IP-8e

Release Notes

Version 2.13

02.06.2023

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Version 2.13

02.06.2023

New Functionality

- Added Mono Downmix option to headphone output
- 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 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 gain configuration to all input sources
- Added playlist support (m3u, m3u8, pls) to the file input source
- Added VLAN support to the Icecast client input source
- Major Livewire integration enhancements:
  - Added optional Livewire Sources for the audio inputs, providing its physical XLR audio inputs as Livewire audio streams.
  - 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).
  - 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-8e to the GPIs of a different Livewire device.
- Added status tab to Overview page for optional Livewire Sources
- Added PIN lock option for LCD menu
- HLS encoder: added support for FLAC and Opus
- Added VLAN interface status

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.

- 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

- 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
- 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
- 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
- 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
- Fixed a possible crash with NTP synchronization and “Bind to interface” option enabled
- 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 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
- AES67 input: fix problem with mono input streams
- Fixed MPEG TS signaling for MPEG2-AAC
- Improved transcoding of MPEG TS with ancillary data
- 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 Livewire routing, advertisement and GPIO configuration
- Added support for Multichannel AES67/Ravenna input streams – it's possible to select the channel(s) to use from it
- Added support for Multichannel AES67/Ravenna input streams – it's possible to select the channel(s) to use from it
- Added support for FEC according to RFC 2733 (e. g. used by Digigram IQOYA)
- Added support for user authentication to Icecast client
- Added the additional possibility to configure a DNS server per IP interface (including VLAN)
- 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 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.

Fixed Issues

- 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
- Switching off all TS/IP multiplexer outputs did switch off a possible active ASI output, too
- 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)
- Changing e. g. a file input source used by an encoder (changing the configured file) lets the encoder stop encoding

- 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

17.06.2022

New Functionality

- Completely revised ancillary data handling for full flexibility
  The fixed linkage between ancillary inputs and corresponding audio inputs is entirely removed. Instead a variable number of UDP ancillary inputs can be configured besides the available DTE inputs.
  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.
- 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.
- Optional audio PID removal from TS Multiplexer in case of audio input loss (to e. g. trigger external backup)
- Added new ancillary input alarms for the dedicated inputs
- 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)
- Ember+: added support for "subscribe" mode

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
- 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 4 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
- 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)
- 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
- 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
- MPEG TS decoder: further enhancements to private data decoding
- 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

Known bugs

- n/a

Version 2.08

07.02.2022

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
- SNMP/Ember+ status information for 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.
- 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

28.12.2021

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.
- 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

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)
- 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 in DualStreaming mode may exhibit audio errors even if the combined stream has no errors
- 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: 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
- 'keep-alive' detection for optional HLS encoding
- PTP was not available as external clock source if the Livewire license is present, but the Ravenna license is missing
- 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
- 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

25.08.2021

New Functionality

- NTP synchronized playout
- Support for Differentiated Services DSCP tagging
- Switchable audio device mode
  If you have more than 4 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 4, allowing to configure only four audio outputs and eight 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).
- 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
- The table height of the event table on the Log page is now automatically the max. window height
- AES sync lost counter was incorrect

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)
- 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 MPE satellite input source
- New NTP status page is OK now, even if the device has no Ravenna or SFN license
- 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 does support now input 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 via 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) was done as part of the Input Source configuration dialog(s).
  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
- 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

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. 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.

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

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 1.31.3

24.11.2020

New Functionality

- None

Fixed Issues

- Level meter did not work correctly, showing wrong (variant) values for a fixed level signal
- Wrong LCD display, if 5 channels are activated

Known bugs

- None

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