IP-4c

Release Notes

Version 2.17.3

18.12.2025

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Table of Contents

IP-4c Release Notes

Version 2.17.3

18.12.2025

🎄 Christmas present

- LiveListening is now included free of charge!

Fixed Issues

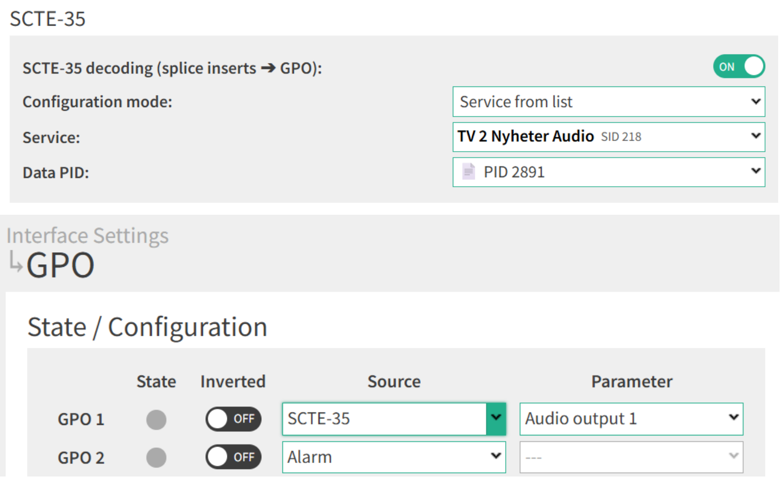
- Security fix according to BSI recommendation
- VLAN not selectable for SNMP Trap Manager destinations (CSM-1476)
- Periodic reboots due to a memory leak in the Pro-MPEG FEC decoder (CSM-1381, since firmware 2.16-rc7)
- Fix possible crash report generation after doing a firmware update or a reboot (CSM-1499)
- 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 2.17.2

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

Version 2.17.1

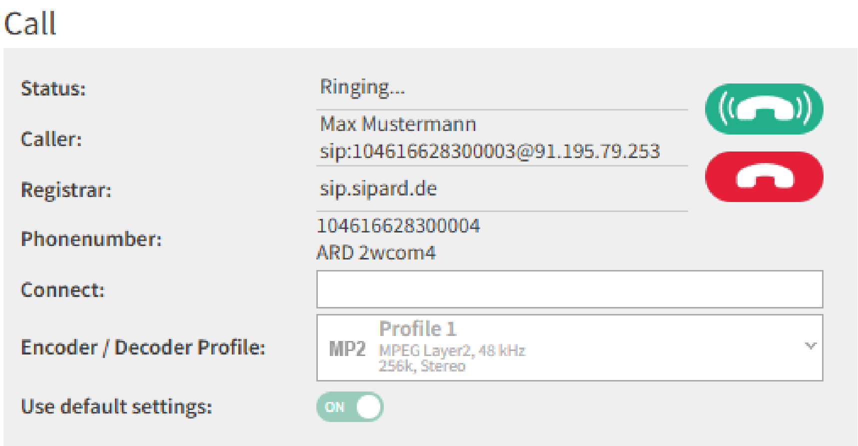
18.11.2025

New Functionality

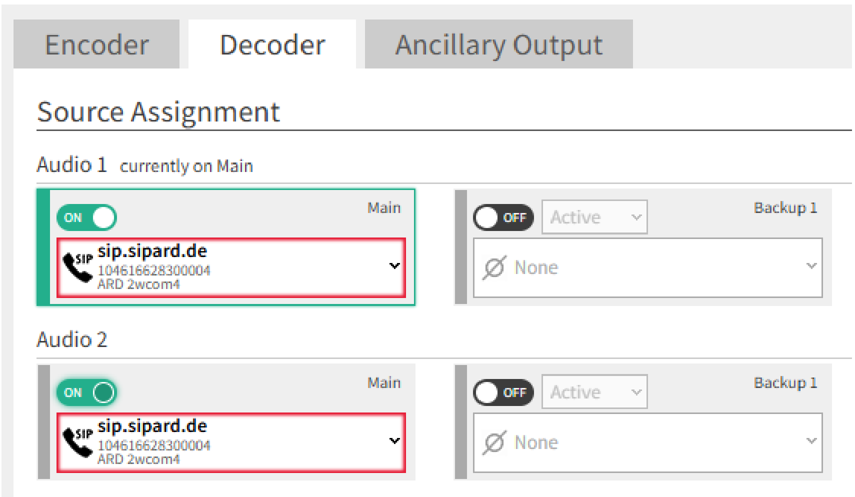
- Added the possibility to limit the bandwidth used by SRT

Changed functionality

- Easy2Connect: In case of an incoming SIP call and call acceptance mode set to “Manual”, the Easy2Connect page will now allow to accept or decline the call. It does now also show information about the caller.



- SIP: Each SIP input source can only be assigned to a single decoder to assure the unambiguous assignment to the audio input used for the back channel. To prevent invalid configurations, multiple assignments of the same SIP input source are marked as faulty and the configuration cannot be saved.



Fixed Issues

- 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)

- Audio buffer is running empty in case the audio output is switched to AES67 or Livewire (CSM-1453)
- GPO source "SIP call ringing" did only trigger one GPO even when two or more GPOs are configured identically
- RIST requested counter no longer resets after timeout and stream resume

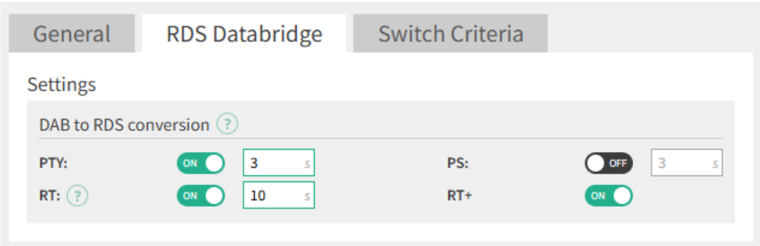
Version 2.17

29.10.2025 – identical to 2.17-rc13

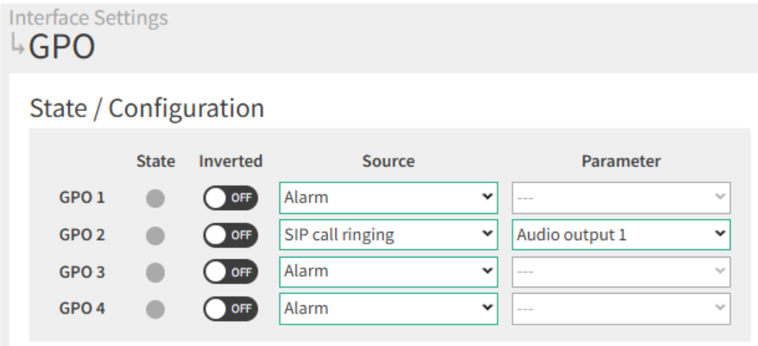
New Functionality

2.17-rc10

- Added support for new Dual DVB-S/S2/S2X Tuner
- 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



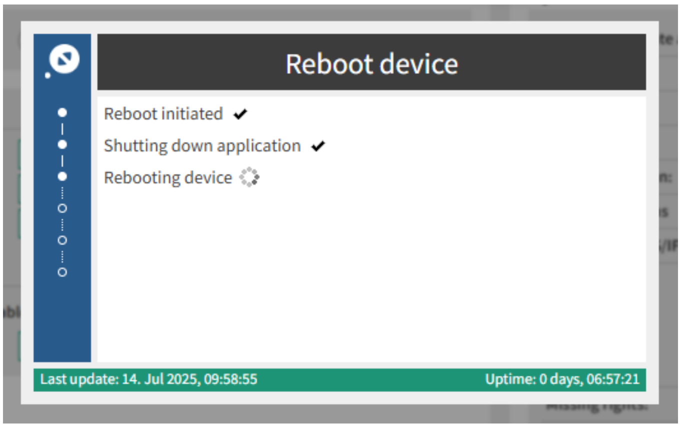
2.17-rc4

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

2.17-rc2

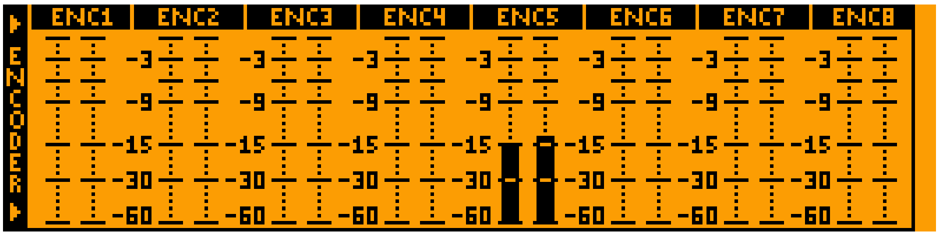
- The possible actions that can be triggered by a GPI are enhanced by two new options, which allow to enable or disable a decoder input source
- Optional SAT tuner: added a second C/N alarm
- For the optionally available SDP files on the Status/Storage page it is now additionally possible to copy the 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

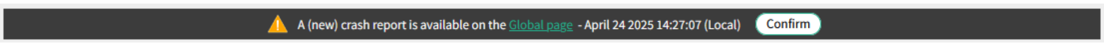


2.17-rc1

- Added a LCD screen for the encoder input audio levels

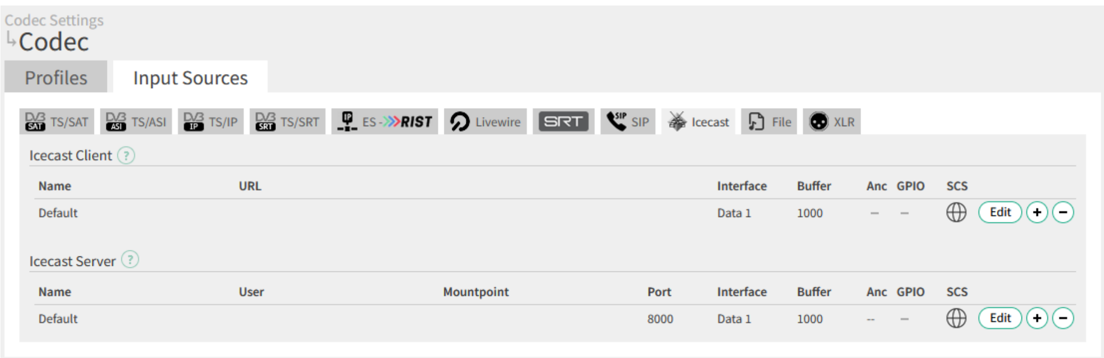


- Added new crash report functionality
  In case the device does crash (performing an unintended reboot) it was until now very difficult to find the root cause for the crash/reboot. In such a case a crash report will now be generated which should be sent to us via the 2wcom Support Center. It will allow 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

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Changed functionality

2.17-rc12

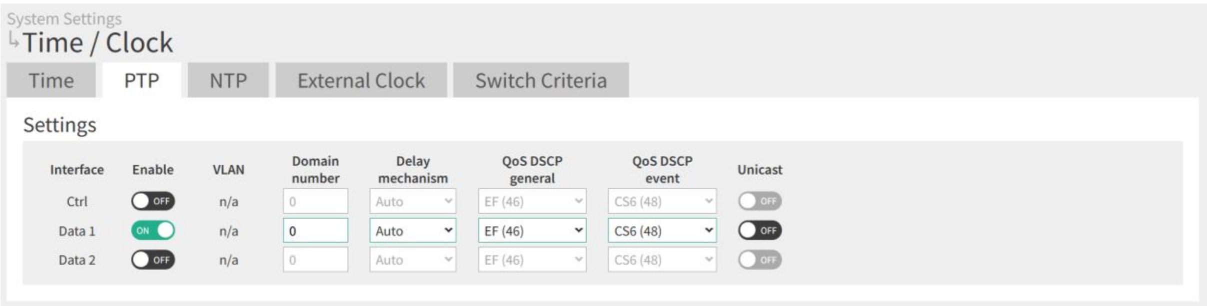
- The list of present and missing licenses/rights on the System Settings / Global page is now presented in a clearer fashion

Rights:

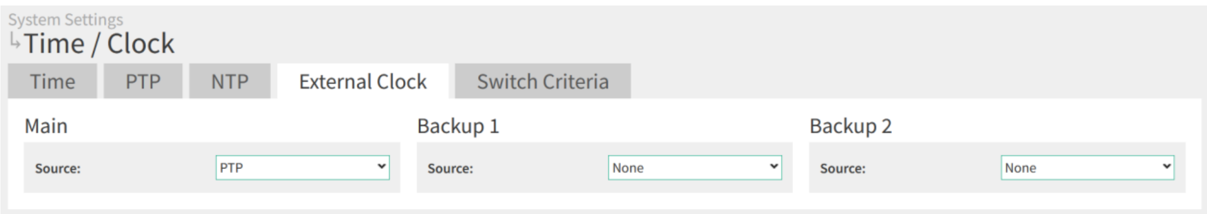
✔ 4 Channels	✔ SRT Encoder	– NFS
✔ Dual DVB-S/S2 Tuner	✔ TS Decoder	– S3
✔ Ravenna	✔ TS Encoder	– DAB+ Encoder
✔ Livewire	– MPE	– FhG MuxEnc
✔ EBU Tech 3326	✔ TS Forwarding	– FhG AAC
✔ Live Listening	✔ HLS Decoder	– DCP Mpxa
✔ SRT Decoder	✔ 5 HLS connections	

2.17-rc2

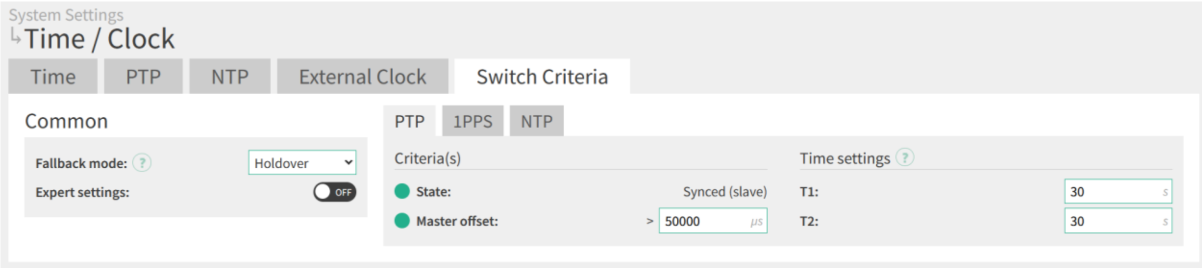
- The PTP functionality did get a complete revision, as it was not working reliably. In the course of these changes we did also revise the way to configure everything related to time and clock handling including NTP and external clock.
  Before this revision several menu options were involved in the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

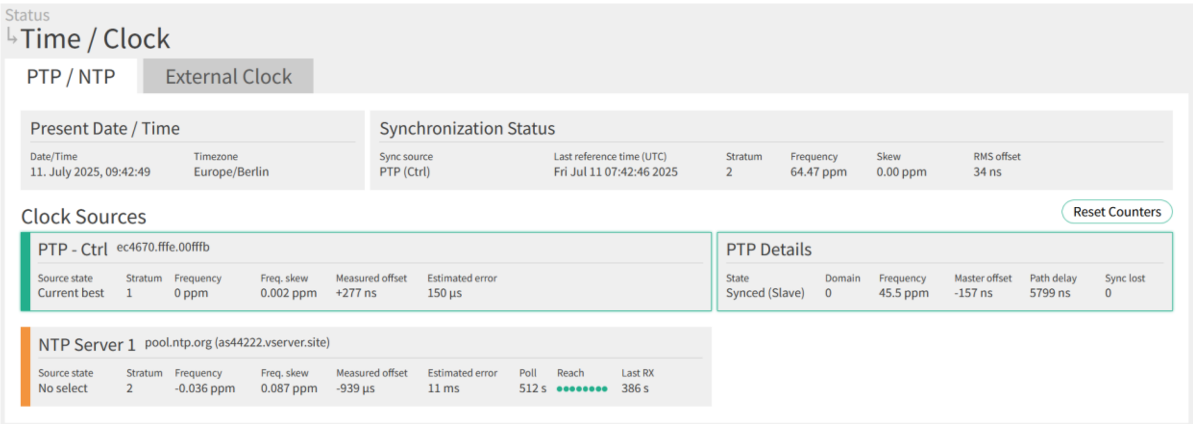


  If a backup for the external clock is configured (e. g. NTP), the switch criteria are now configured via the “Switch Criteria” tab of the “Time / Clock” menu.



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)

2.17-rc1

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

Fixed Issues

2.17-rc13

- 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)

2.17-rc12

- 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.

2.17-rc10

- 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

2.17-rc9

- 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)

2.17-rc8

- 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

2.17-rc7

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

2.17-rc6

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

2.17-rc5

- 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

2.17-rc4

- 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

2.17-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
- 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.

2.17-rc1

- 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

Version 2.16.1

13.03.2025

New Functionality

- The PTP delay mechanism to use is now configurable

Fixed Issues

- Fix a possible application hangup when updating to firmware version 2.16 or when loading settings (leading to recovery page)
- MPE decoding might lead to device crashes since firmware 2.16-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
- 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

Version 2.16

26.02.2025 – identical to 2.16-rc9

New Functionality

2.16-rc9

- DAB+ Encoder: added some status values of the FhG MuxEnc and the PAD bitrate to external APIs like SNMP

2.16-rc8

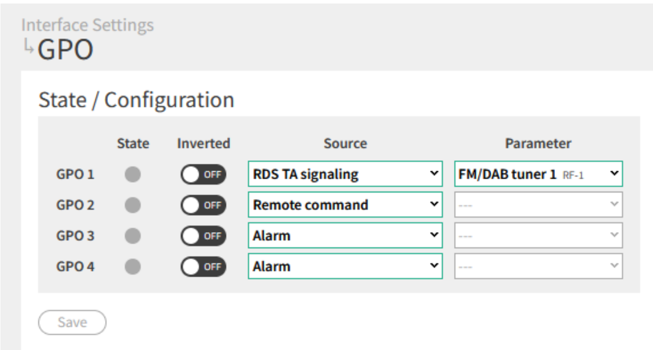
- Extended log now tracks NTP reference server changes

2.16-rc7

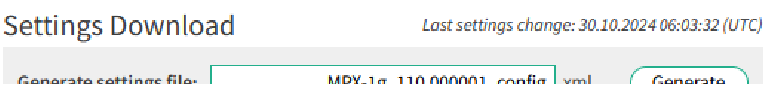
- Added new buttons to the Overview page to reset the counters of all decoders or encoders
- Optional DAB+ encoder: the FhgMuxEnc protocol for the connection to a Fraunhofer DAB Mux is now sending automatically a status for "Audio Silence Detection" together with the audio frames. The Fraunhofer DAB Mux is thereby able to determine if the DAB+ encoder feed is erroneous due to the audio silence detection and can switch internally to a backup source.

2.16-rc6

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

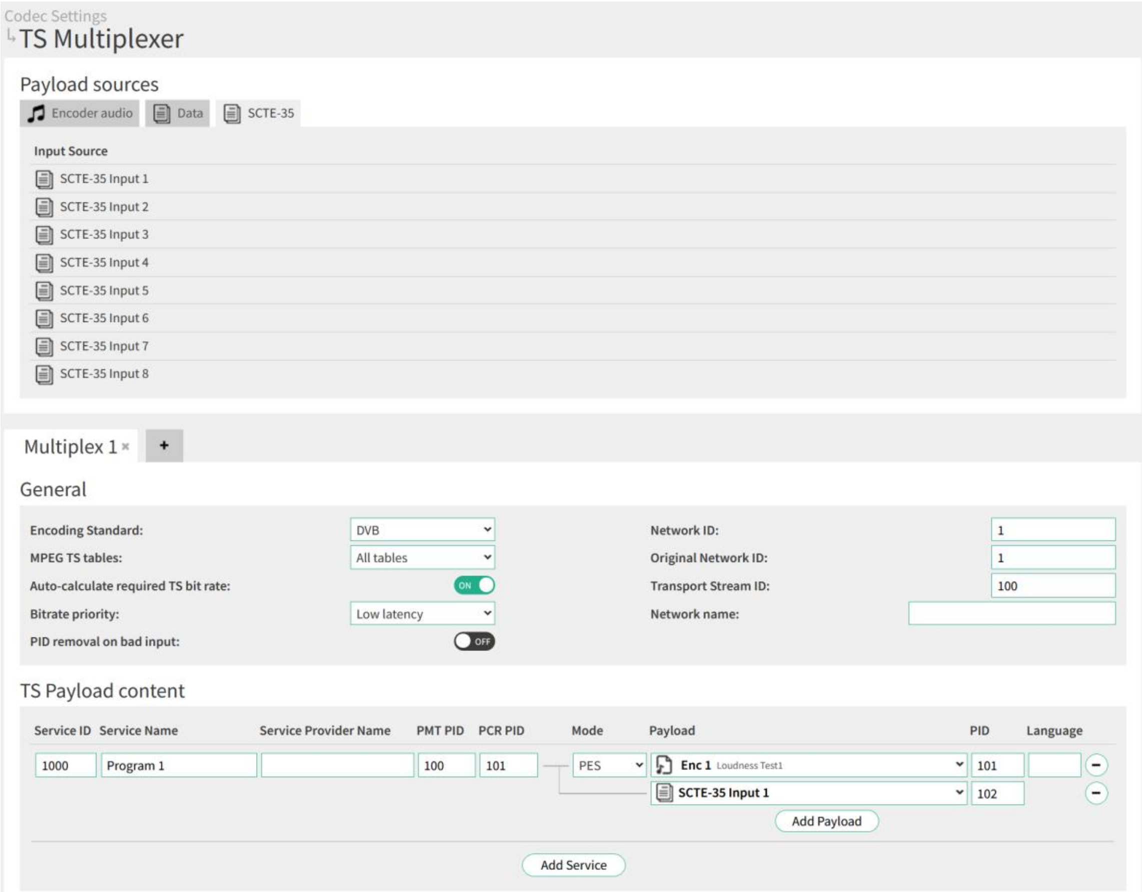


- Scheduler: Enables the execution of various actions at defined times. Currently available actions are activating/deactivating a decoder source
- Last settings change: The time can be read out as a UNIX timestamp or human readable date. This is also displayed on the Global page:



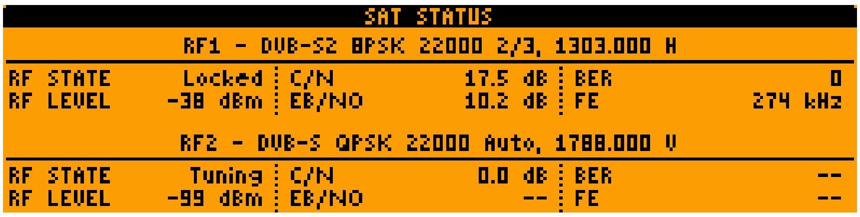
2.16-rc4

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

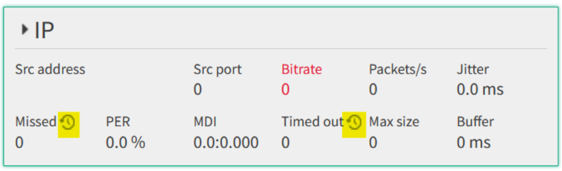


Splice inserts must be sent via one of the external APIs. Via SNMP the node is virtCslAudioEncoderScte35Spliceinsert (OID 1.3.6.1.4.1.21529.1001.2.3.18.466)
The splice information has to be passed in JSON format and should look like this:
{"cue_point": { "event_id": "1073742575", "splice_time": { "year": 2024, "month":9, "day":"6", "hours": "12", "minutes": "17", "seconds": "37", "mseconds": "210"}, "duration": "149.5"} }
(Time information in UTC, more information will follow in the manual)

- If the optional SAT tuner is installed, the LCD screen will now show additionally a status page with reception information:

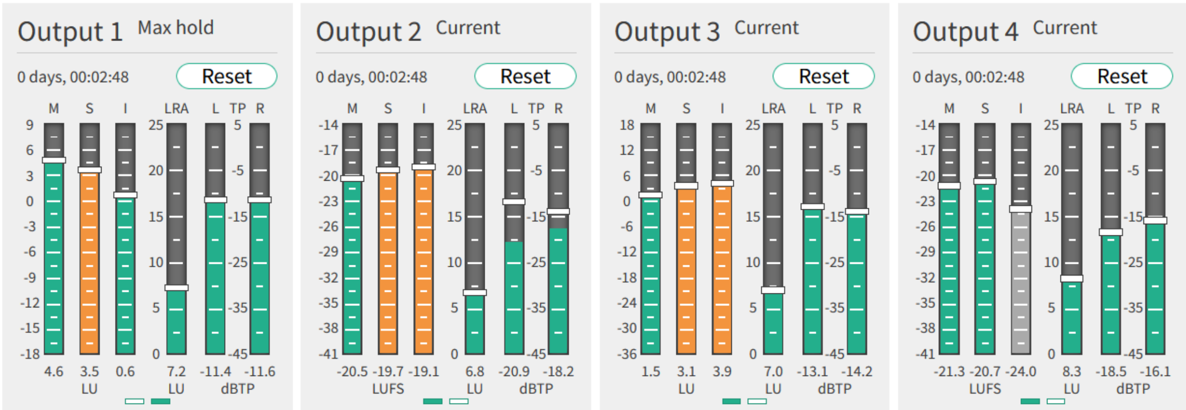


- Started to add the possibility to get a counter history on the Overview page via a small icon next to the label. It's more or less a shortcut to the extended log, showing you only the entries for the counter of this input source. Currently it's added to the "Missed" and "Timed out" counter, others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



2.16-rc2

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

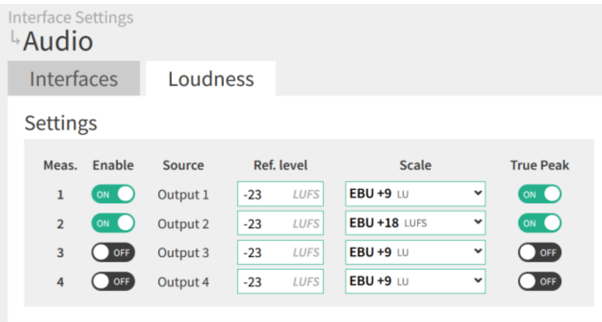


- New silence mode for elementary stream inputs: when enabled, empty packets will be inserted in case of missing stream input (like the “Generate Null samples” option for AES/EBU inputs, when no input is connected)
  This will optionally allow an encoder to generate output from an elementary stream input even if no ES Input is present
- Added the possibility to clear the extended log
- Optional HLS streaming was not working properly with some CDN servers due to cache-control

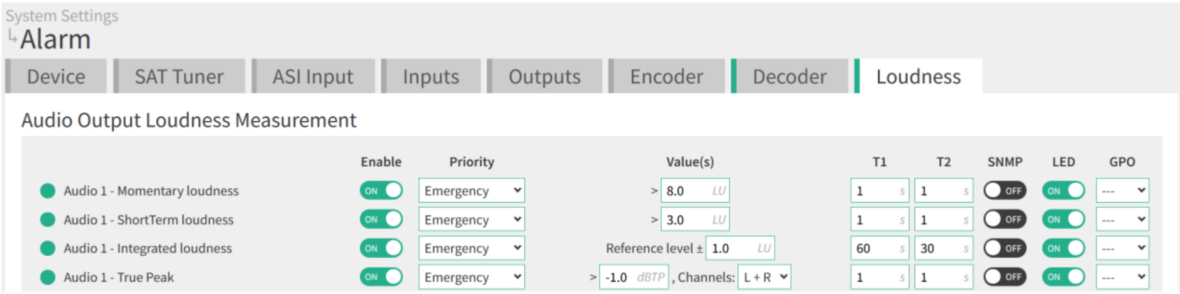
2.16-rc1

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

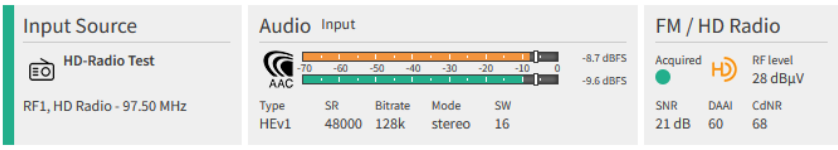
The reference levels and scales can be configured individually via Interface Settings / Audio:



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 separate status values for the TS Multiplexer outputs readable via external APIs (see e. g. SNMP MIB)
- Added uptime values for RTP/SRT input streams and RTP/SRT encoder outputs as separate values readable via external APIs
- Added support for HD Radio to the optional Multiband tuner



Additionally there are new alarms for HD Radio: HD acquired, HD DAAI and HD CdNR

Changed functionality

2.16-rc9

- 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.

2.16-rc7

- The menu of the web interface does now collapse to a hamburger menu in case screen width is limited
- The RDS Databridge configuration for the optional FM tuner is moved to a separate tab in the dialog
- TS Multiplexer: The “PID removal on bad input” functionality has two different options now. It can not only remove both the PID itself and the signaling via PAT/PMT (what was done until now), but optionally only remove the PID, but still signal it via PAT/PMT or stop the TS output completely.
- Show Dual Streaming block on Overview even if FEC is enabled

2.16-rc2

- Icecast server: the maximum number of clients was limited to five. This is increased now to 15 (but only after a “Load factory settings” is done). Alternatively, the maximum number of clients can be changed via the external APIs (the SNMP node is e. g. cslAudioEncoderOutputsIcecastTimeout, OID 1.3.6.1.4.1.21529.1001.2.3.18.20.319.4096.1.11)

2.16-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

2.16-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
- 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 2.16-rc7)
- PTP for unicast configuration improved
- Enhanced compatibility of the Icecast client in case of FLAC Icecast server content
- SIP connections for AAC improved

2.16-rc8

- When changing the silence mode configuration available for Livewire input sources, the change was not applied immediately
- For the audio output configuration, the clock source selection was not available when the signal type was set to Analog, although the clock source still affects the output speed control
- RTP input source: when the input stream changes (e. g. due to an encoder restart), RTP reception might not resume, showing an input bitrate of 0, even if the new stream is received by the device (since firmware version 2.16-rc7)
- The decoder buffer level alarm was broken in firmware 2.16-rc7

2.16-rc7

- Bunch of internal improvements and optimizations
- Security enhancements
- Headphone output may be distorted
- A livewire input source may suddenly start to provide twice as much packets as before when the “Silence mode” is activated (only true for Livewire destinations targeting the encoders)
- FLAC codec did not work in combination with SRT
- SRT in caller mode doesn’t always connect reliable with the SRT listener
- Not currently active TS/Demux backup sources may show wrong ancillary data on Overview page in case a private PID is used as ancillary source
- When changing the ancillary data configuration the ancillary output doesn't work afterwards or the device may even crash
- xHE-AAC encoder: added parameters to control live loudness and DRC
- UDP ancillary input improved (might not work after reboot when DHCP is enabled)

- When changing a TS/Demux source and switching to the Overview page the device could crash with certain transponders
- Enhanced stability
- Check minimum SRT passphrase length
- “Jumping” level meters after decoder backup switching on Overview page
- TS/SRT decoding was broken
- SIP: manual connection handling improved for incoming connections
- Live Listening: switching the input source while listening may not work reliable
- Loading settings might not activate all input source settings
- Avoid alarms of inactive channels due to reduced channel count In “Analog/Digital” device mode
- Fix the possibility to get frequent “Encoder – no input data” alarms without “good” messages In between
- Optional DAB+ encoder: Changed default value of DCP output stream data packet spreading to 50% to avoid problems with the receiving multiplexer

2.16-rc6

- Optional DAB+ encoder: DCP output stream did not work reliable with FEC activated
- Automatic codec detection for TS/Demux input sources may not signal correct AAC type
- Fixed possible application crash when loading settings or loading factory settings
- Individual status values for Livewire input sources did not work via external APIs
- TS Multiplexer: "Low bitrate priority" mode did not lead to the lowest possible audio bitrate overhead
- E-aptX decoding inside TS did not work
- Live Listening might be done with wrong audio speed (when switching between different input sources)
- Enhanced compatibility of the optional HLS client/decoder to certain streams
- Decoder: a backup in “Standby” mode isn't activated when an inferior backup is in “Active” mode
- Optional AES67 outputs had a generic name in the SAP announcements – now the audio output name configurable via Codec / Input Sources / Interfaces gets used (if set)
- Optional AES67 outputs: improved startup behavior, inhibiting a possible output burst at startup

2.16-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

2.16-rc4

- Fixed a problem on the optional Easy2Connect web page, not showing the phonebook correctly
- Optional SAT tuner: Switch criteria C/N does not work
- Elementary stream output with external clock set to NTP, activated SPN/SFN and DualStreaming with send delay ≠ 0 can lead to jumps in the AES67 output of a decoding MoIN instance, leading to errors on devices receiving the AES67 output
- Icecast client: https Icecast streams with credentials may have problems with authentication
- TS Multiplexer: configuring a new multiplex via the web interface can lead to an invalid configuration
- (Limited) support for Internet Explorer 11 was broken

2.16-rc3

- General playout stability improvements

- Clearing the extended log did not work (the latest entries reappeared after a few seconds)
- X.509 certificate error log entries moved from event log to extended log
- Fix accidentally shown number in encoder info blocks on Overview page
- In case of audio errors, the corresponding event log entry may show the wrong reason for the error
- An audio error may occur when an external clock switch happens
- Second stream in DualStreaming setup may show missed packets, although they weren't missed
- Added "Remote command" to GPO switch source select

2.16-rc2

- Elementary output streams with enabled SPN/SFN improved in case NTP synchronisation is active
- 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)
- Icecast server: Fixed long timeout (blocking one of the limited number of client slots) in case of broken TCP/IP connection
- Optional DAB+ encoder: Fixed a problem with the VLAN configuration of the DAB DCP outputs
- Optional DAB+ encoder: Fixed a problem in combination with the IZT DAB ContentServer
- Optional DAB+ encoder: Fixed a problem, when the encoder clock adaptation (called "Encoder SRC" in the Profiles settings) was activated
- In case an audio input is switched to AES67 (via Interface Settings / Audio / Signal type), the switch criteria "AES/EBU no signal" will not monitor the AES/EBU status (which makes no sense in case of AES67), but the status of the incoming AES67 stream

2.16-rc1

- The optional ASI input may not detect the TS coming from the ASI signal connected to the ASI input after a reboot of the device
- Fixed high number of DNS requests with NTP synchronization enabled
- Fixed T1/T2 alarm times not editable
- Fixed possible crash when embedding ancillary data in the encoder with certain codecs
- ProMPEG FEC encoder reports wrong FEC matrix when one FEC port offset is set to 0
- ProMPEG FEC decoder let main input stream fail, if one FEC port offset is set to 0
- Dedicated status values for TS/SRT did not work at all
- "Automatic" decoder config did not work when relying on SAP as the source for the codec detection (e. g. for AES67 inputs)

Version 2.15.5

21.06.2024

New Functionality

- Optional DAB+ encoder: added UTF-8 support for the PAD DLS charset

Fixed Issues

- Fixed a problem with the AES67 output on device startup (in case a file is used as input source)

Version 2.15.4

20.06.2024

Fixed Issues

- Fixed AES67 output clock inaccuracy

Version 2.15.3

13.06.2024

Fixed Issues

- The AES67 outputs may exhibit timestamp jumps in certain cases, e. g. when a SIP call is terminated
- The optional Dolby decoder did not work correctly and produced audio glitches
- Fixed some problems with the optional Live Listening feature

Version 2.15.2

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

Version 2.15.1

29.05.2024

Fixed Issues

- When doing “Load factory settings”, the user account passwords were not reset to default
- When loading a settings file, the user account passwords were not taken from the loaded settings file
- The optional low symbol rate SAT tuner did only work with symbol rates down to 100 kSym/s. This limit is now reduced to 64 kSym/s
- The metronome icon on the Overview page (which should signal external clock usage and its state) was not shown appropriately
- The AES67 output may provide packets that are “too early” in certain situations

Version 2.15

23.05.2024 – identical to 2.15-rc8

New Functionality

2.15-rc8

- Optional DAB+ encoder: Added DAB Submux functionality

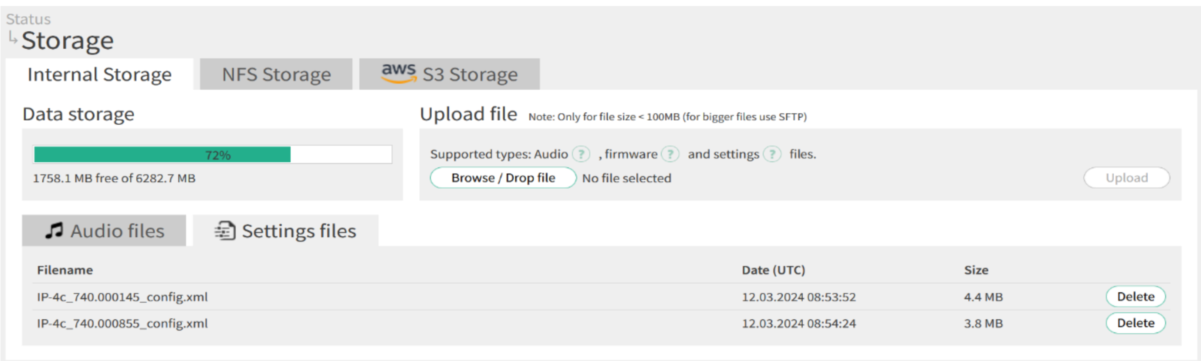
2.15-rc7

- Added parity configuration for DTE outputs
- Optional DAB+ encoder: Added the possibility to update the DLS via SFTP

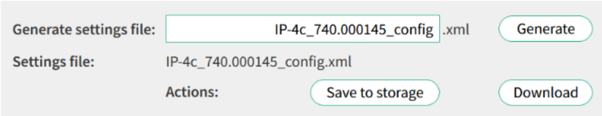
2.15-rc6

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

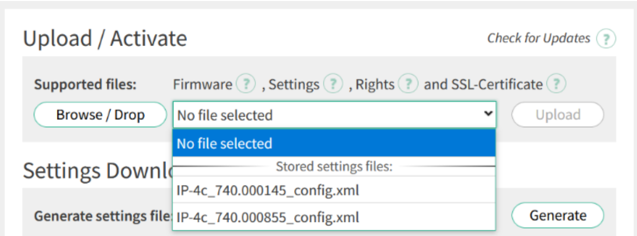
- Added event log entries when loading a settings file or loading factory settings
- Settings files can also be uploaded via the Storage page



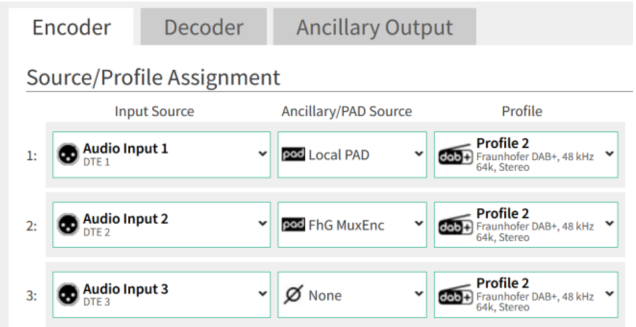
- When generating a settings file on the Global page you can now not just only download the file, but also save it to the internal storage. You can also directly change the name of the settings file.



- 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.



- Optional DAB encoder: regarding PAD insertion it's now possible to choose between local PAD and FhG MuxEnc (or none of course)



2.15-rc2

- Added RTCP switch to AES67 audio input configuration
- Added ping possibility to web interface (Network Settings / TCP/IP / Tools)
- Optional DAB encoder, DCP output: Added ETI and STI-D output mode
- Added configuration of SIP call acceptance mode per SIP input source (Automatic (all) / Manual (all) / Automatic (phonebook only) / Manual (phonebook automatic) / Reject)
  If set to Manual the actual call acceptance is only possible via the external APIs, e. g. Ember+.
- Added encoder input low level detection alarm (in addition to the silence detection alarm)
- Added an event log entry when setting a GPO by GPI tunneling

2.15-rc1

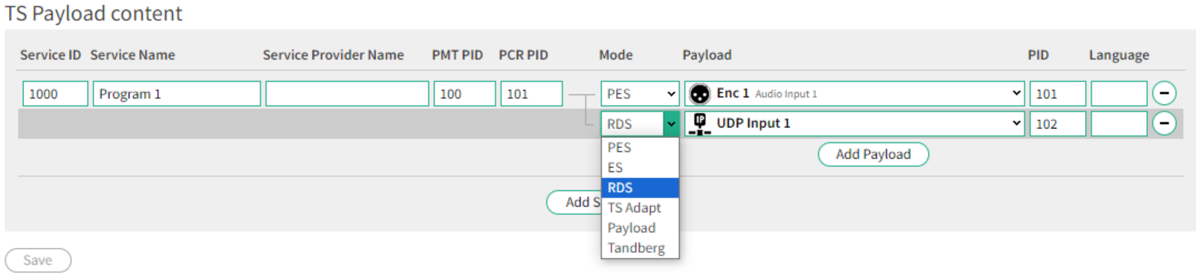
- Added entries to the extended log in case of CC (continuity count) errors in transport stream decoding

Changed Functionality

2.15-rc7

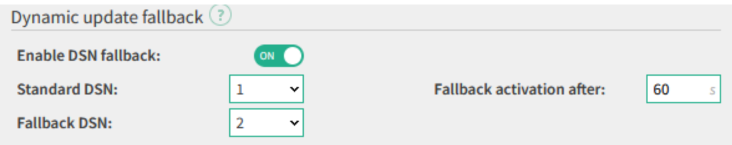
- The PTP configuration did only allow to configure a single IP interface, providing no backup functionality in case PTP fails on that interface. Now all interfaces configured for streaming data (under Network Settings / Services) will be used for PTP, automatically switching between the interfaces when needed
- New version of the xHE-AAC encoder (04.05.04) with default loudness -24 LUFS

- Updated Fraunhofer libraries to latest versions
- TS Multiplexer: Rework of the private data encapsulation mode configuration. Instead of configuring a global mode the mode can now be configured individually per payload



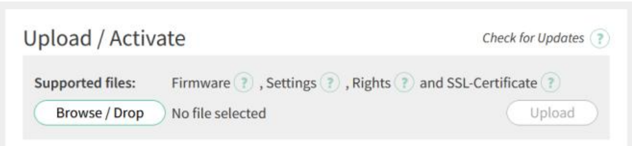
As part of this change we also added three more private data encapsulation modes: TS Adaptation, Payload (behind TS Adaptation) and a special Tandberg compatibility mode.

- RDS Databridge (optional): If no RDS update is received for a period of time, the Databridge can switch the RDS encoder to a "Fallback DSN".



2.15-rc6

- 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

2.15-rc2

- 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
- Allow lower SAT transponder frequencies for C band transponders
- Changed Load Factory Settings via LCD menu to new method without reboot

2.15-rc1

- 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

2.15-rc8

- AES67 output did not work, when no External Clock was configured (since firmware V2.15-rc7)
- 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

2.15-rc7

- 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
- Icecast server: enhanced compatibility to some clients (expecting a “bitrate” information instead of the “ice-bitrate” information)
- RDS Databridge (optional): the EON TA functionality does now respect the EON PI configuration and will not react on EON Ta from a service with a different PI then the configured one
- Improve robustness of the TS decoder (could hang up completely on bad input data, stopping TS processing completely)
- TS private data decoding may stop after bad weather and disturbed SAT reception
- file playback can be disturbed (since firmware 2.15-rc1) -> the minimum configurable audio buffer has been increased to 100ms

2.15-rc6

- 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
- Reverted IGMP rejoin functionality introduced in firmware 2.15-rc2, as it did more harm then being useful in the way it was done
- Optional HLS encoding: fix wrong HLS container formats
- General security and access control improvements
- Again: Improved PTP synchronization of AES67 output, improving AES67 output stability
- PTP did only work with domain 0
- Fixed some problems with HE-AACv2 and the Icecast Source Client and Icecast Server
- Enhanced compatibility to AAC encoded in short frame mode

2.15-rc2

- 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
- Optional DAB encoder, DCP output: optimization of PAD interface

- Changing just the audio buffer level of an input source is not (always) applied
- Fix the possibility that the TS decoder does not provide the service list (showing 0 services)
- After settings updates (loading a settings file or loading factory settings) input source changes may not be applied
- TS/Demux: the audio buffer is automatically adjusted to the PES frame length if the configured value is too low to avoid audio buffer underruns
- DTE output configuration with Private Data MPE demux can disturb other DTE outputs

2.15-rc1

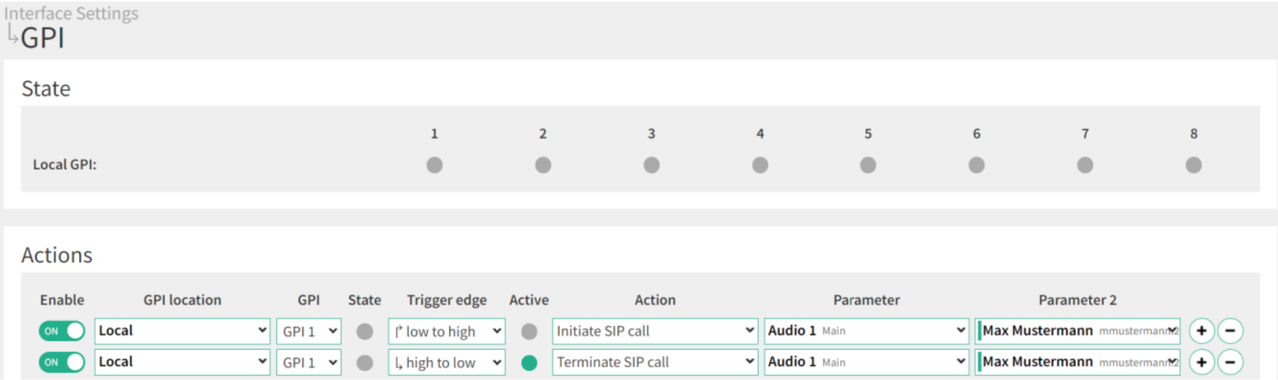
- Under high CPU load IP packets may get lost and reported as missed
- Under high CPU load the audio output may get distorted (event log showing FPGA underruns)
- When doing “Load Factory Settings” and unplugging the power immediately after completion, the new (default) settings might not have been saved
- Optional SAT tuner: For C band transponders the minimum allowed transponder frequency was too high, not allowing to configure some transponder frequencies
- AAC decoding may encounter problems in case of SIP connections to some other vendors (problem with short frames)
- Enhanced SIP connection compatibility via mobile connections (e. g. LTE)
- The AES67 output may still not be synchronous to PTP
- High CPU load in case of incomplete TCP connections (Icecast/HLS)
- Optional HLS decoder: Improved compatibility
- Web interface: the page heading may show an incorrect title when some menu entries are disabled

Version 2.14

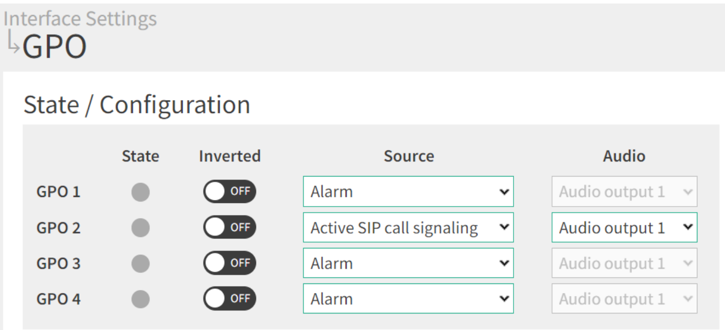
15.01.2024

New Functionality

- Added the possibility to configure actions triggered by GPI.
  Currently the only available actions are “Initiate SIP call” and “Terminate SIP call”.



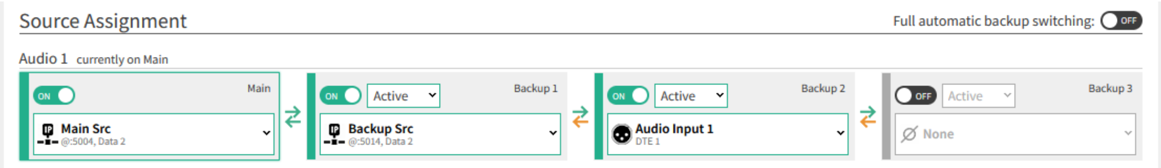
- Added the possibility to trigger a GPO to signal an active SIP call



- Added silence detection alarm for encoder inputs
- Added missing VLAN support to SRT input/output configurations
- For security reasons the password hashes of the user accounts are no longer stored as MD5, but as bcrypt hashes. The migration will be done automatically.
- Added the 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)
- 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.
- 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
- 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 an (optional) HLS/MPEG DASH client for the decoders (no transcoding)
- Added a possibility to configure a delay for the GPI Forwarding/Tunneling (per profile)
- Added AES/EBU CRC Error counter to Overview page
- Added the possibility to 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

- 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
- 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

- Icecast streams with FLAC codec did not work
- AES67 inputs/outputs allow invalid 32bit sample width
- AES67 input/output is always shown with 24bit sample width instead of the configured one
- 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 potential high amount of audio errors due to FPGA audio buffer underruns
- Fixed visual problems with the TS Demux service select update/refresh
- SAT tuner alarms were swapped between RF1 and RF2
- Optional DAB+ encoder - DAB DCP output: the second control channel was not working
- Optional DAB+ encoder - DAB DCP outputs: multiple control connections to >= 2 mux can cause control data loss
- Optional DAB+ encoder: muxenc login was not considered for DCP connections leading to a change of settings without checking the login password
- 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
- DNS configuration per interface did not work via LCD menu
- AES/EBU audio output improved for 192kHz input streams
- TS Encoder: audio decoding may drift due to a problem with the PTS timestamps
- TS Encoder: fix a possible mismatch of TS stream type and audio descriptors in case of AAC
- Device could hang up when the audio file belonging to a file input source is not present (e. g. after loading a settings file)
- Fixed a high CPU load problem when many AES67 receive streams are configured
- MPEG TS decoder: enhanced audio decoding compatibility of RTP streams inside MPE
- Synchronized Playout did not work correctly without SFN license
- Ember+ improvement for read-only items (in the virt subfolder)
- Some special characters in the name fields of the input sources etc. could mess up the display of the defined settings in the web interface
- SAT tuner status may get unavailable (e. g. after disabling/enabling a SAT input source)
- Extended log entries for missed RTP packets may have empty stream names, if the packet was lost in a redundant (DualStreaming) or FEC stream
- Fix possible crashes with AES67 inputs (due to new extended logging)
- External clock configuration - a reconfiguration from an already configured and valid clock source (e.g. from PTP to PPS) was not handled correctly
- SNMP: virtCslAudioDecoderStatusAudioDecoderstate (1.3.6.1.4.1.21529.1001.35.2.3.4.37.3.4096.1.7) did not always provide the correct state
- Optional decoder ancillary data is now also shown on the Overview page if the input source is not the currently active one
- The Icecast Server encoder output did not work with HE-AAC(v2)
- With the optional low symbol rate SAT Tuner the device still could fail to re-establish tuner lock and the audio outputs stay silent if reception gets interrupted due to e. g. bad weather conditions
- SNMP monitoring status was not reset to “disabled” if all input sources are getting disabled
- SNMP monitoring status / Warning LED / Relay were not reset (e. g. for the “Audio – No Input Data” alarm) if all input sources are getting disabled
- The optional SAT Tuner alarms are no longer triggered if the tuner is inactive (unused)
- Alarm "No Input Data" did not check if PES/MPE data is present in case of TS/Demux input source

- Fixed "good" check for ES input source switch criteria "No input data" (T1 time was not considered correctly)
- Fixed "good" check for file input sources (were always bad since latest firmware when “No Input Data” switch criteria was activated)
- Fix AAC-ELD(v2) ES decoding not working when Decoder type is explicitly set accordingly
- File playback from playlist was broken since latest firmware 2.13(.1)
- Fix local audio level display for an active SIP connection
- 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
- Cyclic EON-PS and EON-PTY did not work with optional FM Tuner and RDS Databridge
- Optional TS MPE Encoder: enhanced compatibility to certain 3rd party decoders

Known bugs

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

Version 2.13.2

02.10.2023

Fixed Issues

- Fixed a problem with the second audio output when the sample rate converter is activated and a digital reference input is used (the audio buffer will not be stable and drifting slowly, either running empty or increasing)

Version 2.13.1

10.08.2023

Fixed Issues

- Fixed some web interface annoyances, where it was not remembering the last active tab, especially on the Codec page, and showing visual glitches when adding multiplexes on the TS Multiplexer page

Version 2.13

08.08.2023

New Functionality

- Added compatibility to Qbit GPIO Forwarding in ancillary data
- Added an option to disable the RTCP output in parallel to an RTP output stream
- Added a configurable silence mode to Livewire input sources for the encoders, inserting empty packets in case of missing Livewire stream input, thereby having continuous encoder output
- Added a new status value providing the delta time between the two streams in RTP DualStreaming setup
- Added monitoring/alarms for the optional FM/DAB tuner
- New (optional) DAB+ encoding functionality
- 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, RTP Rx timeout, SIP register/connect/disconnect/ declined/error and SRT connect/disconnect.

Like the counters this extended log is volatile, meaning it is cleared after a reboot. If syslog is enabled, these events will however also be sent out to syslog, making them externally persistent.

- Added optional support for NFS storage (for audio files; in addition to the internal storage)
- Added optional support for AWS S3 storage (for HLS Push encoding)
- Added interface (and VLAN) selection to the SNMP trap manager configuration
- HLS encoder: HLS container format is now configurable
- Added alarm/monitoring for the optional ASI input

Changed functionality

- The Icecast Server did still answer with “ICY 200 OK” to a connection request and only for certain user agents / browsers with “HTTP/1.0 200 OK”. For some time now the ICY answer is however deprecated and should no longer be used, so we do now always answer to a connection request with the HTTP answer, thereby hopefully improving the general compatibility of the Icecast server to certain clients.
- Reported Livewire GPI/GPO states are now high in default state and low in triggered state
- SRC/DST name size increased for Livewire Routing Protocol (LWRP) from 15 to 25 characters

Fixed Issues

- Invert setting in GPO configuration had no influence on reported GPO state via Livewire
- RTP dual streaming improved
- Livewire SRC and DST mapping/numbering was wrong when less than the max. number of channels is licensed, delivering no or wrong audio data
- HLS encoder: small bugfix for FLAC and ISOBMFF
- Sample frequency detection for AES/EBU audio input channels higher than 1 improved in case the sample rate converter at the audio input is switched off (resulting in audio distortions)
- RTP receiver information in case of dual streaming improved
- Fix status problems of optional DAB tuner (RF level etc. are reported as faulty sometimes, although this is not true)
- SAT Tuner: Fixed BER calculation for low symbol rate single tuner
- Improved stability (jitter) of AES67 output
- Fix possible crash when changing the sample rate of an audio input
- RIST receiver improved for low latency transmissions, avoiding sometimes quite aggressive retransmission requests
- RIST receiver improved for dual streams with encoder send delay
- Switching VLAN activation without modification doesn't work
- Control services are not correctly setup in VLAN environment
- When a sampling rate for an audio input in analog mode is set to a value different than 48kHz, the decoded audio is distorted
- File input source from NFS source does not perform auto-reconnect on XML settings import
- Digital interface output level can be displayed greater than 0 dBFS when a gain value > 0 is configured and the audio level is high
- Fixed shown name of radio input source during Drag&Drop

- Drag&Drop of TS Data Demux input sources did not always work
  Changing an active gateway address for a VLAN requires a reboot to become valid
- VLAN modifications could influence the main interface, too
- SAT tuner: if reception gets interrupted due to e. g. bad weather conditions, the device still could fail to re-establish TS decoding and the audio outputs stay silent
- NFS handling improved
- HLS server push improved for S3 storage
- HLS encoder: Fixed issues with xHE audio on stream start with Safari and iTunes
- Fixed Livewire routing protocol LWRP not working after changing the IP address of the device
- MPE decoder: MPE ancillary data output was not working
- RIST encoder retransmission improvements in case of SFN
- Fixed possible timestamp display problems in new Extended Log
- VLAN output streams will be now refreshed after config changes to VLAN parameters
- Decoder input sources with backup policy "Active when needed" might not get deactivated again when activating superior source
- MPEG TS Decoder: optional decoding of 192kHz PCM in PES mode without external clock did not work
- Ancillary data output for SRT input sources may not work when having configured "Audio Output X" as input source for the Ancillary Output
- 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)
- Encoder did not react on SRC on/off in audio input config (potentially changing input sample rate)
- RIST receiver improved for RTP streams, where the encoder has RIST disabled (could lead to audio buffer not building up)
- RIST didn’t work in dual streaming setup for the redundant line
- RTP: Overall packet lost counter could be too large in rare cases
- HLS encoder: fixed a problem with some HLS clients reporting faulty HLS segments (especially with the AAC codecs)
- Fixed compatibility issues when MM01 is the audio encoder and RTP packet fragmentation is activated via “RTP max payload”
- MPEG/TS decoder: enabled the possibility to decode audio streams not announced via PAT/PMT automatically without the need to set the codec type manually
- MPEG TS decoder: fixed compatibility of private data TS decoding if the PID is not announced via PAT/PMT
- MPEG/TS decoder: enhanced audio decoding compatibility of RTP streams inside MPE
- Added support to transcode TS/Demux input sources with private data in Pipe mode
- Changing just the LNB config for a SAT input source was not applied
- Optional Live Listening feature did not work with Safari browser
- Fixed ancillary data descriptor in MPEG TS encoding for better compatibility
- A configured source IP for SSM (Source Specific Multicast) could not get deleted

- When switching between different web interface menu items the page will now always scroll back to top
- DTE baudrate changes were not applied immediately, but only after a reboot – Fixed
- Ancillary data was not shown on the Overview page for XLR audio input sources

Known bugs

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

Version 2.12.1

28.02.2023

New Functionality

- 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".

Changed Functionality

- None

Fixed Issues

- None

Known bugs

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

Version 2.12

31.01.2023

New Functionality

- Added gain configuration to all input sources to e. g. allow alignment of main and backup sources
- Added DAB+ support to optional (formerly FM only) multiband tuner (VER63020)
- 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 inputs, providing its physical XLR audio inputs as Livewire audio streams and optional Livewire Sources for the audio outputs (instead of AES67 streams), allowing to have a Livewire audio stream instead of physical XLR as audio output.
  An IP-4c with 4 channels will announce 8 Livewire Sources, where SRC1 – SRC4 will reflect the audio inputs and SRC5 – SRC8 will reflect the audio outputs
- The Livewire input sources for the Encoder section will now be a fixed number of Livewire Destinations (as much as encoders are available), thereby allowing the configuration via LWRP (corresponding to the fixed number of Livewire Destinations for the Audio Decoder section).
  An IP-4c with 4 channels will now announce 24 Livewire Destinations, where DST1 – DST16 will reflect the possible Livewire input source for Audio1/Main, Audio1/Backup1 and so on and DST17 – DST24 will reflect the possible Livewire input sources for Encoder 1 to 8.
- GPIO Tunneling has now the option to tunnel the Livewire GPO state instead of the physical GPI state.
- Added Livewire level meters for Sources and Destinations (enhance compatibility to e. g. Pathfinder)
- Added support for the Livewire GPIO snake mode, allowing to link the GPOs of the IP-4c to the GPIs of a different Livewire device.
- Added two virtual Livewire GPI ports (GPI 3 and 4), which will reflect the state of GPO port 1 and 2. This will allow other Livewire devices to register for GPI changes via snake mode and thereby follow a GPO change on the IP-4c (e. g. via GPIO Tunneling), reflected as a virtual GPI change. The virtual GPI ports will thereby provide a GPO pass-through to other Livewire devices.
- Added optional RDS Databridge as ancillary data provider for a FM radio tuner input source (having the same functionality as in our FM02 or A30)
- Added status tabs to Overview page for optional AES67 outputs and Livewire Sources
- Added PIN lock option for LCD menu
- HLS encoder: added support for FLAC and Opus
- 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 and Radio)
- 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 GPIO Tunneling info not shown for encoder if no ancillary data source is selected
- Fix xHE-AAC problems with low bitrates
- Icecast input source handling improved for faulty meta data from some Icecast servers
- 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
- SIP: calls could not be cancelled during connection establishment
- SIP: redial improved
- SIP/SDP: (EBUACIP:VERSION) information moved, enhance compatibility to Unify OpenScape SBC
- NMOS: improved compatibility with STAGED mode (SDP parameters)
- AES67 output compatible with DHD (silent mode)
- AES67 outputs loose the PTP synchronization after reconfiguration the AES67 output parameters
- AES67 output: improve stability of empty packets mode
- AES67 input: fix problem with mono input streams
- Fixed MPEG TS signaling for MPEG2-AAC
- Improved transcoding of MPEG TS with ancillary data
- Fixed an issue with the ancillary output not working and providing no data when a TS ancillary data or a private data input source is configured and only assigned to one of the ancillary outputs without using the same TS source in one of the decoders (or maybe encoders)

- 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/Demux config: Fix empty audio track select on config mode change from “Service (auto)” to “Service (fixed)”
- SAP: fixed crash with too many announced streams
- HLS/Icecast Encoder: Fixed a possible deadlock when TLS/SSL encryption is enabled

Known bugs

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

Version 2.11

12.09.2022

New Functionality

- Added VLAN support to the new AES67 inputs/outputs, which can optionally replace the physical XLR inputs/outputs, e.g. for SIP
- Added VLAN support to the Livewire routing, advertisement and GPIO configuration
- The AES/EBU inputs delivered no audio, when nothing is connected and the sample rate converter is switched off (or used by the output). Now there's a new configuration option "Generate Null Samples", which will deliver audio silence, even if nothing is connected at the AES/EBU input.
- SIP: Added support for DualStreaming (SMPTE 2022-7; EBU TECH 3368, Chapter 3.7.4.1)
- Added support for Multichannel AES67/Ravenna input streams – it's possible to select the channel(s) to use from it
- Added support for user authentication to Icecast client
- Added the additional possibility to configure a DNS server per IP interface (including VLAN)
- Added support for FEC according to RFC 2733 (e. g. used by Digigram IQOYA)
- Added the possibility to upload MP4 and M4A files from within the web interface via Status / Storage
- Added "copyToClipboard" icon to IP status on the Overview page to copy the IP address
- In case the physical inputs/outputs are configured as Digital AES/EBU, the sample rate is shown in the level meter blocks at the top of the Overview page

Changed functionality

- SAP: the device is now listening to SAP announcements on all interfaces
- SAP announcements are now done on the interfaces, where the streams are sent (including VLANs)
- Ember+ can now be used simultaneously via more than one interface including VLAN(s). The configuration is done via Network Settings / Services. The switch to disable Ember+ completely is removed – instead disable Ember+ for all interfaces via Network Settings / Services.
- The AES67 outputs, which can optionally replace the physical XLR outputs, were announced via SAP/SDP with the fixed name "AES67x", where "x" was the output number. If a name is configured for the audio output (via Codec / Input Sources / Interfaces / Audio Output), that name will now be used for SAP/SDP announcement.

Fixed Issues

- SAT tuner: if reception gets interrupted due to e. g. bad weather conditions, the device might fail to re-establish TS decoding and the audio outputs stay silent
- MPEG TS Encoder: Better PTS generation, thereby improving compatibility to certain decoder devices (e. g. Ateme DR5000 or SA D9846)
- Ancillary data decoding didn't work with Opus codec
- A snmpwalk could crash the device when the EBU Tech 3326 license is not present
- Switching off all TS/IP multiplexer outputs did switch off a possible active ASI output, too

- Audio outputs 3 and 4 did not work when the device started in reduced channel count mode and is switched to full channel count later on
- Added the missing seconds information to the UECP CT ancillary data status
- Audio silence detection may produce log entries named AES67
- NTP based synchronized playout was broken in version 2.10
- SAP: Enhanced compatibility to Dante Controller
- HE-AAC and MPEG-4 Audio Descriptor: the encoder will now signal level 2 if the sample rate is <= 24 kHz and level 3 above (according to ISO/IEC 14496-3:2019, Table 1.11)
- HLS encoder: decoder switched to mono mode with xHE HLS stream
- HLS encoder: AAC-LC (e.g. 192kbit/s) was not working with Safari as decoder
- Enhanced compatibility of the Icecast decoder to TCP-only streams (without Icecast information)
- File playback startup of huge CBR MP3 files may take several seconds
- Changing e. g. a file input source used by an encoder (changing the configured file) lets the encoder stop encoding
- With the optional FM/Multiband tuner it was possible to enter frequencies with 10 kHz accuracy. This accuracy was however not shown in the Overview and was rounded up to 100 kHz accuracy once leaving the edit dialog and re-entering it.
- Web interface font rendering on non-Windows devices was suboptimal, showing some visual glitches – now it should be more or less equally on all devices

Known bugs

- n/a

Version 2.10.1

07.06.2022

Fixed Issues

- MPEG TS decoder: further enhancements to private data decoding
- Ember+: added support for "subscribe" mode
- SNMP: SRT connectionState could return undefined value
- PHP error messages on TS/IP and TS/SRT input source tabs in case TS decoder license is missing
- Link to ASI configuration page was broken on TS Multiplexer page / Multiplexer Outputs / ASI Output

Version 2.10

01.06.2022

New Functionality

- Completely revised ancillary data handling for full flexibility
  The fixed linkage between ancillary inputs/outputs and corresponding audio inputs/outputs is entirely removed. Instead a variable number of UDP ancillary inputs/outputs can be configured besides the available DTE inputs/outputs.
  When configuring the Encoders on the Codec page, each configured audio input source can be accompanied by one of the ancillary input sources to be encoded together using the given profile. In that way the ancillary input sources are no longer limited to the physical XLR audio input sources, but can now be used together with any audio input source, e. g. Livewire, AES67 or Icecast. In addition there's a special "Pipe" ancillary input selection for the Encoder. Using the Pipe mode the ancillary data of the audio input source (including possible GPI Forwarding) will be preserved and transcoded together with the audio.
  Moreover there's a new "Ancillary Output" tab on the Codec page allowing to configure freely the input sources to use for the available ancillary outputs (DTE and UDP). When using one of the "Audio Output" input sources (that's the default), the ancillary data coming from the input sources configured on the Decoder tab including the backup switching will get used.
- Ancillary data and GPI Forwarding state available on Overview page
  If ancillary data or GPI Forwarding information is contained in an input source, the Audio info blocks on the Overview page will allow to inspect that data. This is especially useful in a transcoding setup, where the ancillary data would otherwise not be available for examination.
- TS private data output
  It's now possible to configure private data only TS Demux input sources (without the need to configure an audio PID). These "data only" sources can be used on the new "Ancillary Output" tab.
- Optional audio PID removal from TS Multiplexer in case of audio input loss (to e. g. trigger external backup)
- Added NMOS support (Networked Media Open Specifications, https://specs.amwa.tv/nmos/)
  The NMOS support can be enabled on the new "External APIs" page, which summarizes the configuration of the external APIs SNMP, Ember+ and NMOS.

- AES67 inputs/outputs can now optionally replace the physical XLR inputs/outputs, e.g. for SIP
- The optional AES67 input stream can also be a Multichannel input stream – it's possible to select the channel(s) to use from it
- Added audio output buffer level alarm (to get e. g. a fast relay alarm before the audio buffer runs empty and trigger thereby an external backup)
- Added audio error count alarm (to get an event log entry in case the audio error counter increases)
- Added new ancillary input and output alarms for the dedicated inputs and outputs
- Added ancillary data decoding support to Icecast input sources
- Added the possibility to bind the optional syslog output to a certain interface
- Added optional support for ASI input (available with the satellite tuner or standalone)
- Enhanced optional MPE encoding capability of TS Multiplexer to support RTP, too
- Added xHE-AAC support to optional HLS encoder
- In case external clock synchronization is active, the transcoding elementary stream output of an unsynchronized RTP stream used as encoder input source can now be synchronized to the external clock (activate the "Synchronous Playout" switch in the RTP elementary stream output settings)

Changed functionality

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

Fixed Issues

- A memory leak caused a reboot of the device after a few hours up to a few days in case PTP clock synchronization was enabled
- SAT input sources didn't work as encoder input only (they only worked, if they were additionally used for the audio outputs)
- In SAT transcoding setup, the Overview page didn't show the original audio input status, but only the output (transcoding) status
- TS/Demux: fetching the service list could sometimes fail to get all service names
- When starting up in reduced channel count mode (to enable Analog input/output usage in case more than 2 channels are licensed), switching back to "Digital only" did not directly work and a reboot was required
- Livewire source select was wrong in Livewire input source configuration dialog
- Livewire was always able to set GPO without respecting switch source setting
- Changing the manual decoder config of a TS/Demux source did not trigger a reconfiguration
- Fixed a crash when changing the TS/IP input source used by a Demux source in MPE mode
- Audio Output as TS input source did not show its levels in the payload info block on the TS Multiplexer overview page
- When only the port of an elementary stream output was changed, the SAP/SDP announcement wasn't updated
- MPEG TS encoder: improved compatibility to certain decoders (PTS offset reduced)

- MPEG TS decoder: private data encapsulated as ES (without PES) wasn’t decoded
- TS/SAT configuration: wrong display of SAT Low/High band in case of transponder frequency input
- display problem with umlauts etc. in service name of TS/Demux sources in config list
- SAP information wasn’t updated on external clock configuration change
- SIP: Support of mobile VoIP devices which don't handle SIP correctly
- SIP: status information for SNMP improved and enhanced
- SIP: Invalid SIP number shown in log if SIP call failed
- SIP: RE-INVITES improved (https://datatracker.ietf.org/doc/html/rfc3581#section-3 is now implemented correctly)
- ASI output was disturbed when only an encoder profile was changed
- The optional SIRC Data Channel was included in the TS only every 2nd time on TS Multiplexer changes

Known bugs

- The link to the ASI output configuration on the TS Multiplexer page is broken. The new menu entry is Interface Settings→AUX.
- Codec page: some PHP error messages will get shown on the TS/IP and TS/SRT input source tabs, when the TS_Decoder license is missing

Version 2.08

08.02.2022

New Functionality

- Added syslog support (event log entries can be forwarded to up to three syslog servers)
- Added VLAN support to Icecast source client and Icecast server encoder output
- Added HTTPS support for Icecast input via proxy server
- Added optional MPE encoding capability to TS Multiplexer
- Added optional ADTS (MPEG-2) TS transport format support for HE-AAC and HE-AACv2

Changed functionality

- Increased possible maximum global delay to 10 seconds (only possible with NTP based synchronized playout)

Fixed Issues

- Fix problems with Icecast input via proxy server
- SPN/SFN/Stream4Sure did not work with AES67 as audio input, but only with the XLR hardware audio inputs
- Elementary Stream output in DualStreaming setup caused collapsed packet rate, when one of the interfaces was DOWN or several outputs did configure RIST or send delay
- SIP: re-registration sometimes failed if SIP server response with internal errors
- SIP: possible crash on TCP connection attempts via e. g. port scans
- SIP: Calls failed sometimes belonging to invalid DNS server requests and SIP registrar timeout handling
- SIP: calls improved (INVITE response message 200 from remote agent were not always answered with an ACK)
- SNMP/Ember+ status information for SIP and TS input sources improved
- SNMP: sysObjectID (.1.3.6.1.2.1.1.2) was always answered with .1.3.6.1.4.1.21529 (twowcom), no matter which device it is. It does now indicate the device in use.
- Audio output input sources were shown with a wrong name in the encoder list on the TS Multiplexer page
- Output gain adjustments were not taken into account for the output level values shown on the Overview page
- Added possibility to download the user manual from within the web interface (if the device has access to the internet)
- Improved “Check for updates” functionality from within the web interface

Known bugs

- n/a

Version 2.07.2

03.01.2021

New Functionality

- Added separate SIP status info block on Overview page in addition to normal IP info block for SIP input sources

Fixed Issues

- PTP alarm did not work
- Left channel level on Overview page info blocks might show empty level instead of -240.0 dBFS
- Easy2Connect: Don't show tabs for assigned, but disabled SIP input sources

Version 2.07.1

28.12.2021

Fixed Issues

- NTP was not available as external clock source if neither the Ravenna nor the SFN license were present
  This was of course not the intended behaviour, as NTP based synchronized playout is now a base feature of the IP-4c.
- PTP was not available as external clock source if the Livewire license is present, but the Ravenna license is missing

Version 2.07

22.12.2021

New Functionality

- New TS Demux configuration option via service selection
  Instead of having to enter the audio PID manually, you can now switch to the new configuration mode “Service from list (fixed PID)”. It will show a list of all available unscrambled audio services in the chosen TS source and a sub selection of the audio PIDs belonging to that service (some services carry more than one audio PID). If the service list is not yet available (the TS source is not yet configured for any decoder or encoder) there’s a refresh button next to the service list, which will temporarily configure the chosen TS source (tune in case of SAT source) and collects the service information.
  This new additional configuration option via service list is also added to the Private Data decoding.
  Limitation: To not interrupt an already configured TS decoding, it will not be possible to get the service list of a SAT source, which is using the same RF input as the already configured and used TS source. So if you want to get the service lists of several SAT transponders (SAT/TS sources) you should fetch the service lists prior to configure one of that TS/Demux sources for one of the Decoders or Encoders.

- New TS Demux configuration option “Service from list (auto PID)”
  In contrast to the above explained new configuration mode “Service from list (fixed PID)” the “auto PID” mode will also show a list of unscrambled audio services to select from by service name, but will not allow to configure/select the audio PID belonging to that service. The device will automatically choose the first available audio PID belonging to that service and will automatically follow any audio PID change that is announced via the PMT. That’s useful for services which change the audio PID for a certain time slot during the day, e. g. for a slot with a different language.
- New TS Demux private data configuration option via service selection
  Corresponding to the new audio configuration via service selection it is now also possible to configure the private data PID via a selection from a list of available services carrying private data.
- NTP based synchronized playout
  A synchronized playout can now be achieved via NTP as a base feature. “Synchronized playout” means, that every decoder decoding the same encoder signal will playout the audio at the same time, no matter of different network delays. Formerly it was only possible based on 1PPS or PTP and linked to the optional SFN license/right. Please note that the accuracy, that can be achieved via NTP, is of course inferior to the one based on 1PPS or PTP and will be in the range of 10 to 20 milliseconds, depending on the accuracy of the NTP server, whereas the accuracy via 1PPS or PTP is in the range of some microseconds.
- Backup configuration for the external clock
  It’s now possible to configure two backup external clock sources besides the main external clock source together with corresponding switch criterias. NTP is added as a possible source (besides 1PPS and PTP).
- Support for bit transparent transport of 32bit AES3 signals according to Ravenna AM824 / SMPTE ST 2110-31
- Added Ancillary Data timeout alarm (for both input and output)
- Added AES3 CRC error alarm
- Added TLS/SSL support for optional HLS encoding
- Added TLS/SSL support for Icecast client (allowing to receive streams via HTTPS)
- Added TLS/SSL support to Icecast server encoder output
- Added support for burst on connect to Icecast server encoder output
- Added proxy server configuration for Icecast input streams (on TCP/IP page)
- Added VLAN support to PTP interface configuration
- Added stream/connection uptime information to Elementary Stream and SIP info block on Overview page

Changed functionality

- Increased the number of possible presets per input source type from 16 to 64
- For TS/Demux sources the name shown in the Source Assignment section and on the Overview is no longer the name of the TS source, but either the service name or the TS/Demux name (if in Manual mode)
- Rework of the TS service list on the Overview page
  It will now show a lock symbol for scrambled services and we added symbols to better differentiate audio, video and other streams.
- When hovering over a TS/Demux source with the mouse cursor, the associated TS source is now highlighted (the other way around – hovering over a TS source and highlighting associated TS/Demux sources – did already work)

- Icecast server output streams now use the given name of the output settings as stream name (icy-name) instead of the fixed icy-name "2wcom Live Stream"
- The metadata (StreamTitle) used for the Icecast server output streams can now be set via the external APIs, e. g. SNMP (where it is the node virtCslAudioEncoderOutputsIcecastMetastr, OID .1.3.6.1.4.1.21529.1001.35.2.3.18.20.319.4096.1.5)
- The optional 192 kHz decoder output does no longer require a 1PPS or PTP external clock signal to work properly
- Removed the need to enable ancillary data usage for XLR inputs (actually there's no need for it – the usage can still be enabled/disabled via the profile)

Fixed Issues

- Elementary Stream decoding may exhibit audio errors even when no IP streaming issues did occur due to accidentally internal Stream4Sure activation
- Elementary Stream decoding in DualStreaming mode may exhibit audio errors even if the combined stream has no errors
- Fixed a problem with 44.1 kHz upsamling to 48 kHz (e. g. a 44.1 kHz input source played out at an output configured to have 48 kHz sample rate)
- RIST decoder accuracy improved - sometimes RIST requested packets unnecessarily
- MPEG TS decoder: fixed several issues with not decodable Layer 2 audio streams and audio decoding not starting after configuration changes (needing Decoder Off/On cycle to start up)
- MPEG TS decoder: when changing the audio PID to decode of a TS/Demux source, the change will not get active directly (needing Decoder Off/On cycle to get active)
- MPEG TS decoder: changing the RF intput of a SAT source was not applied instantly (needing Decoder Off/On cycle to get active)
- MPEG TS decoder: fixed some MPE reconfiguration problems in TS/Demux sources instantly (needing Decoder Off/On cycle to get active)
- MPEG TS decoder: improved character table support for e. g. service names
- Fixed a possible crash when changing the configuration of a SAT/TS input source and afterwards a TS/Demux source, which is using that SAT/TS source
- When deleting a TS/Demux source, the corresponding TS source delete button was not re-enabled
- SIP: Enhanced OPUS compatibility
- Incoming SIP calls for eSIP trunks fixed
- 'keep-alive' detection for optional HLS encoding
- Audio distortion in Mono (Downmix) mode for signals greater -6 dBFS
- Headphone output not working correctly with 44.1kHz audio
- Elementary Stream configuration dialog: The list of available streams via Ravenna/SAP is sometimes wrong
- Elementary Stream configuration dialog: When copying the settings of an available stream (announced via Ravenna/SAP), the Save button won’t get active

Known bugs

- n/a

Version 2.06

06.10.2021

New Functionality

- Added support for SNMPv3
- Added support for PTP Unicast
- Added optional HLS encoding capability
- Added Unrecovered counter to RIST info block for the Overview
- Added BER value to SAT Tuner info block for the Overview
- Rx (Receive) and Tx (Transmit) bitrates added to LCD status screen

Changed functionality

- Rework of the System Settings / User web page
- When the transponder input frequency method is used in the SAT configuration dialog, it’s no longer necessary to configure the frequency range (Low Band / High Band)

Fixed Issues

- MPEG TS decoder: fixed a possible memory leak (in case of PIDs referenced by multiple programs), resulting in a crash of the device after some time
- AAC decoding in MPEG TS did impose an audio delay of 2 to 3 seconds
- Lost packet counter for RTCP receiver reports (shown in the Encoder details on the Overview page) improved (negative value could occur in case of duplicate packets)
- In DualStreaming Elementary Stream setup, only the send delay of the first stream was shown on the Codec and the Overview page
- Increased disconnect/reconnect timeout for SRT connections in caller mode with no audio from 5 to 30 seconds
- DNS and Gateway entries in the TCP/IP settings couldn’t get deleted by entering 0.0.0.0

Known bugs

- n/a

Version 2.05.1

27.08.2021

Fixed Issues

- The switchable audio device mode did not work correctly – in Analog/Digital mode it did not allow to switch the signal type for the audio inputs

Version 2.05

20.08.2021

New Functionality

- Cold backups
  Backup input sources can now be set to “Active when needed” instead of being always active as it was the case until now.
  With e. g. Icecast input sources the server connection will only be established, when the backup is really needed, meaning superior input sources did fail, thereby not unnecessarily eating up e. g. possibly limited LTE volume.
- Switchable audio device mode
  If you have more than 2 channels licensed, until now the device turned into a Digital only device, meaning the audio inputs and outputs can no longer be switched to Analog mode.
  For such a device you now have the option to switch to Analog/Digital mode. You will thereby reduce the useable channel count to 2, allowing to configure only two audio outputs and four encoders, but you will regain the possibility to switch the inputs and outputs to Analog mode.
- Configurable maximum payload size for Elementary Streams, increasing the number of packets per second (to e. g. reduce the latency implied by FEC usage)
- Rx (Receive) and Tx (Transmit) bitrates added to Overview page (per interface)
- Optional support for SIRC (Satellite In-Band Remote Control)
- MPEG TS encoder: selectable mode for private data insertion (ES - stream type 0x89 or UECP/RDS – stream type 0x80)
- RIST support added to SIP
- Configuration option for the stream ID added to SRT caller mode
- Configuration option for maximum reorder tolerance added to SRT input sources
- Web Login/Logout actions are now logged in the EventLog
- When the connection to the device is interrupted, a dialog will pop up

Changed functionality

- FEC encoder: setting the column port offset to 0 will now enable a row only FEC
- In RTP DualStreaming setup (without FEC), the Missed counter of the first stream was in fact the overall missed counter (after combining the two streams). This is corrected – that Missed counter will now reflect

the missed packets of only the first stream. Instead there's a new block "Dual streaming" showing the Unrecovered/Missed counter of the combined streams.

- In RTP DualStreaming setup (without FEC), the PER (packet error rate) of the first stream was in fact the overall PER (after combining the two streams). This is corrected – that PER value will now reflect the PER of only the first stream. Instead the overall PER is now shown in the new "Dual streaming" block.
- In RTP setup with FEC, the Missed counter did not reflect the missed packets of the audio stream, but the number of packets not available after FEC correction (it was identical to the Unrecovered counter in the FEC block). This is corrected – the Missed counter will now reflect the missed packets of the audio stream (without FEC correction).
- In RTP setup with FEC, the PER did reflect the overall PER after FEC correction and not just the one of the audio stream. This is corrected – that PER value will now reflect the PER of only the audio stream. Instead the overall PER is now added to the "FEC" block.

Fixed Issues

- PTPv2 support was accidentally broken since firmware V2.03 (seldom audio dropouts)
- Livewire Routing Protocol (LWRP) was broken since firmware V2.00
- Ancillary data decoding in MP3 did not work
- PCM 24 bit elementary streams with ancillary data and FEC could cause audio distortions
- A crash could happen when using an encoder send delay for RTP streams together with PCM or Opus encoded data, which has ancillary data and/or GPIO Forwarding active
- Automatic reconnect in SRT input source caller mode, when no audio packets are received for some time
- SRT bitrate gauge did not show the current bitrate, but a mean value over the whole connection period
- PER calculation failed for SRT connections (resulting in always being 0.0% PER)
- Jitter measurement for first RTP stream in DualStreaming setup was wrong
- Elementary stream decoding may fail to start up correctly when a higher jitter is present (the dejitter buffer level is 0 and the audio buffer will not build up).
- SIP: Possible audio distortion a few seconds after the connection is established when connecting between two 2wcom 4audio devices
- SIP: Receiver reports were not shown on the Overwiew on callee side
- RIST info on Overview page was missing for TS/IP input sources
- RIST info on Overview page was wrong for input sources: the Requested counter did show the number of retransmitted packets and the Retransmitted counter was always 0
- SNMP node virtCslAudioDecoderInputsourcesRtpStatusFecunrecoveredcnt (OID .1.3.6.1.4.1.21529.1001.35.2.3.4.5.10.37.4096.1.23) was wrong and did not reflect the unrecoverable packets after FEC (but did reflect the number of internal FEC errors)
- MPEG TS decoder improved for high number of audio frames per PES
- MPEG TS encoder: SRT output didn't work if the entry was copied (only the first one did work)
- MPEG TS encoder: RTP output did not allow to select Ctrl interface, even if it was enabled for streaming via Services menu
- After pressing "Save" on the Ancillary Data configuration page, the embedding of ancillary data into a private TS PID did not work any more

- Decoder Streaming Input alarm shows yellow led, even if only Main source is configured and faulty (which should result in a red led)
- Distortions on headphone output
- The audio output may stuck completely in very rare cases, which could only be recovered by rebooting the device
- Ctrl interface was not selectable for ancillary output even if enabled via Services menu
- Ctrl interface was not selectable for TS Multiplexer outputs even if enabled via Services menu
- SIP Phonebook: buttons did not work in newly added entries
- The table height of the event table on the Log page is now automatically the max. window height

Known bugs

- With PTPv2 it’s still possible, that a small buffer drift does occur (around 1-2 ms per day). Currently under investigation
- Disable and enable an encoder stream (which is using SFN or external clock activated for audio output) within 10 seconds could lead into audio errors on decoder. Fixed in 2.06, workaround: reboot decoder or disable encoder for at least 10 seconds and enable it again.

Version 2.04

07.06.2021

New Functionality

- Finally added full (optional) Stream4Sure functionality (compatible to MM01 and MM08E)
- The audio outputs can now be used as regular input sources. When used by an Encoder, you can thereby benefit from the backup switching mechanism of the audio outputs, configured on the Decoder tab of the Codec page. This is especially useful if you don't need the audio output(s) itself, but are in need for a backup mechanism for the Encoder input source(s).
- You can configure meaningful names for the audio outputs.
- Added RIST (Reliable Internet Stream Transport, https://www.rist.tv/) support as an alternative to the already supported SRT protocol for a more standard based (RTP and SMPTE-2022) option to reliably transport your audio while being fully interoperable with standard RTP.
  We currently support the RIST Simple Profile, which in contrast to SRT already supports multicast, but on the other hand does not support encryption like SRT does.
  RIST will not require a new license, but will automatically be available if you have the SRT Encoder/Decoder license.
- Added VLAN support to the Ctrl interface

Changed functionality

- n/a

Fixed Issues

- MPEG TS encoder improved for input sources with a higher jitter
- MPEG TS encoder: External PCR will be signalled as "PCR" and not as "PES Priv. Data"
- MPEG TS encoder: improved compatibility with some decoders (TS stuffing bytes handling)
- Improved transcoding from MPEG TS to MPEG TS
- FEC encoder improved for RFC3640 and RFC4598, RFC4184 (marker bit, Dolby and MPEG4 AAC). The problem could manifest on decoder side as audio buffer running empty with AAC encoding and FEC enabled.
- The device could crash when switching the RF Input for a satellite input source
- New NTP status page is OK now, even if the device has no Ravenna or SFN license
- In SFN mode, LiveListening of an audio output was distorted
- When toggling the selected TS Multiplex on the Overview page, the output table did not update correspondingly
- For a newly added TS Multiplex, the "Auto-calculate required TS bit rate" toggle switch did not work (did not show the Bit rate input field in case "Auto-calculate" was switched off)

Known bugs

- n/a

Version 2.03

10.05.2021

Important!

- This update recommends an updated recovery system. Please update the device using the file **recovery_2.00.upd**. A reboot of the device is not necessary.

New Functionality

- Ancillary Data input and output does support now input and output via UDP, too
- A double click on a row in the Profiles, Input Sources or Encoder Outputs table will open the corresponding Edit dialog
- New Ancillary Data status page
- New Services configuration, allowing to enable/disable HTTP, HTTPS, SFTP, SNMP and Streaming Data access/usage for each network interface (including VLANs).
  This does also mean, that you can now grant access to HTTP(S) or SNMP via the Data interface – that's no longer limited to the Ctrl interface.
- The Ctrl interface can now also be used for Streaming Data like input source and encoder outputs (has to be enabled via the new Services configuration)
- New TCP/IP configuration page, allowing up to 10 VLANs for Data1 and Data2. Gateways can now be configured for VLANs, too.
- Added source specific multicast configuration to Elementary Streams dialog
- Added NTP status page

Changed functionality

- Ancillary Data handling
  The Ancillary Data handling setup has changed. Formerly the configuration (should DTE input be used in conjunction with an audio input / should DTE output be used to put out the Ancillary Data received via an Input Source) was done as part of the Input Source configuration dialogs.
  To clean this up and to add UDP support, the configuration for the Ancillary Data handling was moved to a new Ancillary Data menu item.
- New NTP implementation (allowing up to four servers to be configured), giving higher precision
- Several packages have been updated (see OSA – Open Source Acknowledgment – on the Global page) to fix e. g. security issues

Fixed Issues

- FEC Encoder now supports the maximum of 64 elementary output streams
- Encoder Outputs: it was not possible to have more than 16 encoder outputs for the same encoder (now all 64 encoder outputs can have the same encoder source)
- MPEG TS Encoder: Bit rate overhead reduced
- MPEG TS Encoder improved for ancillary data in private PID

- MPEG TS Decoder crash fixed on special EIT tables and large SDT tables
- Deleting all payloads of a service did not work - after page reload one service is kept
- The device could crash when starting or stopping RTP output streams
- Removed obsolete MPEG TS Packetizing format from Multiplexer settings
- When deactivating encoder and decoder input sources at the same time via the web interface, the device could crash
- Wrong analogue level display in web interface in cases where the device is digital only
- Added missing interface and VLAN info to input sources and encoder output on Overview page
- Inactive SAVE button in the Livewire+ dialog after changing only the LW source
- Added audio buffer configuration for Livewire+ input source

Known bugs

- The new NTP status page does only work, if the device has the Ravenna or SFN license

Version 2.01

10.03.2021

New Functionality

- Icecast Server Encoder Output added
  In addition to the Icecast Source Client Encoder Output it's now also possible to enable an Icecast Server as an encoder output. Each Icecast Server will allow a maximum of 5 client connections.
- DSCP configuration for Elementary Stream Encoder Output added

Changed functionality

- The configuration for the usage of an external clock source (1PPS, PTP) has been changed. In former firmware versions the usage of the external clock source had to be enabled for the specific input sources (Audio Input for the audio inputs, Elementary Stream Input for the audio outputs).
  As this led to some confusion, the configuration has been re-designed.
  The usage of the internal or external clock source is now configured on the Audio/XLR page, as the clock source is in fact a characteristic of the audio inputs and/or audio outputs.
  The new reference input selection introduced in firmware V2.00 will be (once enabled) the clock source master, therefore not allowing the selection between internal or external (1PPS, PTP) clock source.

Fixed Issues

- PSU alarms were not available (since firmware V2.00)
- Fixed a crash when changing the interface of an active SRT input source
- After device boot/reboot, some input sources may not start up correctly, when DHCP is used on the Data interfaces. The input source itself will work, but no audio is decoded / played out
- SRT input sources will now (again) be able to follow an SRT encoder, which does encode a file (this was broken in firmware V2.00)
- MPEG TS Encoder: PCR insertion improved for streams in “Low bitrate overhead” mode
- The Jitter information for the redundant part of an RTP output stream was wrong
- Improved Livewire compatibility (e. g. to Omnia One)
- The “Reset Counters” button on the Overview page did not reset the RTP output statistics for a redundant stream
- The “Reset Counters” button on the Overview page did sometimes not reset the FEC decoder status

Version 2.00

09.02.2021

New Functionality

- Reference Input: the audio clock control does now allow to configure one of the audio inputs to act as the reference input (clock source) for the outputs (or other inputs)
- Added missing separate status information (via SNMP/Ember+) for all input sources types and all encoder output types
- Configuration of VLAN added for Livewire input sources
- Configuration of the dejitter buffer level added for Livewire input sources
- Added No Input Data alarm for streaming inputs
- SDP files can be downloaded for the output streams
- MPEG TS Encoder: allow referencing an external PCR (earlier firmware versions always used one of the audio streams for the PCR)

Fixed Issues

- PSU alarms are not available, if the device is equipped with hot swappable power supplies
- The (SAT)TS-Bitrate was no longer valid, but always displayed as 0
- TS Sync and C/N switch criteria for SAT(TS) input sources did not work
- IP input source status information was accessible using the decoder index (audio output index / decoder rank) via SNMP/Ember+. Therefore it was not possible to get IP status information, if the input source is only used for an encoder.
  Now the index to use is the virtual input source index, following the visual order in the web
- Ancillary data output stopped working after switching to a backup source and didn't recover even if the main source is available again, but all input sources had to be disabled and enabled to get it working again
- DTE Ancillary input data is not routed to DTE output, if corresponding Audio Input is used as input source in the Decoder (Audio Output)
- SRT decoder was not restarted if the configured profile to use has been changed
- Activating NTP time synchronization does cause an audio glitch at synchronization time when using SRT
- Icecast Source Client encoder output did not allow to configure ports below 1024, but sometimes port 80 is used, so now every port starting with 1 is allowed
- Icecast Source Client did not work with AAC codecs
- MPEG TS Encoder: Optimized IP packet size for low bitrate
- SNMP Trap status binding delivers wrong value
- In rare cases the device may crash when changing the input source

Changed behavior

- SNMP Traps: ATTENTION – the old SNMP traps are no longer available!
  We made a move from device specific traps to general traps if applicable.
  Example:
  notificationVirtIp4cDeviceStatusMonitoringSilenceaudio1in (OID .1.3.6.1.4.1.21529.1001.0.7)
  is now available as
  notificationVirtCslMonitoringStatusAudioInputsXlrsilencedetection
  (OID .1.3.6.1.4.1.21529.1001.0.1098907649)
  Please note however, that in contrast to the traps the old status information, which can be polled, is still available, e.g. virtIp4cDeviceStatusMonitoringSilenceaudio1in (OID .1.3.6.1.4.1.21529.1001.35.1.23.37.57.7).
  This old information is however marked as Obsolete in the SNMP MIB, as in the future only the new status information will be available
  (e.g. virtCslMonitoringStatusAudioInputsXlrsilencedetection, OID .1.3.6.1.4.1.21529.1001.35.2.57.37.3.313.4096.1.1)

Known bugs

- **None**

Version 1.31.4

11.12.2020

New Functionality

- None

Fixed Issues

- HTTPS web access to DATA1 and DATA2 disabled
- SFTP upload files cannot be saved

Known bugs

- PTP synchronization incorrect after reset (turn off, turn on)
  Workaround: Reboot the device after configuring your PTP setup
- When using SAP all multicast parameters are set correctly, but the audio parameters are not used from the session description.
  Workaround: setup the audio parameters manually when you configure the reception of an SAP announced stream.

Version 1.31.3

30.11.2020

New Functionality

- None

Fixed Issues

- Fixed a routing problem when Ctrl and Data 1 or Data 2 were in the same subnet. This issue could result in the traffic being sent over the gateway instead of the subnet. Depending on the network setup this could cause unreachable devices.

Known bugs

- PTP synchronization incorrect after reset (turn off, turn on)
  Workaround: Reboot the device after configuring your PTP setup
- When using SAP all multicast parameters are set correctly, but the audio parameters are not used from the session description.
  Workaround: setup the audio parameters manually when you configure the reception of a SAP announced stream.

Version 1.31.2

05.11.2020

New Functionality

- None

Fixed Issues

- Device not reachable because of wrong or missing gateway
- IP VLAN changes are not updated after configuration - only after reboot
- Redundancy stream handling in SFN mode improved
- Headphone output did not work in SFN mode
- Enhanced compatibility to some special Icecast streams

Known bugs

- None

Version 1.31.1

05.11.2020

New Functionality

- None

Fixed Issues

- Log page shows error message about wrong localtime
- Device not reachable because of missing gateway. Has been fixed by changing DHCP metrics.
- Ravenna compatibility problems (e.g. to Lawo devices)

Known bugs

- None

Version 1.31

28.10.2020

New Functionality

- Web Interface: Profiles are now moved to a separate tab, separated from the Input Sources

Fixed Issues

- Audio output may stop after several days or weeks (although input source(s) are still running)
- A network configuration using different gateways for the different interfaces (CTRL, DATA1, DATA2) did not work properly (policy based routing)
- Web interface is not available in seldom cases after a reboot (if DHCP is used for the CTRL interface)
- PSU failure alarm may produce false positive alarms
- “AES/EBU No signal" alarm did not work
- TS Decoder: audio decoding stops after changing demux configuration in MPTS setup
- SIP: Re-dial active connection after reboot if reconnect is enabled
- SFN mode: output disturbed in DualStreaming setup, if only redundant stream is received
- Web interface / Codec: VLAN may be set to 0 instead of right value in edit dialog
- VLAN may get set unintentionally, if interface is changed from one with VLAN to one without VLAN
- Web Interface: Overview page is empty (more or less), when logged in as user
- TS Encoder: enhanced ATSC compliance

Known bugs

- None

Version 1.30

15.09.2020

New Functionality

- TS Encoder: added configuration possibility for the encoding standard (DVB/ATSC) and language codes
- On the TS Multiplexer web page, changes done are monitored now, too and the Save buttons will only get enabled when changes are present. Faulty values are not allowed to be saved.
- Added configuration possibility (on Audio XLR page) for the critical level marker threshold for the level meters on the overview page (threshold/level, at which the level will be shown as orange instead of green)
- PTPv2 support is now fully functional (provided the Ravenna license is present)
- Added support for the optional FM Dualtuner module

Fixed Issues

- SAP service behavior on DHCP changes improved; sometime SAP was not available after changing the network interface to DHCP
- RTP jitter buffer performance and accuracy improved PSU failure alarm may produce false positive alarms
- Problems with MP3 VBR decoding
- Wrong RTP timestamps for G.722 encoding
- Ancillary data output not working in case of SRT input source
- Ancillary data not working with E-aptX and 32 kHz sample rate
- DHCP leases and TCP/IP reconfiguration improved

Known bugs

- none