Audio Live Product Release

Product Release: v1.5.2.18

Release Date:	14th November 2019
Supported Operating System(s):	Linux 64-bit (Ubuntu 16.04)
System Components:	GV Live (v1.5.2.18) SAMLive_1U_Standard_56Core_1.2.tar (v1.2) SAMLive_1U_Standard_64Core_1.0.tar (v1.0)

Audio Live v1.5.2.18 (14/11/19)

New Features

1. The GV Live console application has been improved to provide more information when upgrading a product.
2. The GV Live Host service now detects and verifies the hardware configuration of the unit to ensure that all PCIe devices are correctly inserted in a supported slot configuration.
3. The GV Live console application now compresses the operational logs before transferring them to USB which speeds up the process; it reports on the progress of the transfer and captures additional settings to better record the state of the unit.
4. The GV Live Host menus now include an ‘About’ page which covers open source software license agreements.
5. The GV Live console network configuration now includes an ‘Advanced’ option which is an alternative to the basic approach offered previously and allows full manual configuration of the unit’s Debian-style network script; this allows more sophisticated routing options should they be required.
6. The GV Live console now offers a ‘Diagnostics’ menu option which displays the network routing table and allows ping requests to be sent from any of the network interfaces.
7. Audio Live now has an additional 8x64, 96x16 spigot configuration.
8. Changes made to Audio Live control and monitoring allow it to better integrate with Grass Valley’s IP Routing controller GV Convergent.
9. Audio Live now stores the interface names as well as the IP addresses for its network media ports; this allows Primary and Secondary selections to be maintained if the IP addresses are changed and the unit is restarted.
10. Audio Live now supports next generation (v3) Turnkey hardware (1RU, 2x2x16 CPU core).
11. Added support for 2110-31 audio streams to Audio Live: transparent channel pair routing and translation to/from 2110-30 (assuming PCM payload).
12. Additional controls have been added to Audio Live to set the RTP Payload Type ID.
13. Removed the automatic selection of 500us packet time for Audio Live output flows with a channel count between 9 and 16 as it isn’t well supported by other IP products; this packet time can still be manually selected.
14. The channel count control for Audio Live output streams can now be set to ‘Max’ which automatically configures the output for the maximum available channel count given the output spigot configuration and the selected packet time option.
15. Allow proprietary RTP header extensions to be enabled or disabled on a per-output basis for Audio Live.

Bug Fixes

1. Prevented the display of an erroneous error message on the GV Live console application when upgrading an agent type without SDI cards (for example Audio Live).
2. The GV Live console application is now case insensitive when scanning a USB drive for new licenses (previously upper case file extensions caused a license file not to be detected).
3. The GV Live Host service now no longer deletes an agent GUID when clearing its settings; this allows its identity within IP Routing software to be maintained.
4. Corrected a bug with Audio Live that caused the agent to fail when set to auto detection of channel count whilst receiving audio flows generated by Grass Valley's Integrated Playout Platform iTX.
5. Fixed an Audio Live memory leak that occurred when reconfiguring the agent using the TAKE button on the RollCall 'Setup' page, or when input flow details were updated.
6. Corrected the 'Source' and 'Destination' labels on the Audio Live 'Input' and 'Output' menu pages.
7. Removed an erroneous 'Not Locked' warning message which could sometimes incorrectly be reported for Audio Live inputs.
8. Fixed a bug that prevented the primary flow of an Audio Live from being stopped even when disabled in the RollCall menus or via routing control software.
9. Audio Live now populates the SSRC bytes in the RTP header of its output audio flows; previously these bytes were fixed at zero.
10. Default stream names for Audio Live are now reinstated if the spigot configuration is changed: this removes the potential for name corruption after reconfiguration.

Audio Live v1.5.1.0 (07/02/19)

New Features

1. The installer package name has changed to reflect the re-branding from SAM Live to GV Live.
2. Audio Live will now automatically select suitable network interfaces for primary and secondary RTP media interfaces prior to manual configuration override.

Bug Fixes

1. Corrected a bug introduced in the 1.5.0.14 release that prevented Audio Live from responding to a PING request.
2. Corrected a bug introduced in the 1.5.0.14 release that caused Audio Live to incorrectly report input streams as LOST if shared Multicast addresses were used.

Audio Live v1.5.0.14 (25/10/18)

New Features

1. SAM Live has been renamed GV Live.
2. To assist in the licensing process, the unit lock code is now displayed on the GV Live console output main menu.
3. Network interface configuration can now be performed via the GV Live Host RollCall template.
4. The system clock can now be set via the GV Live Host RollCall template and can be configured to synchronise automatically with an NTP server.
5. The GV Live console application now shows the details of any licenses that have been added to the unit.
6. The GV Live Host now details the configuration and status of all the unit's network interfaces via RollCall logging.
7. The GV Live console application now captures extra system logging information in addition to the GV Live operational logs when the 'Retrieve Operational Logs' option is selected.
8. Audio Live can now receive IP streams that do not contain proprietary header extensions; it will still automatically determine the number of input audio channels per stream.
9. Proprietary header extensions can now be disabled at the output of Audio Live.
10. Audio Live now offers a per-stream setting to configure the output packet time (up to 1ms); packet times above 250us require Audio Live to be switched into a different mode which can be accessed via the RTP Output Configuration settings on the Setup page of the RollCall template.
11. If a restrictive spigot configuration has been selected for the input or output of Audio Live, the channel restriction is now indicated on the RollCall template via the labelling of the Show Stream control.
12. Audio Live now listens for spigot name information via DDS and updates the RollCall template Input, Output and Routing pages accordingly.
13. A new 'Safe' Control Mode has been added to Audio Live that prevents settings that could disrupt the audio processing from being changed via the RollCall template.

Bug Fixes

1. Fixed a GV Live console bug that caused an error if an attempt to add a license was made with a USB drive that contained too many files.
2. Fixed a bug with the GV Live console application that caused it to fail if multiple USB drives were inserted when the 'Retrieve Operational Logs' option was selected.
3. Removed the option to set input and/or output channel and stream delays when the input spigot configuration is set to 192 streams x 8 channels; this input configuration combined with additional delays was overloading the system and causing spikes in system latency which risked interrupting audio processing.
4. Audio Live spigot change requests are now scheduled so a reply to the request can be returned quickly; this prevents errors occurring when multiple routing requests are sent to Audio Live within a short space of time.
5. Audio Live now correctly reports the speed of its network interfaces via DDS; interfaces were previously reported as 10Gb/s irrespective of the hardware that was fitted.
6. The initialisation of Audio Live has been reviewed and improved to address reports that audio processing was sometimes incorrect at start-up and the Timestamp Tolerance control had to be adjusted to correct the problem.
7. Fixed a bug in Audio Live that could occasionally cause an application error and agent failure directly after changing the channel count of an output stream.
8. Fixed a bug in Audio Live that caused incorrect behaviour if an input spigot was configured with a Multicast address but with a missing Source IP address.
9. Fixed a bug in Audio Live that caused it to show an incorrect Timestamp Tolerance setting on agent start-up if the input spigot configuration had been changed.
10. Prevented Audio Live from occasionally wrongly reporting a reference timestamp misalignment when configured for Auto reference stream selection.

<u>Audio Live v1.3.0.16</u> (24/04/18)

New Features

1. Audio Live now offers a number of different spigot configurations on the input and output enabling a more flexible, asymmetric routing matrix.
2. Audio Live will now synchronise up to 16 of its input streams to match the timing of its reference stream.
3. The time at which Audio Live locked to a reference stream is now available in a RollCall log field.
4. Audio Live now reports the packet time of all its input streams on the RollCall template.
5. The Audio Live channel matrix can now be controlled via DDS using the SAM IP Routing (IPRA) software.
6. The license entitlement ID (EID) is now shown on the SAM Live Host template.
7. All times shown on the RollCall template and logged are now UTC and are labelled accordingly.
8. System Shutdown and Restart buttons have been added to the RollCall template for the SAM Live Host.

Bug Fixes

1. The ‘START ALL’ and ‘STOP ALL’ buttons on the SAM Live Host template are now greyed out when they do not offer a useful function (for example, ‘STOP ALL’ is not enabled if there are no running agents).
2. The ‘Add New License’ option on the SAM Live console now allows for license files with a .TXT file extension.
3. The Audio Live reference locking logic has been adjusted so that the primary reference stream is always chosen ahead of the secondary if both reference streams are available.
4. The Audio Live channel matrix port numbering has been changed from a 0-based to a 1-based number range to make it consistent with other SAM products.
5. The Audio Live spigot numbering as reported in the operational logs and in the RollCall logging has changed from a 0-based to a 1-based number range to make it consistent with the numbering on the RollCall template.
6. Audio Live will no longer allow an input stream with greater than 1ms packet time to be used as a reference or to dictate the system timing; streams with high packet times are unsuitable to supply reference timing as they do not maintain a sufficient update rate.
7. The Audio Live front-end packet store (used for packet re-ordering and redundancy) is now automatically scaled to ensure that it imposes the same latency on all incoming streams regardless of their packet time.
8. Details of the SAM Live’s management network are now published via DDS to allow the SAM Live Host RollCall template to be opened directly within the Orbit Network Monitor application.
9. Fixed a bug causing very slow shutdown of the SAM Live RTP management service.
10. Fixed a bug that could occasionally cause a crash of the RTP management service when joining an IGMP stream.
11. Fixed a bug that could occasionally cause a RollCall timeout when the main TAKE button was clicked on the Audio Live Setup page of the RollCall template.

Audio Live v1.2.1.8 (26/01/18)

New Features

1. The xStream framework has been ported from Windows to Linux (Ubuntu 16.04) and renamed to SAM Live.
2. Preset agent configurations are now available in the SAM Live Host based on standard hardware specifications.
3. The SAM Live Host now automatically includes a license server (running at 'localhost') and is not able to pick up a license from a remote server.
4. The SAM Live Host now reports the lock code of its specified license server.
5. Audio Live (previously Audio XS) now offers per-channel delays at its input and output (previously only per-stream delays were available).
6. The maximum delay that Audio Live offers at input and output has increased from 2 seconds to 5 seconds.
7. The SAM Live Host now reports the server hostname on the Setup page of its RollCall template.
8. A new control has been added to Audio Live to allow the front-end timestamp alignment to be disabled.
9. The Audio Live operational logs now report which input stream a packet drop has been reported against.

Bug Fixes

1. Audio Live (previously Audio XS) automatically clears misleading reports of multiple dropped packets on all inputs at start-up.
2. Audio Live now reports Network FAIL if Primary and Secondary network interfaces are unconfigured.
3. RollCall saveset restoration for Audio Live has been corrected so that settings are recalled even if there is no valid network configuration.
4. Fixed a bug that could cause an Audio Live agent failure when increasing the channel count of an active output stream.
5. Corrected the logic used to establish if the total output channel count exceeded the allowed level for a given hardware specification.
6. Corrected the Audio Live menu logic to prevent the same input stream from being selected as the Primary Reference and the Secondary Reference.
7. Corrected the initialisation of the controls to enable/disable the available network interfaces for DDS traffic; it was being performed too early after a system reboot before all the available interfaces had been established.
8. Fixed an error in the RollCall saveset generated by the SAM Live Host as it could cause a crash when reapplied.
9. Stopped the Unit Name from being stored in the RollCall savesets of any of the SAM Live agent types.
10. Corrected the System Latency reporting in the Audio Live RollCall run log so that it matches the value reported on the RollCall Template.

<u>Audio XS v1.0.1.38</u> (14/11/17)

New Features

1. Audio XS will now continue to process input packets received that are outside the PTP timestamp range set up by the timing reference stream even though it is not possible to align the underlying audio.

Bug Fixes

1. [NONE]

Audio XS v1.0.1.36 (09/10/17)

New Features

1. To improve the robustness of Audio XS, manage the rate of update using the nominated primary and secondary reference streams.
2. Detect and ignore Audio XS input streams with an invalid sample rate.
3. Detect and report (for guidance only) Audio XS input streams that are receiving too many packets to be locked to the reference.

Bug Fixes

1. Fixed the occasional misreporting of Audio XS timestamp offsets due to incorrect negative number handling.
2. Prevented the application of an input with an unreliable or unlocked packet rate from impacting correct operation of Audio XS for the other inputs and outputs.
3. Prevented the application of an input with continually changing channel count from swamping operation of the Audio XS RollCall template.
4. No longer latch the Audio XS input buffer overflow error as this is misleading after the buffer has recovered.

Audio XS v1.0.1.28 (17/07/17)

New Features

1. [NONE]

Bug Fixes

1. Reverted re-ordering of Audio XS spigots from release 1.0.1.26 to prevent disruption to pre-release installations.

Audio XS v1.0.1.26 (21/06/17)

New Features

1. Increased the number of configurable xStream Agents from 10 to 16.
2. Audio XS now logs the input stream it used to provide reference timing.
3. Added a control to Audio XS to allow selection of a secondary choice for reference input stream; this will be used if the primary selection is not available at a time when locking is required.
4. Added a control to Audio XS to set the wait time allowed for the selected reference input stream to arrive; during this period, audio processing will not be possible.
5. Added controls to the xStream Host which allow the assignment of CPU cores for running agents.
6. Added an option to Audio XS that ensures that it will reject a routing request if the number of audio channels routed exceeds the stated limit.
7. Added Set and Preset buttons to the Input and Output pages of the RollCall template for MBG XS and Audio XS.

Bug Fixes

1. Fixed a bug that stopped the Audio XS and the MBG XS from stopping cleanly.
2. Prevented xStream Agents from logging multiple erroneous errors after their inputs are reconfigured.
3. Ensure that Audio XS is correctly announced to Orbit if it is started before the network is available.
4. Fixed the dropping out of Audio XS input streams with packet times of 1ms and 4ms.
5. Fixed a bug that caused disruption to all streams and re-evaluation of the reference timing if an asynchronous, out of range input was routed to Audio XS.
6. Fixed an occasional Audio XS crash if the number of channels sent within an output stream was reduced while the stream was connected.
7. Re-ordered the Audio XS spigot numbering to match the IQMIX and IQAMD products.
8. Fixed a bug that caused the MBG XS to crash when simultaneously receiving SMPTE 2022 and RFC 4175 packets from the same source spigot.

<u>Audio XS (Pre-release) v1.0.1.20</u> (04/04/17)

New Features

1. Added a control to Audio XS to allow the reference input stream to be manually selected (as long as the reference input is present at the time).
2. Added a control to Audio XS to allow configuration of the input packet buffer size allowing greater delay between redundant networks.
3. Added the publishing over DDS of device status information by Audio XS.

Bug Fixes

1. Fixed a bug in Audio XS that caused disruption to all output streams when the input stream delay of any input was changed.
2. Fixed a bug that could cause RTP spigots to broadcast if the multicast address of the secondary output was left unconfigured but the spigot was enabled.

Audio XS (Pre-release) v1.0.1.16 (23/03/17)

New Features

1. Now displays the RTP timestamp offset (in ms) between each input stream and the Audio XS reference.

Bug Fixes

1. Fixed a crash that could occur when changing the number of output channels for an Audio XS output stream with a non-zero delay.
2. Fixed an Audio XS memory leak that occurred if input streams were present but no output streams were enabled.
3. Fixed an Audio XS bug that caused corrupted audio for input streams with a packet time of 250us.
4. Prevented the network statistics from falsely showing traffic on a disabled network.
5. Fixed a bug that caused the Audio XS to sometimes hang when stopping the agent or re-starting its network interfaces.
6. Corrected start up behaviour when one or more networks have failed: Agents will not now begin processing if any of their configured networks are not present. This avoids the danger of starting up without network redundancy and being unaware.

Audio XS (Pre-release) v1.0.1.12 (10/03/17)

New Features

1. [NONE]

Bug Fixes

1. Corrected spigot ordering to be ‘source’ followed by ‘dest’; this now matches other SAM products.
2. Fixed a bug that stopped the Audio XS from starting if the primary network was disabled or unplugged.
3. Fixed a bug that caused occasional packet drops at the input of Audio XS when using redundant networks.
4. Allowed for mis-alignment of primary and secondary networks to prevent a brief but audible discontinuity if one of the networks was removed.
5. Added automatic re-evaluation of input PTP timestamps if the only inputs available to the Audio XS had a timestamp error (i.e. all other previously good inputs had been lost).
6. Added backup of spigot configuration to provide a work-around for previously seen problems with configuration being lost.
7. Prevented mis-leading ‘buffer overflow’ warnings from being occasionally displayed by the Audio XS (and having to be manually cleared) when a new input arrived.
8. Fixed a bug that caused a discontinuity in established ‘good’ inputs to Audio XS if a new ‘bad’ input arrived (with out of range PTP timestamps).

Audio XS (Pre-release) v1.0.1.8 (24/02/17)

New Features

1. Version of SAM Turbo UDP network driver now reported on xStream agent RollCall template.
2. Added support for RollCall savesets to Audio XS and MBG XS.
3. Created a RollCall log field that reports the master health of the Audio XS.
4. Extended the number of possible input stream replications in the SAM Turbo UDP network driver from 16 to 64.

Bug Fixes

1. Windows shutdown now captured in xStream log files.
2. Fixed failure of agent to restart on host server reboot.
3. Fixed an occasional agent crash when clicking TAKE on the RollCall template Setup page.
4. Fixed an occasional crash when setting output delays for Audio XS.
5. Fixed an issue that prevented Audio XS from supporting more than 32 audio channels per stream.
6. Fixed an occasional crash when navigating through different stream selections on the RollCall template Output page of Audio XS.
7. Fixed a bug that caused Audio XS to crash if two unrelated streams were routed to its input as primary and secondary redundant flows.
8. Fixed an occasional Audio XS crash when disabling an output spigot.

Audio XS (Pre-release) v1.0.1.0 (12/01/17)

New Features

1. High performance network driver now included in the xStream installer (although requires manual installation).

Bug Fixes

1. Now able to extract a PTP timestamp for SMPTE 2022 by translating from the RTP timestamp information.
2. High performance network driver now allows the same flow to be routed to more than one spigot within the host server.
3. Fixed the name of license log-fields from _STATUS to _STATE (to match how the template names them).
4. xStream host is now more robust to a loss of configuration in the event of a crash.
5. High performance network driver now allows multiple xStream agents to run in parallel.

Known Issues

1. Audio XS does not allow more than 32 audio channels in a stream; this will be fixed in the next release.

Audio XS (Pre-release) v1.0.0.12 (22/12/16)

New Features

1. First release.

Bug Fixes

1. First release.

