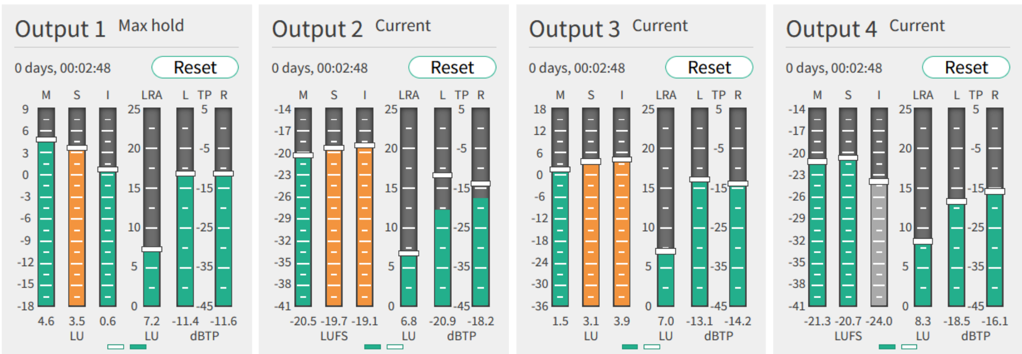


- Started to add the possibility to get a counter history on the Overview page via a small icon next to the label. It's more or less a shortcut to the extended log, showing you only the entries for the counter of this input source. Currently it's added to the "Missed" and "Timed out" counter, others should follow.



1.05-rc2

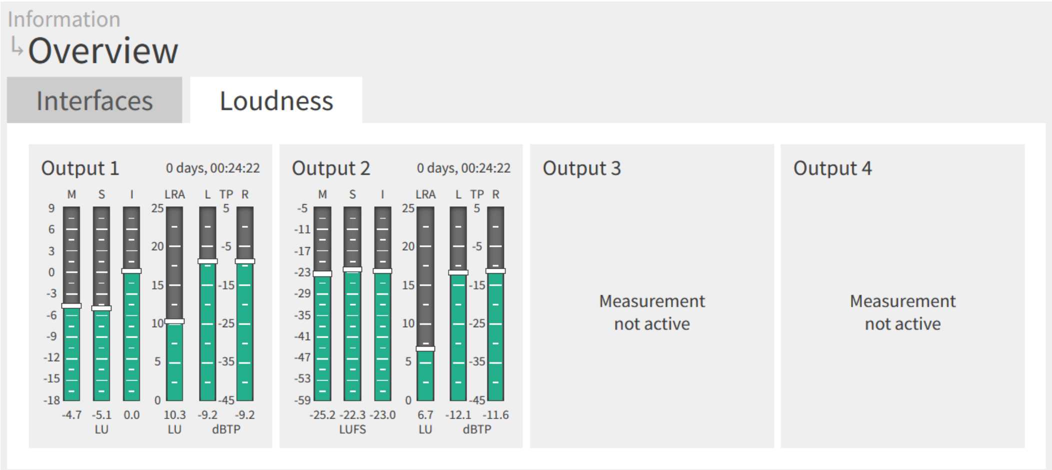
- Enhanced the loudness metering to show either the current values or the max hold values



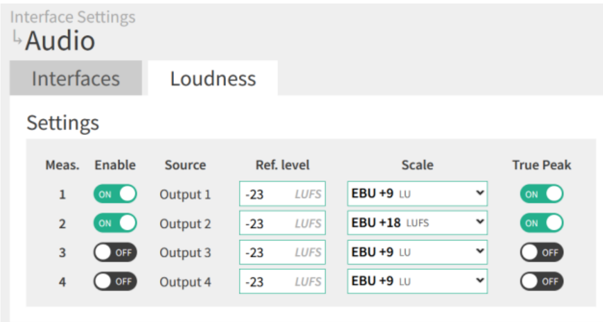
- New silence mode for elementary stream inputs: when enabled, empty packets will be inserted in case of missing stream input (like the “Generate Null samples” option for AES/EBU inputs, when no input is connected
- Added the possibility to clear the extended log

1.05-rc1

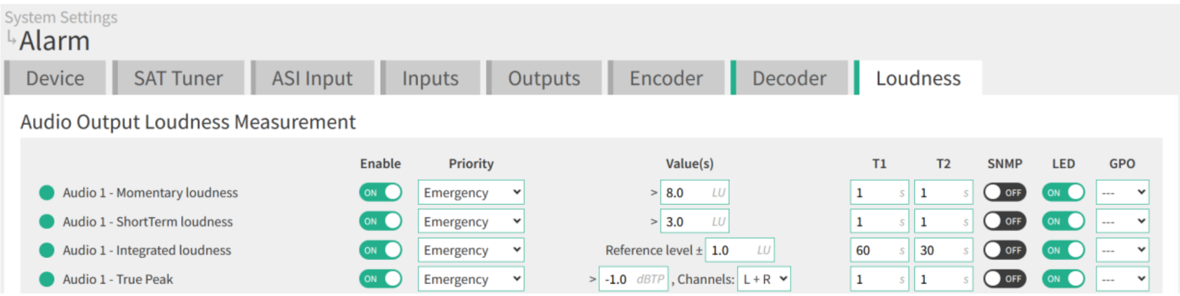
- Added optional Loudness metering/monitoring according to EBU R 128 for the audio outputs



The reference levels and scales can be configured individually via Interface Settings / Audio:



Additionally there are of course new alarms for the loudness monitoring:



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

Changed functionality

1.05-rc9

- In the 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.

1.05-rc7

- 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

1.05-rc2

- The optional BISS descrambling was not applied to the optional TS Forwarding, leaving the forwarded TS scrambled. That's changed now – the forwarded TS is now descrambled, if the BISS descrambling is activated.

1.05-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

1.05-rc10

- MPE decoding might lead to device crashes since firmware 1.05-rc7
- Improved AES67 output in case PTP is activated
- The silence mode available for elementary stream input still had some problems when enabling/disabling it
- Icecast client: enhanced compatibility to streams providing MPEG Layer 2 audio data
- Fix a possible application hangup when loading settings (leading to recovery page)
- DAB+ Encoder: local PAD insertion for DAB DCP output had some problems when inserting DLS+ tags
- Fixed a problem with the diagnostic report not including all relevant information

1.05-rc9

- 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
- 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 1.05-rc7)
- PTP for unicast configuration improved
- Enhanced compatibility of the Icecast client in case of FLAC Icecast server content

1.05-rc8

- Fix broken SAT tuner status in case of EXM02 (advanced low bitrate single tuner)
- Security enhancements

1.05-rc7

- Bunch of internal improvements and optimizations
- TS Decoder couldn't handle some special character encodings in the service list
- Headphone output may be distorted
- 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
- 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
- TS/SRT decoding was broken
- 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

1.05-rc6

- 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
- 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

1.05-rc5

- 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
- Icecast client: https Icecast streams with credentials may have problems with authentication
- (Limited) support for Internet Explorer 11 was broken

1.05-rc4

- Switch criteria C/N for SAT does not work

1.05-rc3

- General playout stability improvements
- The descrambled TS Forwarding had still the scrambling control bits set

- 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
- 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

1.05-rc2

- Optional MPE Forwarding does not resume after uplink problem or headend switch
- 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)

1.05-rc1

- The 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
- 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)

Version 1.04.2

13.06.2024

Fixed Issues

- The AES67 outputs may exhibit timestamp jumps in certain cases
- The optional Dolby decoder did not work correctly and produced audio glitches
- Fixed some problems with the optional Live Listening feature

Version 1.04.1

07.06.2024

Changed Functionality

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

Fixed Issues

- PTP might not work correctly in certain environments
- PTP might not be active after loading a settings file with a corresponding configuration
- Changing the SNMPv3 authentication protocol (MD5 <-> SHA1) did not work
- After IP address change the SRT decoder (in Listener mode) did not receive any SRT streams
- Headphone Downmix did not work in Analog/Digital Mode

Version 1.04

31.05.2024 – identical to 1.04-rc5

New Functionality

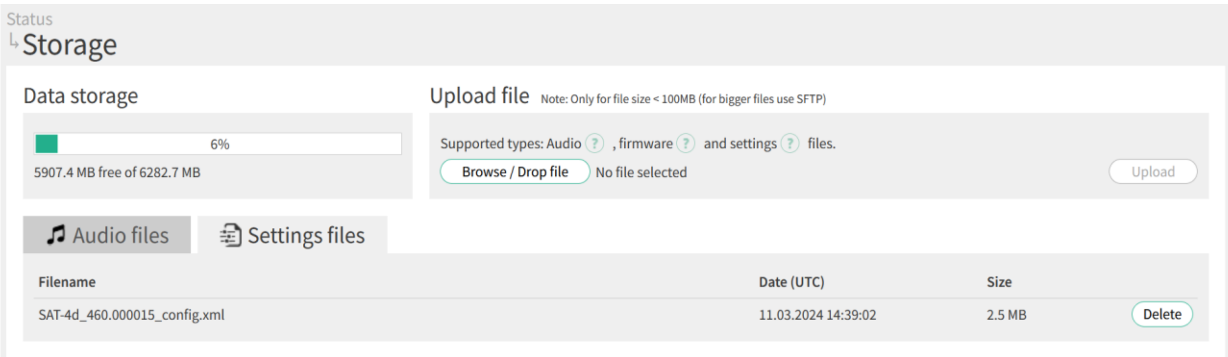
1.04-rc4

- Added parity configuration for DTE outputs

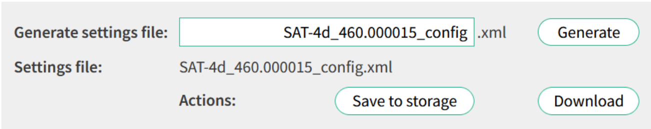
1.04-rc3

- Added traceroute possibility to the web interface (Network Settings / TCP/IP / Tools)

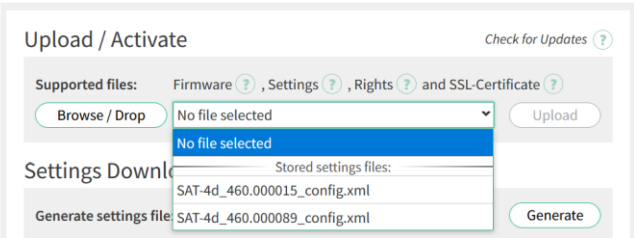
- Added event log entries when loading a settings file or loading factory settings
- Settings files can also be uploaded via the Storage page



- 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.

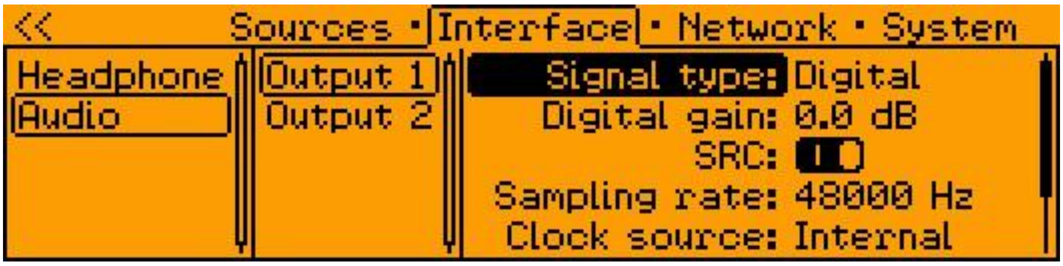


- Once there are settings files on the internal storage the user can select one of the stored files instead of uploading one and load it via the new consolidated Upload/Activate section, thereby having a sort of preset functionality, allowing to quickly switch between different configurations.



1.04-rc2

- Added ping possibility to the web interface (Network Settings / TCP/IP / Tools)
- Added audio output configuration to the LCD menu



1.04-rc1

- 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

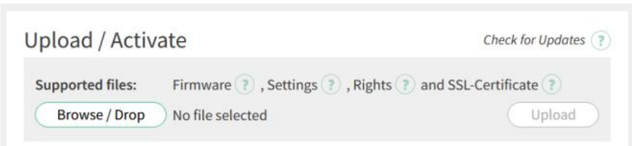
Changed Functionality

1.04-rc4

- 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
- Updated Fraunhofer libraries to latest versions

1.04-rc3

- 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



1.04-rc1

- Cleanup of SRT / TS/SRT Overview info block fields
- Reduced sensitivity of the jog wheel for better user experience when doing configurations via the LCD menu
- Changed Load Factory Settings via LCD menu to new method without reboot
- 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

1.04-rc5

- 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 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 AES67 output did not work, when no External Clock was configured (since firmware V1.04-rc4)

1.04-rc4

- 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 - Fixed
- 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
- 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 1.04-rc1) -> the minimum configurable audio buffer has been increased to 100ms
- 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
- When switching between different elementary streams (e. g. Ravenna streams) the device may crash and gets unresponsive
- General security and access control improvements

1.04-rc3

- The device could get blocked when configuring the SAT input source via the LCD menu
- Again: Improved AES67 output PTP synchronization
- 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

1.04-rc2

- LNB local oscillator frequency was not configurable via the LCD menu for the SAT input sources when the frequency input method was set to “Transponder Frequency” (but instead if the method was set to “L-Band Frequency”)
- Reduced sluggishness/lag of the LCD menu handling once a service to demultiplex is configured via the service name
- 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

1.04-rc1

- Changing just the audio buffer level of an input source was 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
- Corrected available SAT FEC options in LCD menu config
- Fixed problems with configuring the TS input sources via the LCD menu
- 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)
- When doing “Load Factory Settings” and unplugging the power immediately after completion, the new (default) settings might not have been saved
- For C band transponders the minimum allowed transponder frequency was too high, not allowing to configure some transponder frequencies
- 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

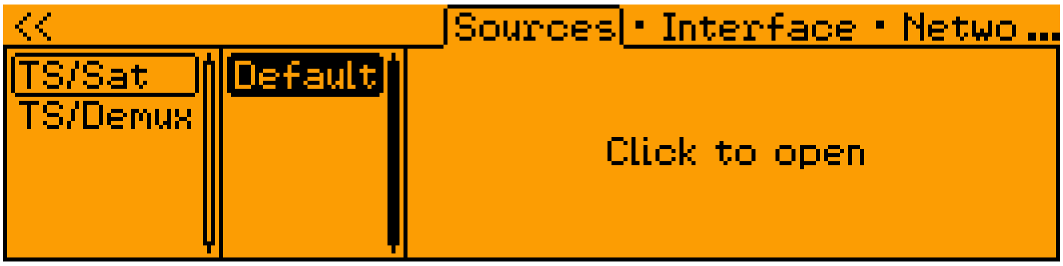
Version 1.03

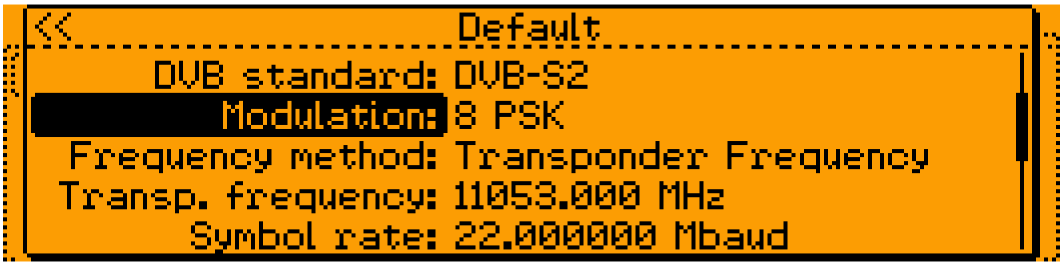
02.02.2024 – identical to 1.03-rc9

New Functionality

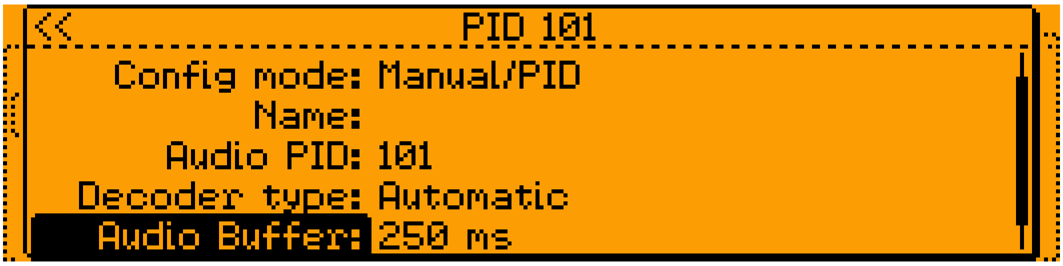
1.03-rc6

- Added the possibility to change the TS/Sat configuration via the LCD menu





- Added the possibility to change the TS/Demux configuration via the LCD menu



1.03-rc4

- Added missing VLAN support to SRT 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 possibility to update the firmware locally from USB
- Added the fan status to Status→Device in the web interface (needs System Controller firmware 1.07)

1.03-rc3

- When loading settings via the web interface it can now optionally include the TCP/IP configuration
- 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.

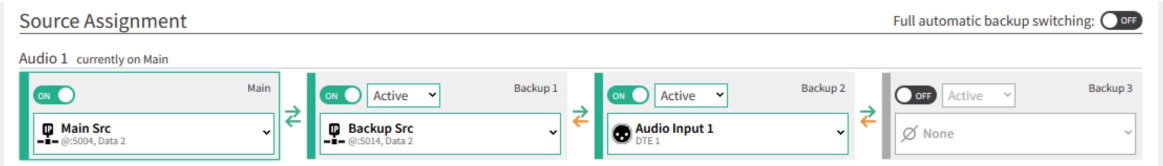
1.03-rc2

- SAT Tuner: added the frequency offset to the status parameters

- SRT: added source address and port as well as uptime (time since last connection start) to the decoder status information

1.03-rc1

- Added the possibility to configure individual switch criteria per input source
  Instead of having one global switch 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 the possibility to locally load settings from and save settings to USB via the LCD menu.
  The settings loading from USB can be done in two different ways – one without and one including the TCP/IP configuration.
- 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

1.03-rc4

- The individual gain adjustments per input source can now be done with a resolution of 0.01 dB steps
- “Load factory settings” from within the web interface does no longer require a reboot

1.03-rc1

- Loading settings via the web interface will no longer require a reboot of the device, but will be done very quickly (within a few seconds)

Fixed Issues

1.03-rc9

- Sample rate converter could not be configured for the AES/EBU audio outputs
- AUX1 could not be configured between ASI input / 1PPS input

1.03-rc8

- When doing “Load Factory Settings” some settings may not be back to default, but only after a reboot
- When uploading and activating a settings file some settings may not be activated correctly and may need a reboot
- Icecast streams with FLAC codec did not work
- Audio on headphone may be distorted if listening to an audio input with disabled SRC
- 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 none working 1PPS input status since firmware 1.03-rc7

1.03-rc7

- Fixed potential high amount of audio errors due to FPGA audio buffer underrun
- Fixed visual problems with the TS Demux service select update/refresh
- SAT tuner alarms were swapped between RF1 and RF2

1.03-rc6

- The fix for the SAT tuner (done in 1.03-rc5) was badly wrong and let to cyclic audio interruptions if only one of the two RF inputs is used and tuned. In fact that fix was not necessary at all – the dual tuner (EXM01) is re-establishing the tuner lock without a problem.
- 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 optional 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

1.03-rc5

- A bad combination of two changes introduced in firmware 2.14-rc5 (migrating password hashes from MD5 to bcrypt and doing “Load factory settings” without reboot) could lead to the impossibility of being able to login to the web interface after having done “Load factory settings”.
  A reboot would fix this problem, but one might not be able to initiate the reboot due to the impossibility to login to the web interface.
- 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
- SAT tuner: if reception gets interrupted due to e. g. bad weather conditions for a longer period of time, the device might fail to re-establish tuner lock (with the normal dual tuner EXM01)

1.03-rc4

- Saving settings to USB may fail, if the USB stick is removed too quickly after the OK message is shown
- Fix visible name field in TS/Demux configuration when in “Service from list” mode
- Device could hang up when the audio file belonging to a file input source is not present (e. g. after loading a settings file)
- The new individual switch criteria did not work correctly (were not applied correctly)

- Synchronized Playout did not work correctly without SFN license
- 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
- 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

1.03-rc3

- Configuration changes done in the Codec dialogs may not get saved, when the dialog is left while the new (individual) Switch Criteria tab is selected
- 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
- If a decoder does provide ancillary data this data is now also shown on the Overview page if the input source is not the currently active one

1.03-rc2

- Optional MPE Forwarding: TS source selection was unavailable since firmware 1.03-rc1
- 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

1.03-rc1

- 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
- File playback from playlist was broken since latest firmware 1.02
- VLAN interfaces were not correctly activated after a firmware update
- When updating the recovery bundle the wait page was shown endlessly
- FEC: Fix audio frames being unnecessarily recovered in case of 1x4 matrix

Known bugs

- xHE-AAC: Ancillary data and GPIO Forwarding does not work

Version 1.02

08.05.2023

New Functionality

- Added a new status value providing the delta time between the two streams in RTP DualStreaming setup
- Added compatibility to Qbit GPIO Forwarding in ancillary data
- Added Mono Downmix option to headphone output
- 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
- Added Mono Downmix option to headphone output
- 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 and RTP Rx timeout 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.
- Released REST API that enables customers to use an Open API 3.0 compliant API to control and monitor 2wcom's devices.
  - More info about Open API 3.0 can be found here.
  - The API can be enabled and browsed on page "External APIs" -> "REST API".
  - The openapi.json for the device and additional documentation can be found on the devices page "External APIs" -> "REST API".
- Added alarm/monitoring for the ASI input
- Added interface (and VLAN) selection to the SNMP trap manager configuration

Changed functionality

- Reported Livewire GPI/GPO states are now high in default state and low in triggered state
- SRC/DST name size increased for LWRP from 15 to 25 characters
- 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

- Fix a memory leak with enabled TS MPE Forwarding, leading to periodic reboots

- Invert setting in GPO configuration had no influence on reported GPO state via Livewire
- RTP dual streaming improved
- SAT Tuner: Fixed BER calculation for low symbol rate single tuner
- SAT Tuner: Added configuration option for PLS and ISI (in case of DVB-S2)
- Improved stability (jitter) of AES67 output
- 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
- Digital interface output level can be displayed greater than 0 dBFS when a gain value > 0 is configured and the audio level is high
- Drag&Drop of TS Data Demux input sources did not always work
  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
- Fixed Livewire routing protocol LWRP not working after changing the IP address of the device
- MPE decoder: MPE ancillary data output was not working
- Fixed possible timestamp display problems in new Extended Log
- 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
- Changes to the headphone config via web interface were not applied
- 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)
- 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
- 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
- Changing just the LNB config for a SAT input source was not applied
- Optional Live Listening feature did not work with Safari browser
- 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

- xHE-AAC: Ancillary data and GPIO Forwarding does not work

Version 1.01

08.02.2023

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
- Added previously missing RF level value for the optional SAT tuner
- Added SAT tuner alarms for RF level, C/N and TS Sync
- Added Mono Downmix option for the audio outputs
- Major Livewire integration enhancements:
  - Added 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.
  - Added Livewire level meters for Sources and Destinations (enhance compatibility to e. g. Pathfinder)
  - 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 SAT-4d (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.
- Added status tabs to Overview page for optional AES67 outputs and Livewire Sources
- Added PIN lock option for LCD menu
- 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.
- The “NTP Server Quality” parameter options are renamed from “Internet” and “Local” to “Logging” and “Clock Source” to better describe the usage and thereby the internal evaluation parameters used. With NTP expert settings enabled you can now also change the NTP quality settings “RMS Offset” and “Skew” (used as evaluation parameters).
  NTP validation is now much faster on device startup or NTP activation.

Fixed Issues

- Fixed a possible crash with NTP synchronization and “Bind to interface” option enabled
- When loading a settings file, some input sources may not show up, although there were such input sources in the settings file (probably only TS/SAT)
- SAT tuner: there was a LNB supply limitation of 200mA, which led to compatibility issues to some LNBs (not switching polarization). This limitation is removed by software, allowing for a current > 400mA

- SAT input sources: Fixed a possible larger offset of the audio buffer level with certain streams
- 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 input source assignment check for SAT input sources (did not allow some valid combinations)
- Fix xHE-AAC problems with low bitrates
- Icecast input source handling improved for faulty meta data from some Icecast servers
- Improved Icecast client compatibility to “bursty” Icecast streams similar to HLS
- 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
- 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 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)
- TS/Demux: In "Service (auto)" mode the PID shown in the input source table was the PMT PID and not the audio PID
- TS/Demux: Fix decoding startup problems with “Service (auto)” mode
- TS/Demu Config: Fix empty audio track select on config mode change from “Service (auto)” to “Service (fixed)”
- SAP: fixed crash with too many announced streams

Known bugs

- xHE-AAC: Ancillary data and GPIO Forwarding does not work