



## Axia Audio Node Update Instructions

With Release Notes – v2.5.2g for Analog Node, AES Node, Microphone Node, Router Selector Node, GPIO Node

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### AXIA NODE SOFTWARE

An Axia node has two internal memory “banks”. Each bank contains room for a complete version of operating software. This approach allows a software update download to be completed and checked without danger of making the unit inoperable if the download were to be incomplete or corrupted. It also provides an easy way to try a new software version and still return to the old version.

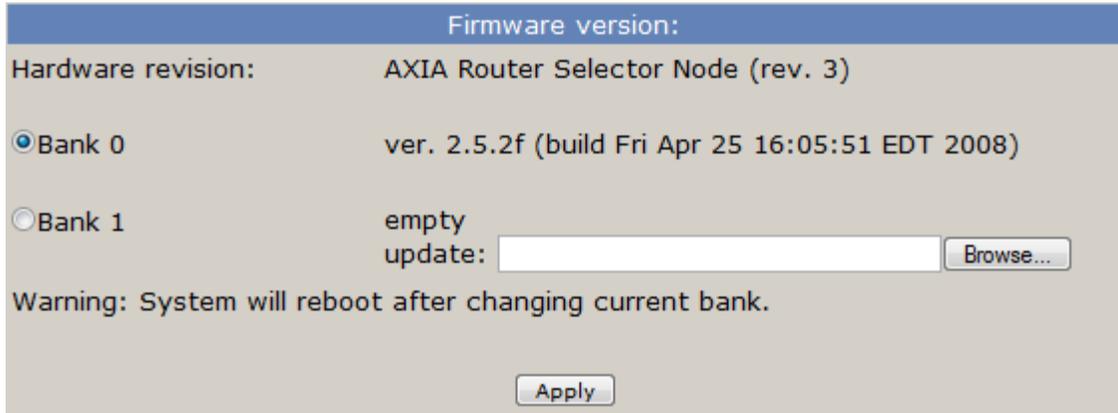
The software version in each bank is displayed from the System web page. The lower half of this screen, shown below, shows the current firmware versions and allows you to select what version will be used at startup. To change banks simply click in the “radio button” for the desired bank and then click on Apply.

*IMPORTANT! The node will reboot after you click Apply if you change the software version. This will result in loss of audio locally, and at any unit using the local sources.*

#### Downloading new software

A new software version can be downloaded into bank 1 as follows:

1. Visit [www.axiaaudio.com/downloads/](http://www.axiaaudio.com/downloads/) and download the desired software update for the node to be updated to your computer (this should be the computer that you will use to access the node’s web page). Your local computer operating system should display a prompt to permit you to choose where you wish to locate the downloaded file. You can choose any convenient location, just be sure to note the drive and location where the file is to be saved.
2. Open a web browser and connect to the node to be updated. Enter the complete path and file name for the software file (e.g. the file downloaded from the Axia site), or click on the Browse button to locate the file. Once the proper path and filename are displayed, click on Apply to download the file.
3. A successful download will be indicated by the new version being displayed in the Bank 1 field. If the download is unsuccessful the field for Bank 1 would be blank.
4. Click the radio button for Bank 1 and click “Apply” to reboot and make Bank 1 active.

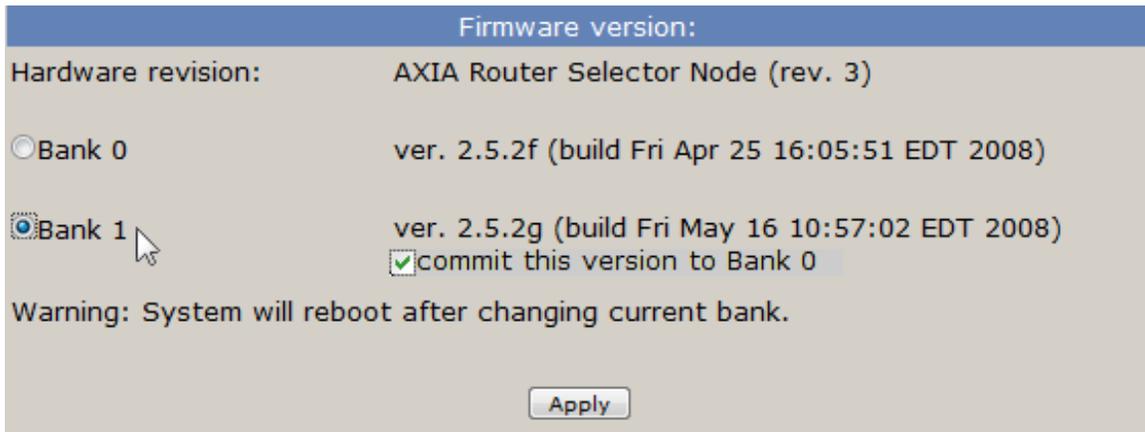


*“System” screen showing empty Bank 1 prior to uploading new software*

### Saving Bank 1 Software

Software is always downloaded to Bank 1 (the secondary bank). Downloading software (see below) will overwrite the software currently in this bank, if any. If you wish to save the software currently residing in Bank 1 you can save it by moving it to Bank 0 as follows:

1. Make sure Bank 1 is active – its radio button should be “on” and you’ll see the screen shown below.
2. Place a checkmark in the “Commit this version to Bank 0” box (see below).
3. Click on Apply. The node will now reboot. After reboot, you’ll see that the software that was in Bank 1 has been transferred to Bank 0, and Bank 1 is empty again.



*Check “Commit” box prior to moving Bank 1 software to Bank 0; select “Apply”*



## Software Release Notes

Version 2.5.2g, 07/28/2008

### New:

- Added to All Nodes – Syslog messages notifying about changes in AES inputs synchronization state, as well as AES/Analog automatic switch.
- Added to AES/EBU and Analog Nodes – Mono Mode. In Mono Mode every other port is used. This supports splitting and combining of dual mono sources.
  - Each source can be independently configured as mono or stereo.
    - Sources 1, 3, 5, 7: in mono mode, left channel of 1, 3, 5, 7 inputs are duplicated to L and R and sent to corresponding Livewire stream.
    - Sources 2, 4, 6, 8: in mono mode, right channel of 1, 3, 5, 7 inputs are duplicated to L and R and sent to corresponding Livewire stream.
  - Destination pairs (1,2), (3,4), (5,6), (7,8) can be configured in dual mono mode by changing type/mode of the lower port. The following options are available: "From source: Dual Mono", "To source: Dual Mono". Left channel from Livewire stream corresponding to the lower port feeds output channel left. Left channel from Livewire stream corresponding to the upper port feeds output channel right. Lower and upper ports are fed the same signal.
- Added to Router Selector – New AES related features controlled by settings on the QoS WEB page.
  - AES sync source [Livewire 48kHz/AES sync/AES input] selects which clock source is used by the AES output.
    - In "Livewire 48kHz" position, default Livewire internal clock is used.
    - In "AES sync", the dedicated AES sync input is used. To use this mode, make sure the jumpers inside the unit are in SYNC position. If there is no valid input signal present, the AES output will be inactive.
    - In "AES input" mode, the input signal provides a clock. If there is no valid AES input the output chip will default to Livewire internal 48kHz timebase.
  - AES mute on LW stream fail. If this feature is enabled, the AES output will be active only if there is a valid audio stream being received from the Livewire Ethernet network.
  - Added AES synchronization related controls on the WEB pages.
- Added to Router Selector and AES Node – Continuous software monitoring of AES circuits to correct momentary error conditions caused by interruption of the AES signal.
- Added – Visual identification of allocated sources. A "padlock icon" identifies these sources and a popup window, accessible from the "Sources" page, shows real-time source allocation status.
- Added to AES node – Output gain control on the "Destinations" page.
- Added – WEB interface improvement protects Nodes from data being submitted multiple times when the "Refresh" button on the WEB browser is used.
- Added to Router Selector – "Volume Mode" control on System Screen allows user to change the default function of the front-panel control knob.
  - In "Phones" mode, the control alters volume of the headphone output only.
  - In "Ph&Out" mode, the control alters volume of the headphone, analog and AES outputs simultaneously.
- Added to Router Selector – Front-panel display contrast control.
- Added to Router Selector – Channel list filter. This filter is configured on Destinations WEB page. Once the filter is specified, only listed channels will be available on the front

panel. Multiple ranges in the channel filter are supported (e.g. "1211,1212-1214,301-308").

**Fixes (all Nodes unless specified):**

- Fixed – Corrected QoS settings applied to audio Livestreams after changing destination.
- Fixed – FPGA loading issue present in 2.3.3d version, which was preventing the device from booting consistently. It is possible that unit will boot after multiple attempts. After unit boots, new software can be loaded to fix the problem.
- Fixed – Corrected 802.1p and TOS audio packets marking. Since 2.3.x, values set for Standard Streams were also incorrectly applied to Livestreams.
- Fixed – GPIO Node correctly handles port configuration changes.
- Fixed, Router Selector – Bug that could infrequently cause front panel reset.
- Fixed – Heat sensitivity problem with reads of audio port from FPGA.
- Fixed – Synchronization of Livewire to AES input.
- Fixed, Microphone Node – “Meters” page Java applet output gain adjustment range.
- Removed, Router Selector – Erroneous Surround option from the destination configuration.
- Fixed – HTTP redirection is now automatic after a Node's hostname is changed.

**Changes (all Nodes unless specified):**

- Changed – The factory default Livewire Master Priority default value on all Nodes has been changed from 0 (disabled) to 3. This change makes initial node configuration easier, since users who do not use the mode switch on the front panel and relied on the Nodes' HTTP interface to perform the entire configuration tend to forget to enable the Priority setting on the QoS page, resulting in having no clock in the network. The new default enables the clock automatically. *Note that with this change it is no longer safe to plug in an unconfigured, factory-fresh Node into a nonqualified network.*
- Changed – In Destinations WEB page, surround bundle name column is now hidden for consistency with Sources WEB page.
- Changed – Factory default for "Source shareable" attribute.

To provide “backfeed” and GPIO control, Element v2.0 requires sources to be non-shareable in order to allocate corresponding Livewire channel in exclusive mode. Element v2.0 will not allocate the required channel if GPIO and “backfeed” are disabled in the source profile — this is the new “listen-only” mode; Element v1.x does not have listen-only mode and allocates every source.

In Element v1.x, the only way to share a source was setting the “source shareable” value to “yes”. But then the mechanism that prevents multiple Elements from sending backfeed and control to the same source was not active. The new default suggests the correct setting for users. Although Element 2.0 installations do not use the “source shareable” attribute, the setting is retained in the nodes for compatibility with users running earlier versions of Element software.

- Changed – Nodes no longer allow allocating shared sources.
- Changed – Hostname field now allows only letters, digits and hyphen characters. All other non-allowed characters are automatically converted to the hyphens (-).
- Changed – HWID (Hardware ID field) is updated in Livewire advertisements after a local IP address change.
- Changed, Router Selector – "Nominal analog level" moved from front panel System screen on the front panel to the Sources WEB page.
- Changed, Router Selector – Channel list sorting is now case-insensitive.



### **Version 2.3.3b2, 02/29/2008**

#### **Changes:**

- Changed – Factory Defaults. All sources are now set to “non-sharable” by default.

#### Explanation:

Element v2.0 software supports full sharing of sources, including those with backfeeds. An interlock prevents multiple consoles from sending simultaneous backfeeds or logic commands. Element shows the user the owner of the source in this case, so that it can be released. When a source is selected and has Backfeed enabled or Logic port (GPIO) enabled and another user attempts to select the same source elsewhere with the same settings enabled, an 'x' will appear where the channel number is shown, indicating the source is in Listen Only mode. Listen Only Mode allows auditioning of source audio, but no backfeed or GPIO control is generated.

Element v1.x software does not have this Listen Only mode, and allocated every source. The only way to share a source was to set sources as “shareable”, but this defeated the interlock that prevented multiple Elements from sending Backfeed and GPIO.

So with Smart Source Sharing implemented in Element v2.0, it is no longer necessary to set audio sources as “Sharable” using the settings contained on the Audio Node HTTP pages, and this software update sets all new sources as “non-sharable” by default. The option field is kept in the HTTP pages for compatibility with clients still running Element v1.x.

We recommend that you set all Node “Sharable” fields to “No” after upgrading to Element 2.0.

### **Version 2.3.3b1, 10/19/2007**

#### **Fixes:**

- Fixed - Logic timing problem that affected some nodes at temperature extremes.

### **Version 2.3.3b, 11/17/2006**

#### **Changes:**

- Changed - Default network mask changed from 255.255.255.0 to 255.0.0.0. This helps during setting up Axia in Class B. After assigning IP address to a device from the front panel, default network mask is used.
- Changed - Disabled 802.1Q VLAN ID field on QoS page. This parameter did not work consistently.
- Changed - New Default QoS values for Axia streams
- Removed - NTP server address field from “System Page”. Nodes do not run NTP client.

#### **Fixes:**

- Fixed - Increased process priority level of Livewire Routing Daemon, which is also responsible for GPI port scanning. This solves the problem with missing GPIO contact closures.
- Fixed - NVRAM access problem, which was causing Livewire Routing Daemon lockup after read or write failure.
- Fixed - For GPIO stand alone node, if user entered invalid IP address using Bootps application it sent 255.255.255.255 to the device. Node would accept this address and you can no longer communicate with it, Now IP address change to 255.255.255.255 is ignored by node.



- Fixed - default values of IP TOS field. Wrong values in the TOS field were causing audio packets forwarding delay in the switch and drop outs, when switch was configured to use TOS rather than COS.

**New:**

- Added- Syslog support. Various services generate syslog (RFC 3164) messages, which can be forwarded to remote daemon.

- 1.) Remote syslog daemon IP address can be entered on System WEB page. Entering a valid IP address ENABLES the logging. Entering 0.0.0.0 DISABLES the logging.
- 2.) User can choose log detail level.
- 3.) Port scanning rate is being monitored. Should the scan rate be larger than 30ms, a syslog (User.Warning) message containing the delay amount will be generated. Rate of those alert messages is limited to one every 30s.
- 4.) Enabling the Syslog severity level filter on "Debug" enables forwarding of GPIO pin changes information to the remote logger. Currently there is no rate limit for Debug messages and the "Debug level" is not recommended for setting and leaving this option enabled on all nodes for long periods. We recommend disabling the logging (set IP address at 0.0.0.0) and enabling only when troubleshooting severe issues with the node.

For testing, any syslog daemon can be used; such as a popular freeware Kiwi Syslog available for download from [www.kiwisyslog.com](http://www.kiwisyslog.com).