ISOSTEM

High end multichannel from stereo

ISOSTEM Expert

ISOSTEM Live

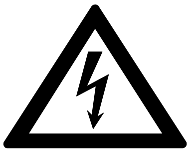
User Manual

DSPECIALISTS
Digitale Audio- und Messsysteme GmbH
Helmholtzstr. 2-9 L
D-10587 Berlin
www.isostem.de
www.dspecialists.de

Manual version: v1.2
Firmware version: Min. v3.0
GUI software version: Min. v3.0

CONTENT

SAFETY INSTRUCTIONS



This symbol, wherever it appears, alerts you to the presence of uninsulated dangerous voltage inside the enclosure – voltage that may be sufficient to constitute a risk of shock.

This symbol, wherever it appears, alerts you to important operating and maintenance instructions in the accompanying literature. Read this manual.

CAUTION: To reduce the risk of electrical shock, do not remove any screws of the enclosure. There are no user serviceable parts inside. Refer servicing to qualified personnel only.

WARNING: To reduce the risk of fire or electrical shock, do not expose this appliance to rain or moisture.

1. To assure best performance, please read this manual carefully.

2. Connect this appliance to a grounded AC outlet of 90 V to 250 V, 47 Hz to 63 Hz.

3. Keep the power cord in good condition. If the power cord becomes damaged, discard and replace it. Never isolate the ground of the AC power cord.

4. The power fuses are located on the rear panel of the appliance and may be accessed from the outside. In case the fuses have to be exchanged only use fuses of the same type as labeled.

5. The power switch of the device is located on the front panel of the appliance. The ON and OFF states are marked by “1” and “0” respectively.

6. Install this unit in a well ventilated, cool, dry and clean place. Keep it away from direct sunlight, heat sources, vibration, dust, moisture, or cold. In a cabinet allow about 2.5 cm of free space all around this unit for adequate ventilation.

7. The appliance must be adapted slowly to extreme temperature changes. These extreme changes may cause moisture inside that can cause failure and/or electrical shock.

8. Prolonged exposure to high volume levels may cause hearing damage and/or loss. The use of hearing protection in high volume situations is recommended.

INTRODUCTION

For many years now, TV technology has been developing at a rapid pace. Television in HD, 3D and surround is no longer merely a cinema experience, but can also be enjoyed in the comfort of your own home. The extremely high picture quality has also led to demands for improved sound, providing the broadcasting corporations with a considerable challenge - either a great deal of the material only exists in stereo and the original recordings are no longer available or creating a new mix on Surround 5.1 would be very expensive. As the consumer is no longer satisfied with stereo or alternating sound formats, what is now needed is an option for converting the stereo information into a multichannel surround signal in real time.

ISOSTEM® uses complex algorithms to automatically generate a multichannel version out of a stereo source signal - in real time at a very low latency of 40 ms and with perfect audio quality.
ISOSTEM® analyses the acoustic energy distribution of a stereo signal and separates dominant sources from ambient spaces by dynamic filtering (European patent office reference FR 2908586). These parts are distributed to the 5 channels and create a convincing surround signal. With ISOSTEM®, broadcasters can produce a continuous surround program - independent of the source material format.

In addition, ISOSTEM® exclusively offers management of the intermix (the difference between the downmix of the multichannel signal and the reference stereo signal) to assure the compatibility of the produced signals. (European patent office reference EP 2046076).

ISOSTEM® ensures that the EBU R.128 Programme Loudness of the original signal remains in the upmix and downmix results accurately.

Two hardware variants of ISOSTEM® are available: ISOSTEM® **Expert** contains a full range of functions and setting parameters. Individual setups can be created using the ISOSTEM® GUI software and saved as presets. ISOSTEM® **Live** was created for daily broadcast use where creating individual presets is not a relevant option. This version can use the factory presets as well as any other presets created on an ISOSTEM® **Expert** unit.

Both hardware variants are delivered with a GUI software that runs on Microsoft® Windows computers. The software is used for the real time configuration of all parameters, while the 1U hardware unit performs the necessary signal processing.

(This page is intentionally left blank)

QUICK START

Before going into the system's details, here is what you need to do to carry out a quick trial:

1. NECESSARY ITEMS

Check that you have the following items available:

- ISOSTEM® hardware unit (Expert or Live version)
- Power cable (supplied)
- USB to RS232 adapter cable (supplied)
- CD-ROM with ISOSTEM® GUI software (supplied)
- Driver CD-ROM for USB-to-RS232 adapter (supplied)
- A computer running Microsoft Windows® 2K/XP/Vista/7 (32-/64-bit) operating system with one free USB port
- Stereo audio source with AES3 digital output (2 channels)
- 5.1 surround playback system with 3 digital AES3 audio inputs (six channels)
- D-Sub 25 breakout cable (Tascam Pinout) to connect the source and the playback system to the ISOSTEM® hardware

2. SOFTWARE INSTALLATION

- **USB to RS232 adapter:** Install the appropriate driver of the USB-to-Serial converter which has been delivered with the unit. You find the drivers in the USB-to-Serial-Converter subfolder of the ISOSTEM® CD-ROM provided with the hardware unit.
- **ISOSTEM® software:** The **ISOSTEM®** software is provided on the supplied CD-ROM. Alternatively download the software from the website http://www.isostem.de/en. Start the installation procedure by execution of the file “setup_XXX.exe”

  After a successful installation you find the two GUI applications “Isostem Expert” and “Isostem Live” in the start menu of Windows.

  Use the expert version of the executable to connect to the Expert hardware variant and the live version to connect to the Live variant.

3. Cabling & Power On

- Connect a digital stereo audio source to the "AES In 4" connector of the D-Sub 25 breakout cable.

- Connect 3 digital AES3 inputs (6 channels) of your surround monitoring system to the "AES Out 1", "AES Out 2" and "AES Out 3" connectors of the D-Sub 25 breakout cable. Connect the breakout cable to the unit's AES/EBU I/O port. Use this channel scheme:

L:	AES Out 1 L	**R**:	AES Out 1 R
C:	AES Out 2 L	**LFE**:	AES Out 2 R
LS:	AES Out 3 L	**RS**:	AES Out 3 R

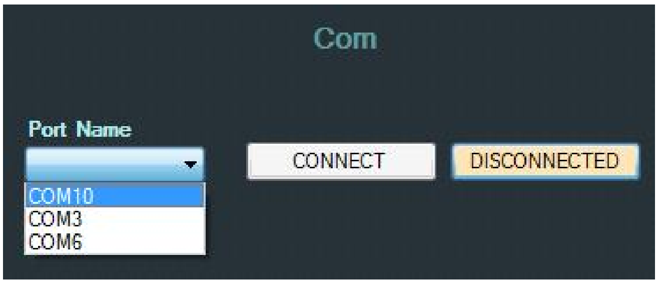
- The USB-to-RS232 adapter provided should already be connected to a USB port on your computer (see above chapter). If not, connect it now. Next, connect the adapter to ISOSTEM®'s RS232 port.

- Use the power cable to connect the ISOSTEM® hardware to a mains outlet.

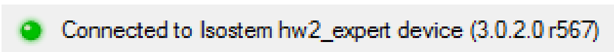
- Power on the ISOSTEM® hardware using the mains switch on the rear side of the unit.

4. Software Launch & Connection

From the Windows start menu execute the GUI program “Isostem Expert” or “Isostem Live” whichever suits your device variant.

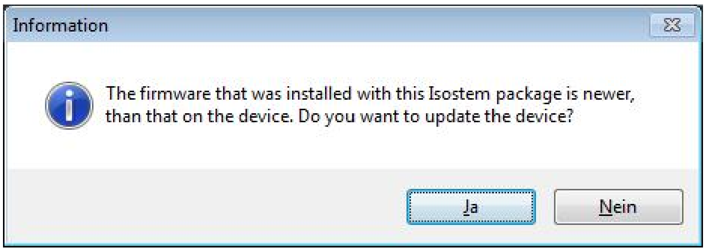


In the GUI window, the "Hardware" tab should be selected. In the "Com" section (top left), use the "Port Name" option to select the COM port that your USB to RS232 adapter emulates. Press the "CONNECT" button. If the button changes to "CONNECTED", the communication between the GUI software and the ISOSTEM® hardware has been established successfully and you should see a message like this in the topmost GUI area:



If no connection has been established, try another COM port ("Port Name"), be sure you have started the GUI variant which suits your hardware variant and press the "CONNECT" button once more.

After a successful connection the GUI may detect an older device firmware. In this case following message comes up:



It advises to update the firmware. Of course you may reject the update.

The update procedure is explained in the Firmware section.

5. Hardware Init & Preset Selection

If your ISOSTEM® unit has been used by others before, it might be a good idea to reset it to factory defaults first.

Please note: Doing so will erase all internal user-specific presets as well as the unit's startup configuration and overwrite it with the factory defaults.

- In the top left-hand section of the GUI, press the "Admin" tab.
- In the top right-hand GUI section, press the "Init" button below "Factory Reset". If sure, confirm both confirmation request windows. Now, your hardware will use the factory default configuration.

Next, load a preset. This is a 2-step process: First, load a preset to the GUI interface. Then, transmit this preset to the hardware. Here's how:

- If not yet selected, press the "Admin" tab in the top left-hand section of the GUI.
- The Read/Write area used for preset management is positioned directly below the tabs. For our example, let's select "Preset 1" behind "from" in the first line. Press the "READ" button. The preset is loaded to the GUI now and all parameters are displayed in the Admin window.

- Select "Current" behind "to" in the second line. Press the "WRITE" button. This transfers the preset settings to the hardware. The transfer is confirmed by a pop-up window. Additionally the front display of the unit shows the active preset number.

6. Listening

- Start your external signal source to play back some stereo audio.
- Switch your surround monitoring system on. Starting with a low volume setting in order to protect your speakers and your ears, set your desired playback level.
- You should hear a 5.1 surround signal now that has been generated by ISOSTEM® using your stereo source.
- Follow the steps above to load preset 3. Compare the results.

Please note: At this point, there is no need to care about the clock synchronization of ISOSTEM®'s AES input. ISOSTEM® uses sampling rate converters (SRC) in all inputs if necessary to obtain proper sync. However, your playback system must be synced to one of the ISOSTEM®'s AES outputs if it does not use SRCs on its own.

- Switch the GUI to the "Expert" tab ("Expert" model only). In the lower middle section of the window, there are Mute and Solo switches for each loudspeaker channel available. Among others, they can be useful for listening to discrete channels. In the "Monitor" section below, switching between "Surround" and "Stereo" alternatively lets you compare the original stereo source and the 5.1 ISOSTEM® output.

Please note: To learn more about the parameters for individual fine tuning of the upmixing process, please refer to the "Expert" Tab chapter.

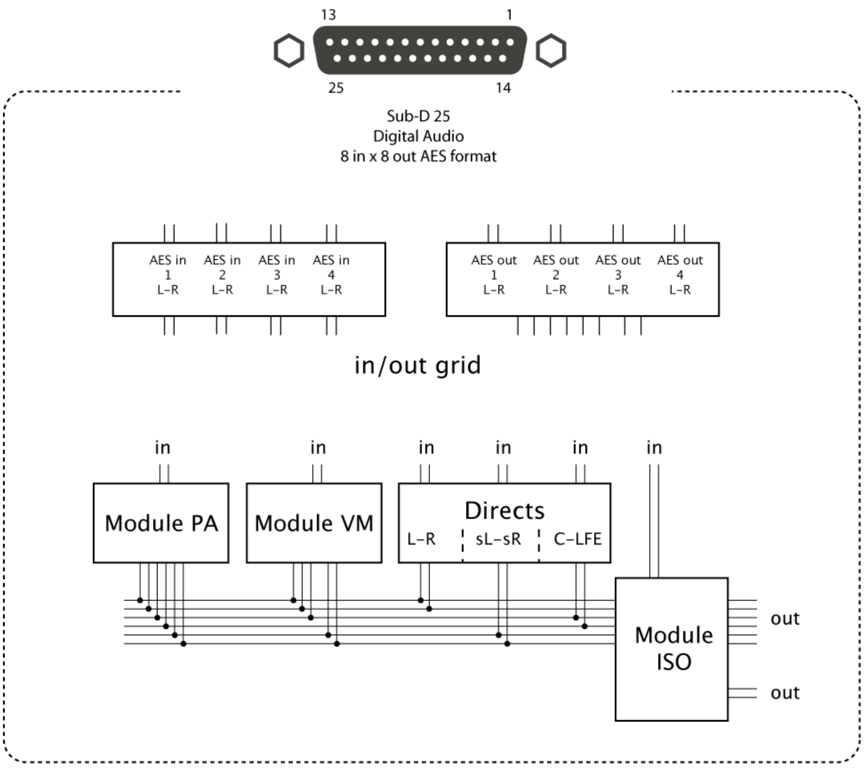
SYSTEM OVERVIEW

DSP TECHNOLOGY

The signal processing is carried out by an energy saving "Sharc" Analog Devices DSP. With just 3W power consumption, no ventilation is needed and the risk of overheating is reduced significantly. The high-performance system handles 32 bit real-time audio processing while the dynamic filtering variables are 64 bit. The conversion modules calculation (up and down) is executed in the frequency domain with phase control.

AUDIO PROCESSING

ISOSTEM® features 4 digital audio inputs in AES3 format (8 channels) as well as 4 digital audio outputs in AES3 format (8 channels). All AES inputs are fitted with sample rate converters (SRC) that can simplify system integration considerably. Flexible input and output matrices enable flexible I/O routing to and from the internal audio processing modules. This allows for compatibility with the various surround channel schemes used in environments such as TV (ITU) or cinema (SMPTE).



The two main audio processing modules "Panoramic Analyzer" (PA) and "Virtual Microphone" (VM) are both independent upmixing stages using different approaches with 2 audio inputs and 6 (5) audio outputs. Together with the "Direct" module, they feed an internal bus that in turn feeds the "ISO" module. The "Direct" module can be fed by native multichannel sources to assure their compatibility with the stereo reference signal, or as part of a setup using several stereo sources.

The "ISO" module calculates the difference between an internal stereo downmix of the upmixed 5.1 surround signals and an external stereo reference signal. Controlling the signal in the frequency domain, ISO takes care of the following additional tasks:

- Assures compatibility by injecting the "intermix" signal (L-R-C-LS-RS)
- Phase correction of the L/R front and rear channels (L-R-LS-RS)
- High-frequency attenuation for the rear channels (LS - RS)
- LFE channel management

HARDWARE

ISOSTEM® is available in two different hardware variants:

ISOSTEM® **Expert** contains a full range of functions and setting parameters. Individual setups can be created using the ISOSTEM® GUI software and saved as presets. Up to 6 presets can be stored locally in the hardware's memory, while the GUI also allows for unlimited preset storage on your computer. In addition, presets can be transferred to any other ISOSTEM®.

ISOSTEM® **Live** was created for daily broadcast use where creating individual presets is not a relevant option. This version can use the factory presets as well as any other presets created on an ISOSTEM® **Expert** unit. However, its limited GUI functionality does not allow for editing presets at parameter level. ISOSTEM® **Live** also features a redundant backup function: If one of two connected devices fails, the system will switch to the other device automatically.

FRONT PANEL

The front panels of both ISOSTEM® variants feature the same set of status LEDs:

- **AES**: This green LED shows the status of the incoming AES signal as well as the unit's sync state. Please refer to the Synchronization chapter for more details about the various sync states.
- **ISO**: Indicates that the "ISO" function is active (see Phase, ISO and Master Functions)
- **Status**: This numerical display shows the number of the currently active preset.
- **Alarm**: Shows the current health status of the hardware device.
- **Process**: Indicates processing or bypass mode.
- **Linked**: Indicates an active link to a second ISOSTEM® device
- **Power**: Indicates that the unit is switched on.
- **Power Switch**: Use this switch to turn the unit on or off.

Rear Panel Connectors

IEC:	Mains power 90..250 VAC
RS232 (D-Sub 9):	Serial interface port for communication with ISOSTEM® GUI software
GPI (D-Sub 15):	This is a binary control input port used to recall the six internally stored presets. External switch to electrical ground initiates preset recall
WC In (BNC):	Word clock input
AES/EBU I/O: (D-Sub 25)	4x AES3 digital audio inputs (8 channels) 4x AES3 digital audio outputs (8 channels) Tascam Pinout
ALARM (Phoenix):	Relay output contacts reflecting the unit's alarm state. Contacts closed: No alarm, device fully functional (Alarm LED off). Contacts open: Device failure (Alarm LED on) or no power supplied
LINK (RJ45):	Proprietary port for cross-connecting two ISOSTEM® LIVE units (backup operation)

For a technical description of the above hardware interfaces (pinouts, etc.), please refer to Appendix A: Interface Specification.

Please note: For more details about system integration using the GPI, Alarm and Link ports, please also refer to the Isostem Live - Integration chapter.

Synchronization

The ISOSTEM® hardware always uses a sample rate of 48 kHz. All Sync parameters are set in the "Sync" section of the "Hardware" tab in the GUI software (see "Hardware" Tab).

In order to suppress any risk of digital clicks, each AES channel has an independent transceiver. In addition, all AES inputs are equipped with sample rate converters (SRC). So, even when syncing to the internal clock, the unit will accept external asynchronous AES signals. However, in order to ensure that ISOSTEM®'s audio outputs are in sync with the studio environment, it might be advisable in most cases to use external synchronization.

The following sync sources are available:

- AES input recovered clock
- External Word Clock
- Internal 48 kHz clock

Please note: If the "AES 1 Input " option was selected as the primary clock source (see Sync in "Hardware" tab chapter), ISOSTEM® will only look for sync on its AES 1 input. In order to sync to an external AES signal, always make sure you use the AES 1 input for this purpose.

The **"AES"** LED on the front panel will reflect the current sync status in the following way:

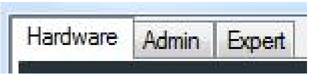
Constant light: ISOSTEM® is in sync with the first clock source selected in the "Master Clock Priority" parameter (see Sync in "Hardware" tab chapter)

Flashing once: ISOSTEM® is in sync with the second clock source selected in the **"**Master Clock Priority" parameter (see Sync in "Hardware" tab chapter)

Flashing twice: ISOSTEM® is synced by its internal clock.

Flashing continuously: No AES signal available at AES Input 1. Configuration should be checked.

GUI Software Reference

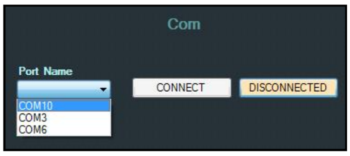


Depending on the hardware variant, the ISOSTEM® GUI software is organized in two or three main windows accessed by tabs in the top left corner. The "Hardware" tab is used for setting up clock and synchronization parameters, firmware updates and system automation, while the "Admin" tab window contains tools for preset management and factory reset.

The "Expert" tab (only available in the ISOSTEM® **Expert** hardware version) features real-time control of all audio processing parameters.

"Hardware" Tab

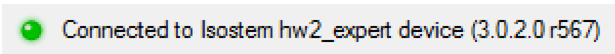
Com



The Com section is used to establish communication between the ISOSTEM® hardware unit and the GUI software.

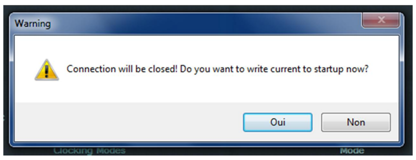
Port Name
Use this option to select your computer's serial COM port for communication. If using the supplied USB to RS232 adapter, its driver software has to be installed first (see Software installation in the Quick Start chapter) and you will have to select the COM port that your adapter emulates.

CONNECT (CONNECTED)
After having selected the COM port, press this button to establish communication. If the button changes to "CONNECTED", the communication between the GUI software and the ISOSTEM® hardware has been established successfully and you should see a message like this in the topmost GUI area:



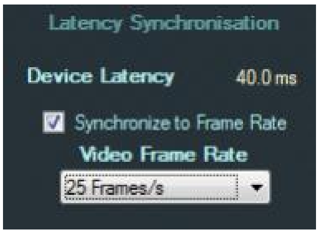
If no connection has been established, try another COM port and press the "CONNECT" button again.

DISCONNECT (DISCONNECTED)
Use this button to disconnect the hardware unit from the GUI software. After doing so, a dialog box will provide the option of writing the current configuration into the "Startup" register (The same dialog will pop up when closing down the GUI software). Doing so ensures that the system will load this configuration during the next startup. This also includes the synchronization and automation settings.



SYNC

Latency Synchronization



The minimum audio processing latency is 40 ms. If working in audio for video environments, it is recommended that you adjust the latency according to the video frame rate used.

Device Latency

This display shows the current audio processing latency.

Synchronize to Frame Rate

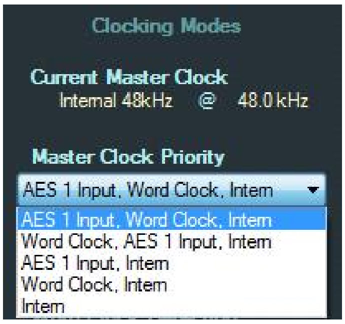
Check this box to increase latency to a value suitable for the video frame rate set below. The increased latency will be reflected in the "Device Latency" display (see above).

Video Frame Rate

This parameter sets the video frame rate that the system latency should be adapted to.

Options: 24, 25, 29.976, or 30 frames/s.

Clocking Modes



Use this section to set up the primary and secondary sync sources to be used for system synchronization. For details, please also refer to Synchronization in the Hardware chapter.

Current Master Clock

This display shows the clock reference currently used by the hardware and its sample rate.

Master Clock Priority

This parameter selects a priority chain of clock sources – i.e. a primary, secondary and fallback clock source. If the attempt to synchronize to the primary source is not successful, the unit will try to sync to the secondary source instead. If this also fails it syncs to the internal clock as a fallback.

Options:

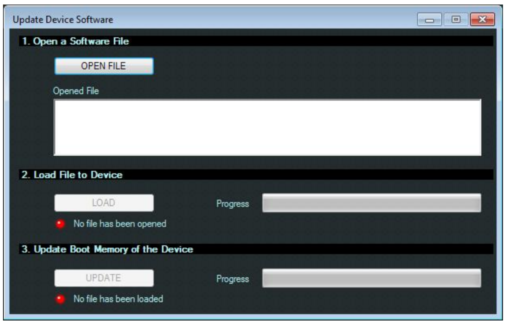
- AES 1 Input (1st), Word Clock (2nd), Intern (fallback)
- Word Clock (1st), AES 1 Input (2nd), Intern (fallback)
- AES 1 Input (1st), Intern (fallback)
- Word Clock (1st), Intern (fallback)
- Intern (fallback)

Please note: In order to sync to an external AES signal, select the appropriate clock source and make sure you use the AES 1 input for this purpose.

Input Data Stream
This display shows any valid audio data streams on the four AES inputs.

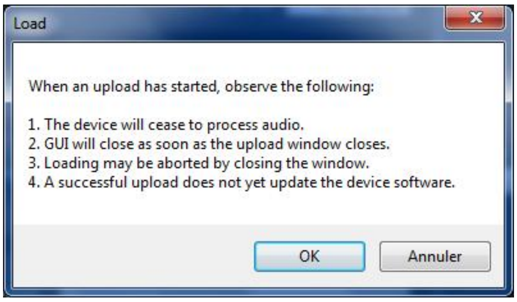
FIRMWARE

UPDATE DEVICE
Use this option to update the system firmware. Please follow the steps described below. After the installation of the GUI the firmware files are located in the installation folder within the "firmware" subfolder.

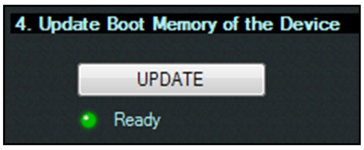


After clicking the UPDATE DEVICE button, the GUI asks if the presets, stored on the device, should be saved onto disk. It is advisable to save them because the update overwrites all presets. After clicking on Yes, the destination folder can be chosen in a dialog. After that the "Update Device Software" window will lead you through the update process consisting of steps 1 – 3.

Step 1 - "Open a Software File": Click the "OPEN FILE" button to select the ".iup"[1] update file stored on your computer. The file path, name and size will be displayed.

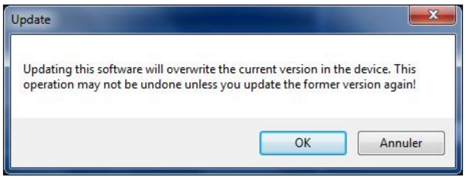


Step 3 - "Load File to Device": Click the "LOAD" button to start transferring the file from the GUI software to the hardware. Before transfer, a dialog box shows some information about the process details. Click "OK" to start the transfer. The progression is shown by a bar.

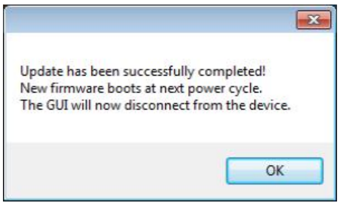


After the transfer is finished, the firmware update is ready to be installed to the DSP. The "UPDATE" button is active.

[1] In the firmware-subfolder the two iup-files „bundled_update.iup" and „Isostem_Update_vXXXX.iup" are located. Both store an identical content – therefore it is irrelevant, which file you select. Both files are linked to the matching iub-firmware files which suit your device variant.

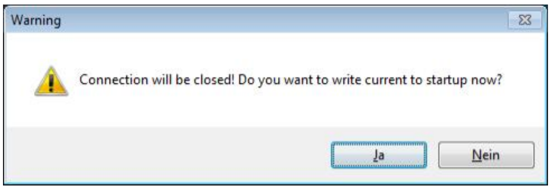


Step 3 - "Update Boot Memory of the Device": The boot memory of the device contains the DSP software. Click the "UPDATE" button that launches the installation of the new internal DSP software. An "Update" dialog box tells you that if you should need to return to the previous version of the internal software, the corresponding ".iup" file will be necessary.



After finished, a dialog box signals the successful update and informs that the new firmware will be active after the next unit´s power cycle.

Before the GUI automatically disconnects from the device, you have the possibility to store the current settings to the start configuration of the unit.



AUTOMATION

The "Automation" section offers several automated operating modes that are selected with the "Mode", “GPI function” and “Channel Automation” options. Among others, these options allow for locking the audio processing parameters to fixed values, external preset switching and automated preset switching triggered by the recognition of certain audio signal types.

Mode

The following mode options are available:

- “**Manual**”: The unit is responding to the GUI software parameters in real time. The “GPI function” is not available.
- “**GPI**”: The unit responds to external GPI commands at the "GPI" Sub-D connector by loading one of the 6 presets stored in its internal memory. Please refer to Appendix A for the GPI connector's pin out scheme.
  Please note: In GPI mode the “Admin” and “Expert” tabs are locked. Additionally the “GPI”/”Manual” mode changes the behavior of the startup configuration (see below “Admin” Tab).

Current Values

This is a status display showing in real-time during a connection to a device the number of the current “Preset Source” and the source of activation which may be the GUI or a GPI switch.

GPI Function

This function allows to configure the action which should take place after the corresponding GPI pin has been activated.

For each of the six GPI inputs you may choose one out of following actions:

- None: No action to be invoked.
- Preset 1…6: The corresponding preset is called.
- Auto 1: Invoke “Parallel” channel automation (see below).
- Auto 2: Invoke “Serial” channel automation (see below).

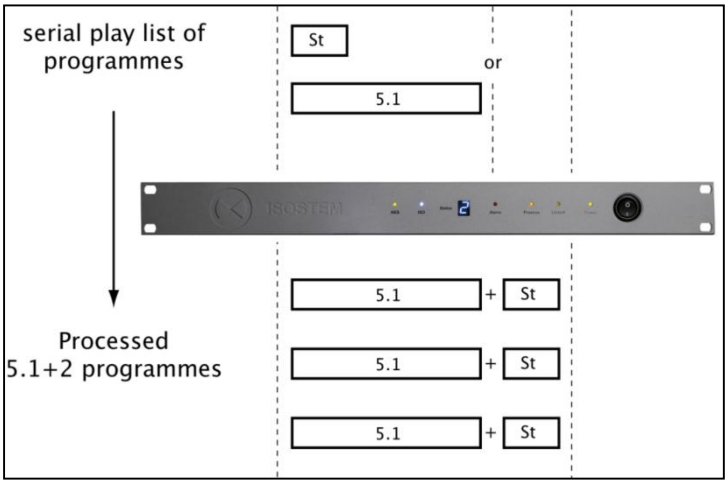
Please note: To use these options the “Mode” has to be set to “GPI”.

Parallel and Serial Channel Automation

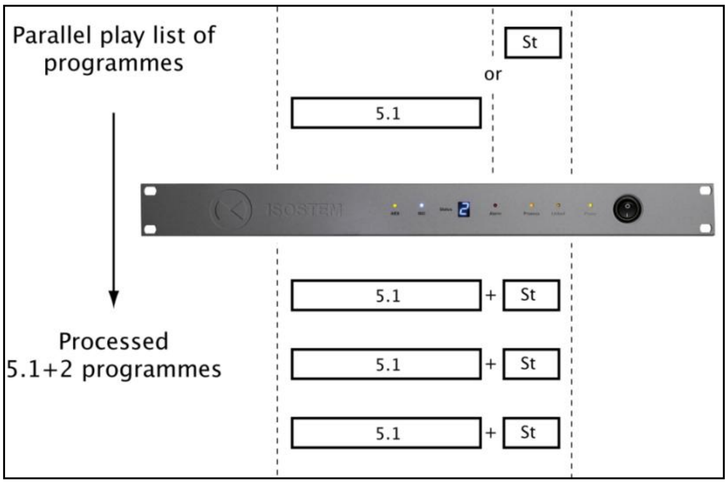
These two modes are dedicated to 5.1+2 broadcasting from various audio program tracks. The idea behind it is to setup a system that will automatically respond to certain signal input configurations (5.1 or stereo) by selecting the appropriate presets. In this way, 5.1 signals may be processed to 5.1 + LtRt while stereo signals are transferred to 5.1 + LoRo stems.

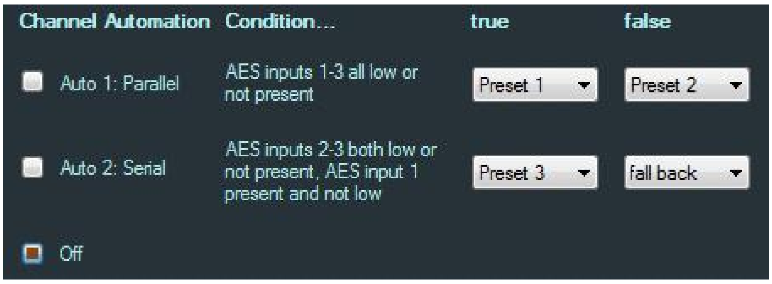
The two available modes address two types of play lists that are presented to the ISOSTEM® inputs:

- The "Serial" mode is used for a playlist containing stereo and 5 .1 tracks on the same channels.



- In contrast to this, the "Parallel" mode is used for a playlist containing stereo and 5.1 tracks on separate channels.





The levels of channels specifically allocated to 5.1 are monitored by the system. If levels pass a certain threshold value, a new preset will be loaded from the internal memory. The transition between the presets is performed smoothly using crossfades.

In “Parallel” mode (“Auto 1”), if the levels on channels L, R, C, LFE, LS and RS on AES inputs 1, 2 and 3 are low, the condition becomes true, otherwise it becomes false.

In “Serial” mode (“Auto 2”), if the levels on channels C, LFE, LS and RS on AES inputs 2 and 3 are low, the condition becomes true, otherwise it becomes false.

You configure the “Parallel” and “Serial” mode of channel automation by activating the corresponding checkbox in the GUI. Checking “Off” disables channel automation. The machine then monitors the corresponding channels as defined in the text below “Condition…” in the GUI and calls the corresponding preset defined in the “true” and “false” column.

“fall back” defines, that the device should re-enter the state it had before it had entered another preset state within the channel automation handling.

Please note: The three checkboxes “Auto 1: Serial”, “Auto 2: Parallel” and “Off” represent also the current status of the device when being connected to the device. This is important to know, because the auto modes also may be invoked by GPI calls when configured in “GPI” mode (see above).

"ADMIN" TAB

The Admin tab is used for administration tasks such as loading and saving presets in the hardware memory or on the computer's hard drive. These tasks are performed in the upper section of the window.

The remaining window space is used for a detailed display that shows all parameters of the currently loaded configuration at a glance.

READ/WRITE

This preset management section is used to load presets into the GUI software and to write them to a target such as the hardware's internal memory or a file on the computer's hard drive.

Loading a new preset to work with is a 2-step process: First, load a preset to the GUI interface. Then, transmit this preset to the hardware.

To load a preset into the GUI software:

- Select a source in the field behind "from" in the first line. The following options are available:

  Single Preset File – loads single preset from a file stored on your computer
  All Presets File – loads a preset file storing all six presets from a file stored on your computer (for a write action to “All Presets”)
  Startup – loads the preset configured for system startup
  Current – loads the configuration currently active in the hardware
  Preset 1-6 – loads one of the presets stored in the hardware
  All Presets – loads all six presets stored in the hardware to a file-buffer (for a write action to a “All Presets File”)

- Press the READ button to load the selected preset source. The preset is loaded to the GUI now and all parameters are displayed in the lower part of the Admin window.
  Please note: When reading “All Presets File” or “All Presets” the parameter display does not change, because these presets are read to a hidden file-buffer.

To write a configuration to a target:

- Select a target in the field behind "to" in the second line. The following options are available:

  Single Preset File – save the displayed settings in a single preset to a file on your computer
  All Presets File – save all presets currently saved in the file-buffer (after having read “All Presets”) to a file on your computer
  Startup – save the currently displayed configuration for system startup
  Current – transmit the currently displayed configuration to the hardware, where it becomes active as a volatile setting not stored in the preset memory
  Preset 1-6 – save the currently displayed configuration to one of the hardware presets slots
  All Presets – save all presets currently saved in the file-buffer (after having read a “All Presets File”) to all hardware presets slots

- Press the "WRITE" button. The current GUI configuration is transferred to the selected target. The transfer is confirmed by a pop-up window.

Please note: The automation mode (“GPI”/”Manual” – see above) changes the behavior of the startup configuration. In “GPI” mode the “Startup” preset is overwritten by the last called preset – i.e. the device starts with the formerly active preset after a power-cycle. In “Manual” mode the “Startup” preset is not changed automatically. You can define it in the “Admin tab” (see above).

File Backup

Using the "File" selection to be the target of a write process, presets can easily be transferred to a computer. The files created in this way carry the "*.ips" extension. This is an easy way of archiving presets or transferring them from one unit to the next.

Please note: Preset files do not contain hardware configuration settings such as the latency synchronization or the preferred sync source. Such parameters usually refer to the global system environment rather than specific audio processing tasks.

Loading a File
After selecting "File" in the "from" menu (first line), the standard file opening dialog box of your computer's operation system lets you select an *.ips file to be loaded into the GUI.

Please refer to Appendix C: Factory Presets for a brief overview of all factory presets integrated in the device. On the supplied ISOSTEM® installation disc, these presets are also available as a backup.

Write Protection

This section is used to protect the six presets stored in the hardware as well as the startup preset from being overwritten. Just check the boxes of the preset(s) you want to protect.

Factory Reset

This function is used to reset the hardware unit to factory defaults.

Please note: Doing so will erase all internal user-specific presets as well as the unit's startup configuration and overwrite it with the factory defaults.

Press the "Init" button below "Factory Reset". If sure, confirm both confirmation request windows. Your hardware will now use the factory default configuration.

"EXPERT" TAB (EXPERT HARDWARE VERSION ONLY)

The Expert tab is used to edit the unit's audio processing parameters in real time. All parameter settings on this page are directly reflected by the audio processing.

The Expert tab is divided into five sections:

1 – "Panoramic Analyzer" (PA)

2 – "Virtual Microphone" (VM)

3 – Multifunctional area with 9 sub-windows

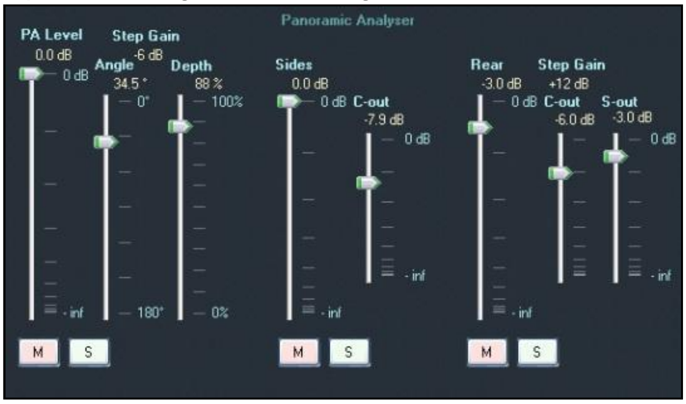
4 – Monitoring and Bypass functions

5 – Phase, ISO and Master functions

PANORAMIC ANALYZER (PA)

Functionality

The Panoramic Analyzer distinguishes the parts of a signal containing mainly diffuse or direct sound by identifying the spatial energy distribution. These signal elements are isolated by dynamic filtering and can then be distributed to the surround channels following aesthetic considerations. It is important to point out the fact that no additional signal elements are created by this process; in fact, the existing elements are only distributed in a new way.



Please note: While editing the Panoramic Analyzer's parameters, it is a good idea to watch its graphic representation in the "PA" tab (see Multifunctional Area).

PA Level
This fader adjusts the direct output levels for all six discrete output signals of the Panoramic Analyzer (PA). However, it does **not** affect the signals that the "Side level" and "Rear Level" faders contribute to the output busses.

Unit: dB
Range: 0 dB - -inf

Please note: On the "Bus" tab of the multifunctional area, the individual levels of the L/R, C, Sub and sL/sR channels can be adjusted separately.

PA Level Mute/Solo
These buttons are used to mute or solo the Panoramic Analyzer's discrete output signals.

Angle
This fader adjusts the opening angle of the stereo signal. Its position is reflected in the "PA" tab of the multifunctional area by two vertical grey lines (see Multifunctional Area).

Unit: Degrees
Range: 0° - 180°

Depth
This fader controls the depth of the dynamic filtering performed by the Panoramic Analyzer.

Unit: Percent
Range: 100% - 0%

Please note: Both Angle and Depth parameters are the crucial basis for the Panoramic Analyzer's ability to separate direct and diffuse signal components from each other. They should be set carefully in order to fit the source material and user's individual application in the best possible way.

PA Level Step Gain
This read-only display shows the Panoramic Analyzer's input attenuation selected in the multifunctional area's "Misc" tab (see Multifunctional Area).

Unit: dB

Sides
This fader controls the level used to inject a mix of the stereo input signal and the "Side C-out" level (see below) into the output busses L and R.

This parameter is a powerful tool to control the selectivity used to separate direct and diffuse signal components in the L and R front channels. By subtracting the separated signals from the original stereo signal, the spatial complements of the dry sources can be added to the L and R front channels.

Unit: dB
Range: 0 dB - -inf

Sides Mute/Solo
These buttons are used to mute or solo the Sides fader signal.

Sides C-out
This fader controls the level used to inject the phase-reverted discrete Center signal output of the Panoramic Analyzer into the output busses L and R.

This parameter is used to fine-tune the presence of the phantom source between the L and R front channels. Higher fader positions increase the removal of Center signal components from the L and R front channels.

Please note: This function depends on the position of the "Sides" fader (see above). It is also controlled by the Sides Mute and Solo buttons.

Unit: dB
Range: 0 dB - -inf

Rear
This fader controls the level used to inject a mix of the stereo input signal, the "Rear C-out" level and the "Rear S-out" level (see below) into the output busses LS and RS.

This parameter is used to control the balance between direct and diffuse signal components in the LS and RS rear channels. Higher fader positions increase the spatial signal components.

Unit: dB
Range: 0 dB - -inf

Rear Mute/Solo
These buttons are used to mute or solo the Rear fader signal.

Rear Step Gain
This read-only display shows the gain setting for the rear channels selected in the multifunctional area's "Misc" tab (see Multifunctional Area).

Unit: dB

Rear C-out
This fader controls the level used to inject the phase-reverted discrete Center signal output of the Panoramic Analyzer into the output busses LS and RS.

This parameter is used to fine-tune the presence of the phantom source between the LS and RS rear channels. Higher fader positions increase the removal of Center signal components from the LS and RS rear channels.

Please note: This function depends on the position of the "Rear" fader (see above). It is also controlled by the Rear Mute and Solo buttons.

Unit: dB
Range: 0 dB - -inf

Rear S-out
This fader controls the level used to inject the phase-reverted discrete L and R front signal outputs of the Panoramic Analyzer into the output busses LS and RS.

This parameter is used to fine-tune the diffuse signal components of the front channels L and R in relation to the direct sound in the LS and RS rear channels.

Unit: dB
Range: 0 dB - -inf

Please note: This function depends on the position of the "Rear" fader (see above). It is also controlled by the Rear Mute and Solo buttons.

Please note: For in-depth signal flow details, please also refer to the block diagram in Appendix B.

VIRTUAL MICROPHONE (VM)

Functionality

The Virtual Microphone is another tool for surround sound creation that can be used independently or in combination with the Panoramic Analyzer. Based on an acoustic model, the stereo source information is translated to a virtual 5.0 microphone array – you may think of it as if a 5.0 microphone setup would be placed in front of a stereo speaker system. Cardioid microphone capsules are modeled with adjustable spacing and directivity curves. The signals feed each of the output busses directly. The VM module does not deliver a Sub (LFE) channel signal. It may be provided by the Sub module instead (see Multifunctional area).

Please note: While editing the Virtual Microphone's parameters, it is a good idea to watch its graphic representation in the "VM" tab of the multifunctional area.

VM Level
This fader adjusts the direct output levels for all five discrete output signals of the Virtual Microphone (VM).

Unit: dB
Range: 0 dB - -inf

Please note: On the "Bus" tab of the multifunctional area, the individual levels of the L/R, C, and sL/sR channels can be adjusted separately.

VM Mute/Solo
These buttons are used to mute or solo the VM fader signal.

Angle
This fader adjusts the opening angle of the stereo signal. Its position is reflected in the "VM" tab of the multifunctional area by two grey lines (see Multifunctional Area).

Unit: Degrees
Range: 0° - 180°

Lobes
This parameter defines the polar pattern of all five virtual microphone capsules used in the VM module. A value of 0% represents omnidirectional patterns, the maximum value 100% stands for figure-of-eights.

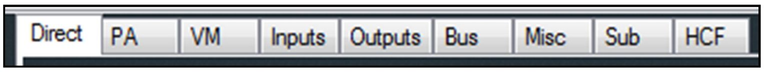
Unit: Percent
Range: 100% - 0%

Gap
This parameter defines the center distance of all five virtual microphones.

Unit: cm
Range: 72 cm – 0 cm

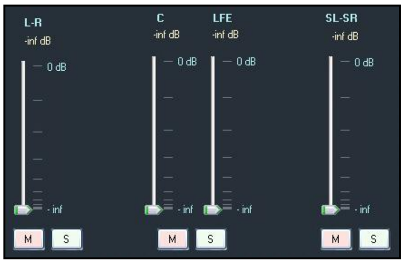
MULTIFUNCTIONAL AREA

The multifunctional area in the top right-hand section of the Expert tab features nine view modes that are selected using sub-tabs. Among others, they can be used to visualize parameter edits in audio processing modules such as the Panoramic Analyzer or the Virtual Microphone.



Direct Window

The direct module connects any multichannel sources present at the audio inputs directly to the internal bus. This module is particularly important for processing native multichannel programs.



L-R
This fader adjusts the direct level of the L and R front channels.

Unit: dB
Range: 0 dB - -inf

L-R Mute/Solo
These buttons are used to mute or solo the L-R fader signal.

C
This fader adjusts the direct level of the Center channel.

Unit: dB
Range: 0 dB - -inf

LFE
This fader adjusts the direct level of the LFE channel.

Unit: dB
Range: 0 dB - -inf

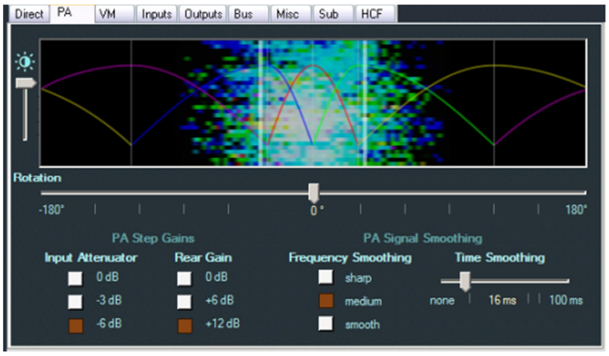
C/LFE Mute/Solo
These buttons are used to mute or solo the Center and LFE fader signals.

SL-SR
This fader adjusts the direct level of the SL and SR rear channels.

Unit: dB
Range: 0 dB - -inf

SL-SR Mute/Solo
These buttons are used to mute or solo the SL-SR fader signal.

PA Window



The PA window features a visualization of the Panoramic Analyzer's dynamic filtering with overlapping filter bands. The angle set in the PA parameter section is represented by two light grey vertical lines. A vertical spectrogram of the incoming stereo source is shown between these lines.

Rotation

This parameter sets the axial rotation of the PA module. Among other things, it can be helpful in order to correct an imbalanced stereo picture. The effect of editing this parameter can be monitored in the visualization.

Unit: Degrees
Range: -180° / +180°

Brightness

This parameter sets the spectrogram's brightness.

PA Step Gains – Input Attenuator

The PA input attenuator avoids any risk of C channel saturation during an up-conversion. If a stereo signal is made from an identical right-left signal with its peak level normalized (0 dBFS), the total quantity of energy cannot be held in only one C channel without overloading. The PA input attenuator prevents this risk.

Options: 0 dB, -3 dB, -6 dB

PA Step Gains – Rear Gain

The gain on the rear PA channels applies to the "Rear" complement channels. This additional gain aims to reinforce the envelopment and to highlight the presence of the multichannel format.

Options: 0 dB, +6 dB, +12 dB

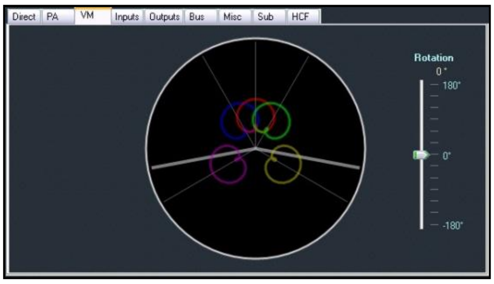
PA Signal Smoothing – Frequency / Time Smoothing

The temporal and frequency smoothing apply to the spectral envelopes of the PA module filters. The temporal smoothing is a weighted averaging of the envelopes in time, while the frequency smoothing is a weighted averaging of the envelopes in frequency.

Frequency smoothing options : sharp, medium, smooth
Time smoothing options: none – 100 ms

VM Window

The VM window features a graphic representation of the virtual 5.0 microphone setup used by the VM module. The angle set in the VM parameter section is represented by two light grey lines; the form of the virtual microphones show the polar pattern set with the "Lobes" parameter.



Rotation

This parameter sets the axial rotation of the VM module. Among other things, it can be helpful in order to correct an imbalanced stereo picture. The effect of editing this parameter can be monitored in the visualization.

Unit: Degrees
Range: -180° / +180°

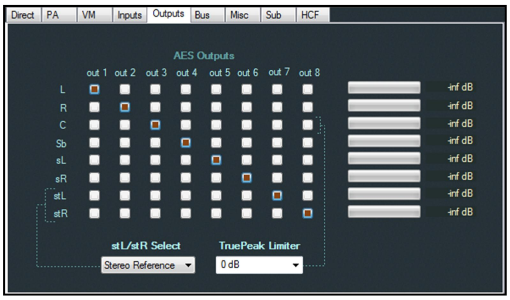
Inputs Window

The Inputs window features an input matrix covering all signal processing modules and all four AES3 inputs. One of the signal inputs can be assigned to each of the modules. Also, horizontal bar graph level meters are available for all module input channels.

Please note: L and R channels of each AES3 input are always switched in common.

Outputs Window

The Outputs window features a matrix covering all eight outputs of the internal bus system and all AES3 outputs. Each physical output can be assigned to any one of the bus outputs individually. However, it is not possible to assign one bus output to more than one physical output or vice versa. horizontal bar graph level meters are available for all bus output channels.



Please note: L and R channels of each AES3 output can be switched independently.

stL/stR Select

"sL/stR Select" defines the source of the "+2" in the "5.1 + 2" format. This "+2" stereo output channels stL/stR alternatively may be Stereo Reference Input, LtRt, LoRo or 7.1 (being a -6 dB version of Ls-Rs channels).

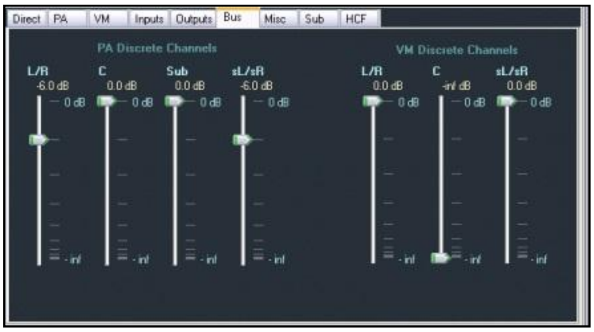
Options: Stereo Reference, 7.1, LtRt, LoRo

Please note: The 7.1 mode does not correspond to the channel format of the same name, which includes five front channels (L, R, C, CL and CR) with two rear channels.

TruePeak Limiter
The function „TruePeak Limiter“ activates and configures a limiter at the output of the center channel to avoid a signal overdrive, which may occur on the center channel in extreme situations by the upmix. Internally the TruePeak Limiter works with a 4 times oversampling to capture the true signal amplitudes.

Options: off, 0dB, -1dB, -2dB, -3dB

Bus Window
The Bus window features discrete channel faders for the outputs of the PA and VM modules used for upmixing. These level adjustments can be useful to model the shape of a conversion: less diverged (strong center), or more square (quadrophonic).



Available faders for the PA module:

- L/R
- C
- Sub (LFE)
- sL/sR (RS/LS)

Available faders for the VM module:

- L/R
- C
- sL/sR (RS/LS)

Misc Window

The Misc Window features several special signal processing functions such as attenuation settings or signal smoothing options.

Downmix Attenuation – C to L-R, LFE to L-R, sL-sR to L-R

The "Downmix" parameters for C, LFE and sL-sR determine the coefficients to be applied to the respective channels for the downmix to stereo.

Downmix attenuation options (all):
0 dB, -1,5 dB, -3 dB, -6 dB, -inf.

Loudness Alignment Mode

This function guarantees an identic EBU R.128 Programme Loudness of the upmix and downmix results in relation to the source material.

Please note: After activation of this feature, the ISO function is minimally limited in its effect, because the loudness alignment carries out a level correction at the end of the signal chain.

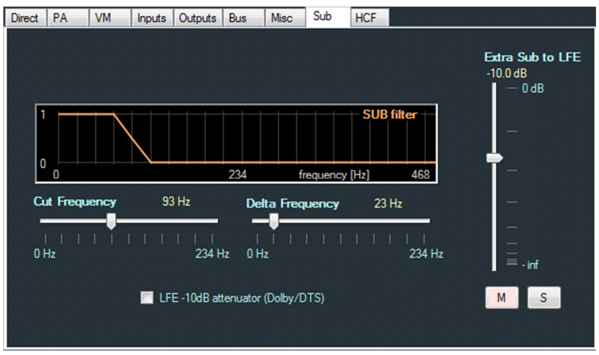
Options for Loudness Alignment: Off, Align Upmix (to StRef), Align Downmix (to 5.1)

Disable ISO if Module Muted

The "Disable ISO" function deactivates the ISO function when "Mute" or "Solo" has been selected in one of the modules. Mute and Solo do not work efficiently as long as the intermix signal is reinjected into the multichannel stem. Some users might get confused with "ISO" reinjection over an engaged Mute or Solo.

Sub Window

The Sub window features a Bass management option for Sub (LFE) channel filter adjustment. This filter acts on the spectral envelope with a gain in amplitude and constant phase. It applies to the Panoramic Analyzer module and the reference stereo signal. Additionally a -10 dB LFE Dolby/DTS attenuator is available.



Cut Frequency

Cut frequency of filter.

Unit: Hz
Range: 0 Hz .. 234 Hz in steps of 23 Hz

Delta Frequency
Half width of the transition region.

Unit: Hz
Range: 0 Hz .. 234 Hz in steps of 23 Hz

LFE -10dB attenuator (Dolby/DTS)
Activates a 10 dB attenuation at the LFE output.

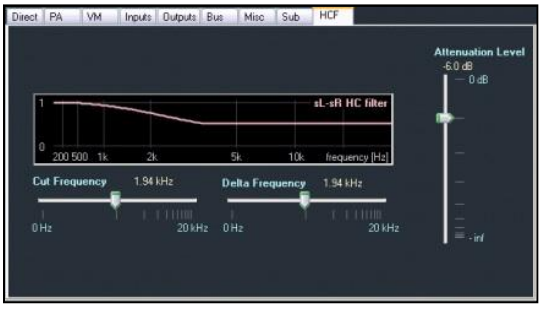
Extra Sub to LFE
The fader „Extra Sub to LFE“ controls the amount of additional bass, which is extracted from the stereo reference signal and which is mixed to the LFE channel of the output bus.

Unit: dB
Range: 0 dB..-inf

Please note: See Appendix B: Schematic Block Diagram.

Extra Sub to LFE Mute & Fade
Mute deactivates the extra bass. Solo mutes all other mute / solo controls. This is useful to monitor the complete signal without the extra bass.

HCF Window
The HCF window features a high frequency attenuator on the rear channels that allows the user to polarize the image and reinforce the attraction of the front channels. Just like "SUB", this filter acts on the spectral envelope with a gain in amplitude and constant phase. This filter applies to the internal bus at the end of the Panoramic Analyzer and Virtual Microphone.



Options: Cut Frequency (0 Hz – 20 kHz)
Delta Frequency (0 Hz – 20 kHz)
Attenuation level (0 dB - -inf)

MONITORING AND BYPASS FUNCTIONS

This section features solo and mute buttons for each channel as well as a monitor source selector giving access to various signals. Also, certain processing elements can be bypassed with the "Functions" buttons. Please note that the monitoring applies directly to the L, R, C, Sub, sL and sR channels of the output matrix.

Channel Solos
Mute and solo buttons are available for each output channel.

Monitor – SURROUND
In SURROUND monitor mode, the ISOSTEM® surround outputs are monitored through the multichannel outputs.

Monitor – STEREO
In STEREO monitor mode, the original stereo input signal is monitored through the multichannel outputs (L/R).

Monitor – DOWNMIX
In DOWNMIX monitor mode, the downmixed surround outputs are monitored through the multichannel outputs (L/R).

Monitor – INTERMIX
In INTERMIX monitor mode, the differential signal between the downmixed surround outputs and the original stereo input signal is monitored through the multichannel outputs (L/R).

Functions – ISO
The ISO button is used to toggle the ISO processing module between active and bypass mode. The ISO function can be activated to ensure that a stereo downmix of the ISOSTEM®'s surround output signals is perfectly compatible with the original stereo source. This is done by creating the difference signal between downmix and stereo signal. This "intermix" signal is then injected into the surround model in a way that downmix and stereo signal are identical. The ISO function can be easily disabled for applications not enforcing perfect compatibility between the two. The ISO function uses the downmix parameters which are configured in the Misc window.

Please note: An activated Loudness Alignment function guarantees an identical Programme Loudness of the resulting downmix and the stereo.

Functions – PHA
The PHA button is used to toggle the PHA processing module between active and bypass mode. For more details on the PHA module, please refer to the following chapter.

Functions – HCF

The HCF button is used to toggle the HCF processing module between active and bypass mode. The HCF function is an attenuation of the high frequencies applied to the rear channels on the internal bus at the output of the PA and VM modules. The filter adjustment parameters can be found in the "HCF" tab of the multifunctional area.

Functions – SUB

The SUB button is used to toggle the SUB processing module between active and bypass mode. The SUB function treats the Sub (LFE) channel of the internal bus at the output of the PA and VM modules. It determines the multichannel signal format issued (5.0 or 5.1). When the "SUB" function is activated, the Sub output channel is active. Otherwise, the Sub channel is assigned to the L and R channels. The adjustment parameters can be found in the "Sub" tab of the multifunctional area.

Phase, ISO and Master functions

Phase
The "Phase" function shifts the phase of lateral and rear channels (L, R, sL and sR) because up-converting introduces a negative tendency in L-R and SL-sR correlation. This phenomenon is due to the fact that the correlated part of the original stereo signal meets in the C channel.

The "Phase" function prevents this aspect by shifting the phase of the L-R and sL- sR signals through only one variable. The variable is calibrated from zero to one hundred percent and applies a phase shift modeled on the physical angled distances of the loudspeakers.

Unit: Percent
Range: 100 % - 0 %

Iso Bal
The ISO balance function assures compatibility between the stereo and multichannel formats. The difference between downmix and original stereo signal is called "intermix". This intermix is inserted in the multichannel signal so that the downmix is mathematically identical to the stereo signal.

In order to be able to better integrate the intermix within the multichannel stem, the ISO function separates the elements M (half sum) and S (half difference) of the "Intermix". Element M is split between channels C and L-R. Element S is split between the L-R pair and the SL-SR pair.

M
Unit: Percent
Range: Center – L/R

S
Unit: Percent
Range: L/R – sL/sR

Master Level
This fader is used to attenuate the master output level of all channels.

Unit: dB
Range: 0 dB - -inf

ISOSTEM® Live - Integration

Scope

ISOSTEM® Live is intended for integration into professional studio and broadcast environments in three different ways:

- Single device
- Two daisy-chained devices
- Two devices in parallel

In single device operation, no functional redundancy is given. In case of a failure, each ISOSTEM® Live has the means for non-disruptive quadruple AES3 bypassing, though, of course, it will no longer be able to process the audio streams.

Two devices may be integrated daisy-chained or in parallel. Both setups offer functional redundancy. Though a daisy-chained integration has no need for further failure management devices, it offers no non-disruptive device exchange in case of a required service procedure.

For 24/7-operation, a parallel ISOSTEM® Live integration is recommended. Failed devices may be exchanged non-disruptively. This setup requires an external AES3 audio routing device.

This section enables users to integrate ISOSTEM® Live into studio and broadcast systems in each way mentioned above. In addition, information for preset and parameter synchronization in a redundant setup is given.

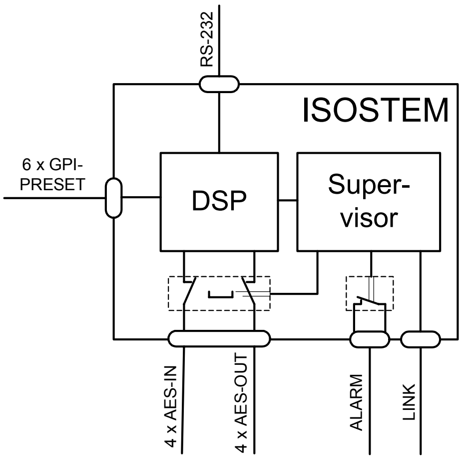
Single Device Audio Routing and Alarm Function

ISOSTEM® Live is equipped with an internal hard-wired supervisor which detects hardware, software and power supply failures. A detected failure results in the following immediate action:

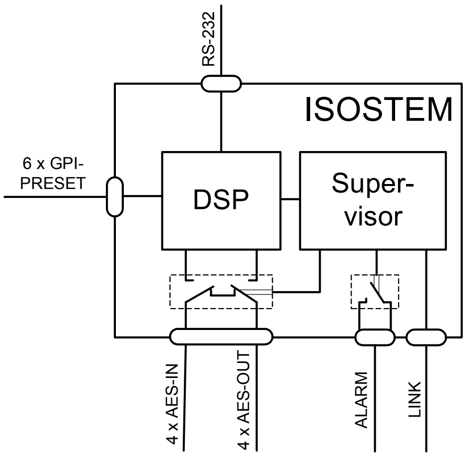
- The audio routing is switched from processing to bypass, i.e. the four AES-Input signals are electrically switched to the four AES-outputs without DSP interference. This is the default power-off state.

- The alarm relay output opens its contacts to signal the failure electrically. This is the default power-off state.

Single device setup and alarm/processing interaction:



OK status - no alarm (contacts closed)
Processing: AES-In → DSP → AES-Out



FAILURE status – alarm active (contacts open)
No Processing = Bypass: AES-In → AES-Out

In a single device setup, the link port is unused.

Daisy-Chained Audio Routing and System States

Two ISOSTEM® Live are intended to operate as a redundant backup system. A daisy-chained topology is the basic redundant setup (see schematic below).

It must be noted that the daisy-chained setup may not apply to systems intended to operate 24 hours and 7 days a week. This is due to the fact that, in cases of failure, the audio chain is disconnected while exchanging the failed device. 24/7-systems are handled by a parallel ISOSTEM® Live audio setup (see next chapter).

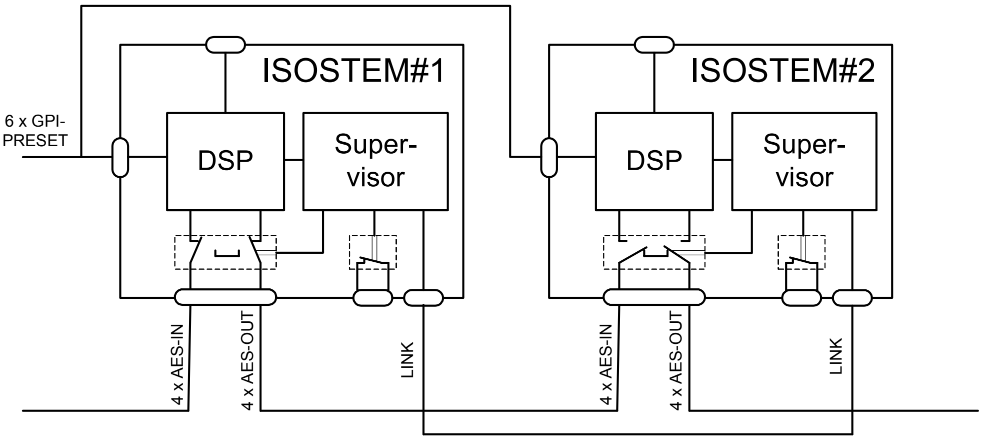
The four AES-In signals from the audio uplink are wired to ISOSTEM® #1, its four AES-Out signals are wired to the next ISOSTEM® (#2) and its four AES-Out signals are wired to the system´s downlink device.

A CAT5-crosslink interconnection of the two link ports is required for proper audio routing. If connected to both link ports, each ISOSTEM® ´s supervisor handles this as a linked-condition.

Thus one of the two devices is forced to the processing state and the other to the bypass state. The bypassing ISOSTEM® is the backup device.

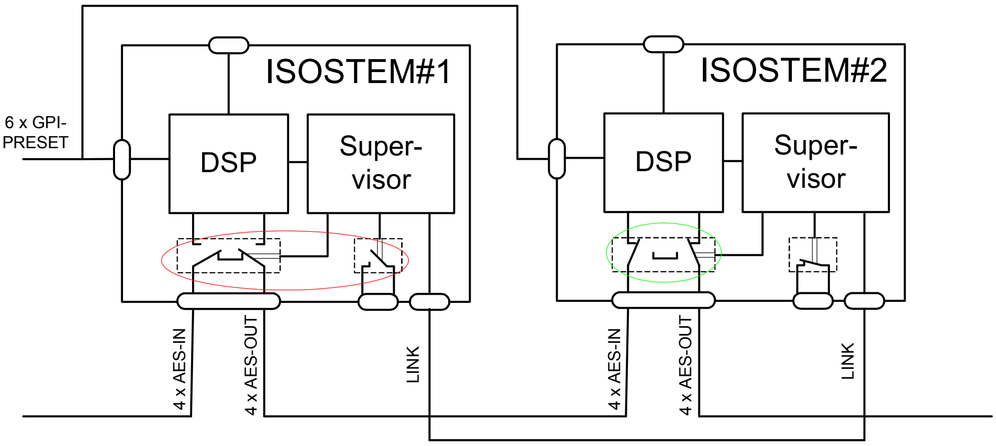
The user defines the initial processing-bypass state of both devices during power-up. When linked, the device powered up as the first is automatically the one in processing state. The second device power-up is automatically forced to bypass audio.

Daisy-Chained Setup:



Example: ISOSTEM #1 processes, ISOSTEM #2 bypasses, no alarm

The alarm state overrides this situation: An alarm always forces the bypass state. The block diagram below presents an example of an alarm situation for ISOSTEM® #1.



Example showing ISOSTEM #1 in alarm state
(ISOSTEM #1 bypasses, ISOSTEM #2 processes)

All possible situations in linked mode are:

No alarm: One device processes (in our example ISOSTEM® #1), the other is the backup – it bypasses audio.

ISOSTEM® #1 with alarm, ISOSTEM® #2 no alarm: The backup ISOSTEM® #2 takes over and processes, whereas ISOSTEM® #1 bypasses and signals its failure state by LED and alarm relay.

ISOSTEM® #2 with alarm (the backup device), ISOSTEM® #1 no alarm: The backup-device ISOSTEM® #2 cannot provide backup any more but of course stays in bypass. It signals its failure state by LED and alarm relay.

ISOSTEM® #1 and ISOSTEM® #2 with alarm: Both devices are forced to bypass audio – i.e. the audio chain is not disconnected, but no ISO-processing is made. Both devices signal their failure state by LED and alarm relay.

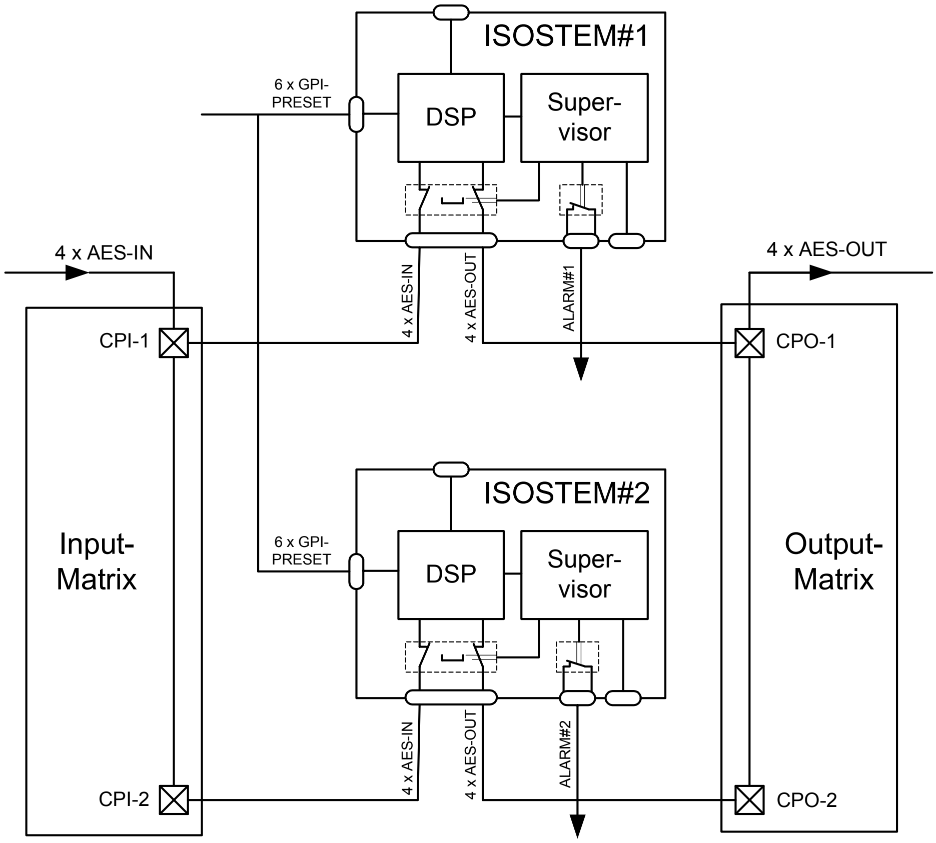
PARALLEL SETUP FOR 24/7-OPERATION

The diagram shown below presents the parallel audio setup of an ISOSTEM® Live pair. An external audio matrix with following capability is required:

- Two quadruple AES-input cross points (CPI-1, CPI-2),

- two quadruple AES-output cross points (CPO-1, CPO-2), and

- two binary control inputs handling the alarm outputs of the ISOSTEM® devices with the ability of conditional processing to control the AES-crosspoints (CPI-1/2 and CPO-1/2).



Parallel audio and alarm setup of an ISOSTEM Live pair for 24/7-operation

For proper parallel operation, the link interconnection is not mandatory. Basically, both devices run as single devices.

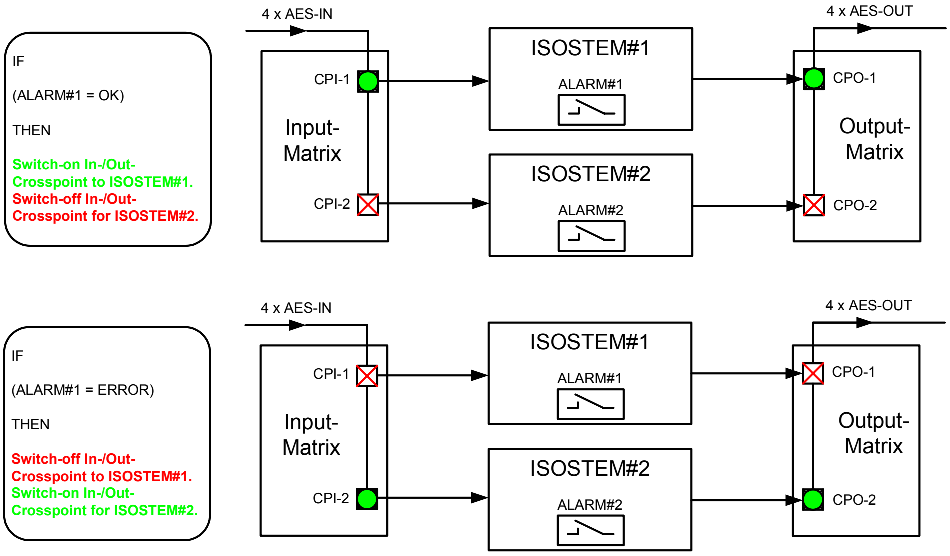
The external audio matrix has to process the devices´ alarm relay states (ALARM#1, ALARM#2) and has to ensure that in all situations, the AES inputs and AES outputs of each device are separated electrically.

The schematic diagram shown below presents an example of the matrix´s conditional processing by merely handling the alarm relay output of ISOSTEM® #1:

If ALARM#1 (independently of the alarm state of ISOSTEM® #2) represents a fully functional state, the audio matrix routes the AES data only to ISOSTEM® #1.

If ALARM#1 represents a failure, switch-over to the backup ISOSTEM® #2 is effected. ISOSTEM® #1 may now be exchanged without interrupting the audio line. It must be noted that, after exchanging the formerly failed ISOSTEM® #1, the audio matrix will switch back to ISOSTEM® #1 because it will signal ALARM#1 = OK.

In this example, in the case that both devices fail, ISOSTEM® #2 receives the audio data. But as it has a failure, it just bypasses the AES-data from input to output.



Example of conditional processing for proper audio routing

Preset Recall Synchronization

In linked redundant mode, both GPI ports have to be interconnected in parallel as shown in the schematic diagram above. Thus preset recall actions are processed by both devices synchronously. This ensures that the back-up device always runs in the same mode as the current processing device.

Preset Management Synchronization

ISOSTEM® Live´s PC-based graphic user interface (GUI) allows for management of the six preset memories. Presets can be read from the PC and stored in the device´s preset memory.

In redundant setups, the user has to synchronize the preset content and memory assignment manually. Future GUI and firmware revisions will offer automatic synchronization of parameters and preset memories via the link port.

APPENDIX

APPENDIX A: INTERFACE SPECIFICATION

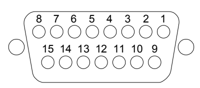
GPI PORT

The GPI port of ISOSTEM® features an easy way to recall six parameter presets remotely. A preset is triggered by pulling the dedicated preset pin to the device´s electrical ground (GND). The GPI inputs are level-sensed. Thus, a momentary push-button has to be pressed (to close its contacts) for a very brief period of time (<1sec.).

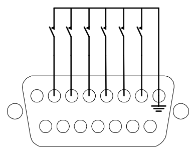
The preset is called during state transition from open to closed, i.e. GND. Thus, a momentary push-button switch is capable of calling a preset.

Function	Pin#
Call Preset #1	7
Call Preset #2	6
Call Preset #3	5
Call Preset #4	4
Call Preset #5	3
Call Preset #6	2
GND	1, 8, 9
reserved	10..15

Pin and function assignment of GPI port



View of female SUB-D 15-pin GPI interface on rear side



Schematic diagram of push-button connection

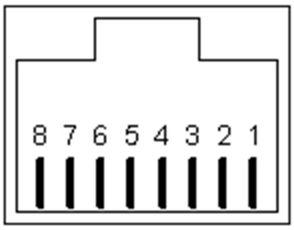
LINK PORT

The link port is used to connect two ISOSTEM® Live units in redundant mode.

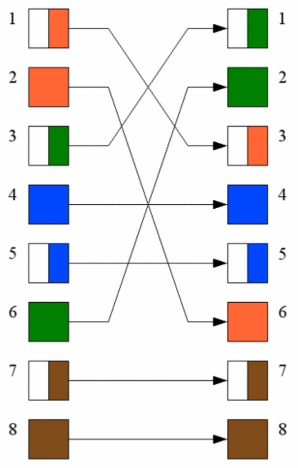
In a daisy-chained setup, a standard CAT5 cross-link cable interconnection is mandatory for automatic processing/bypass-handshake and backup-switchover.

In a redundant parallel setup, a device interconnection is not mandatory.

Future software extensions will enable the link port´s integrated data interface for automatic parameter/preset synchronization.



View of RJ45 link port from rear side of device

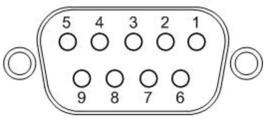


Required CAT5 cross-link pin assignment

Pin assignment of Link Port and specification of cross-link cable

RS232

The RS232 interface enables a PC connection for running the GUI software.



View of female SUB-D 9-pol. RS232 interface on rear side

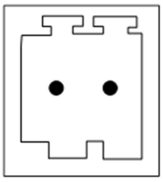
Function	Direction	Pin#
TXD	Out	2
RXD	In	3
CTS	In	7
RTS	Out	8
GND		5
SHIELD		CASE

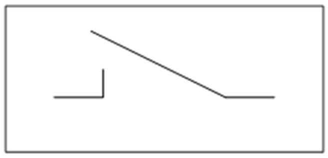
ALARM PORT

The alarm port indicates the device status electrically. It is a 2-pin 2.5 mm PHOENIX-male connector. Internally, the two pins are connected to relay contacts – thus it is a fully passive interface with two states:

Contacts closed: No alarm, device fully functional.

Contacts open: Device failure or no power supplied.





Max. load to relay contacts: 2 mA/250 VAC

WORD CLOCK INPUT

1 x BNC max. 2..5Vp-p

AES/EBU INTERFACE

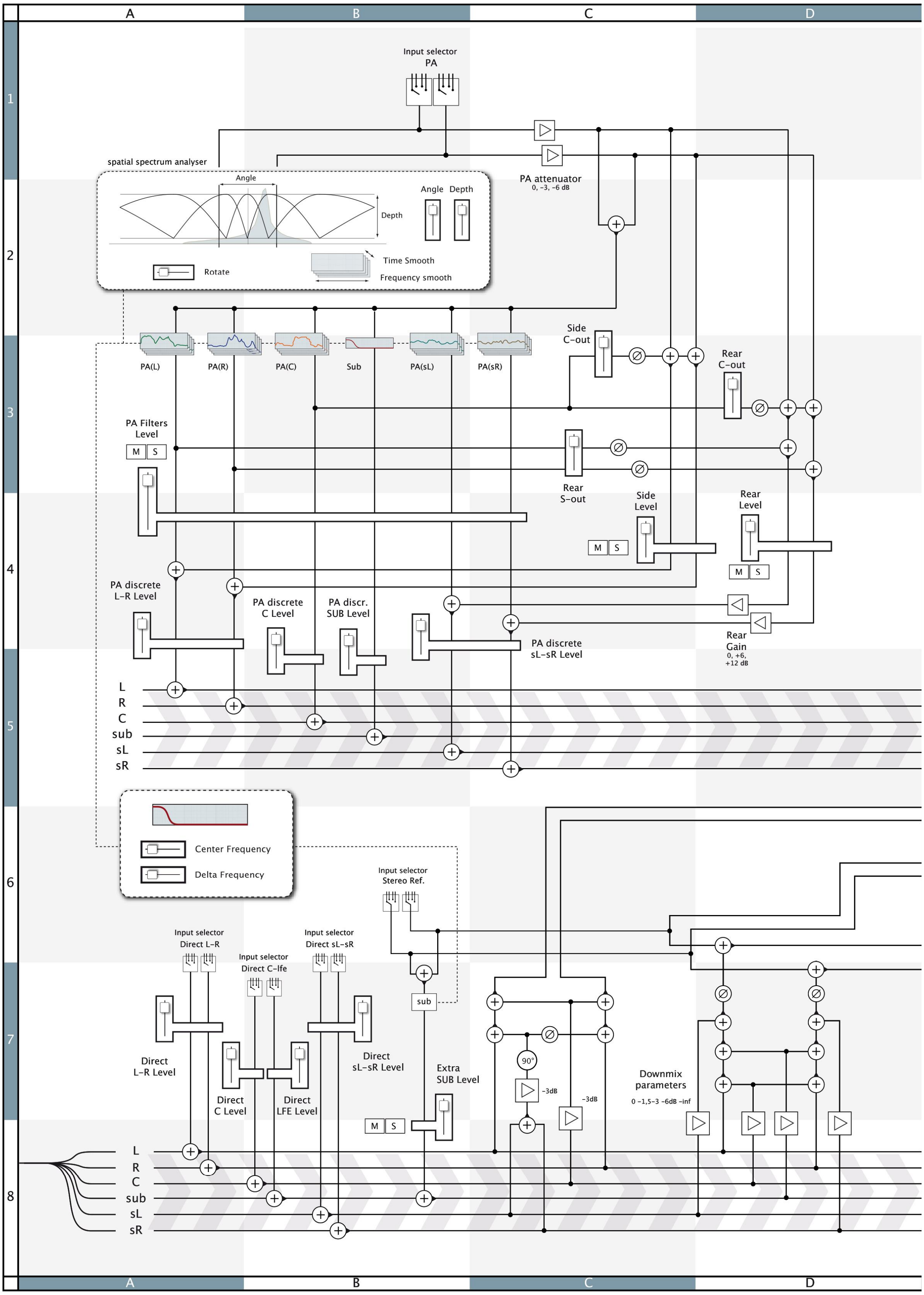
The AES/EBU connector wiring is compatible to the Tascam pinout:

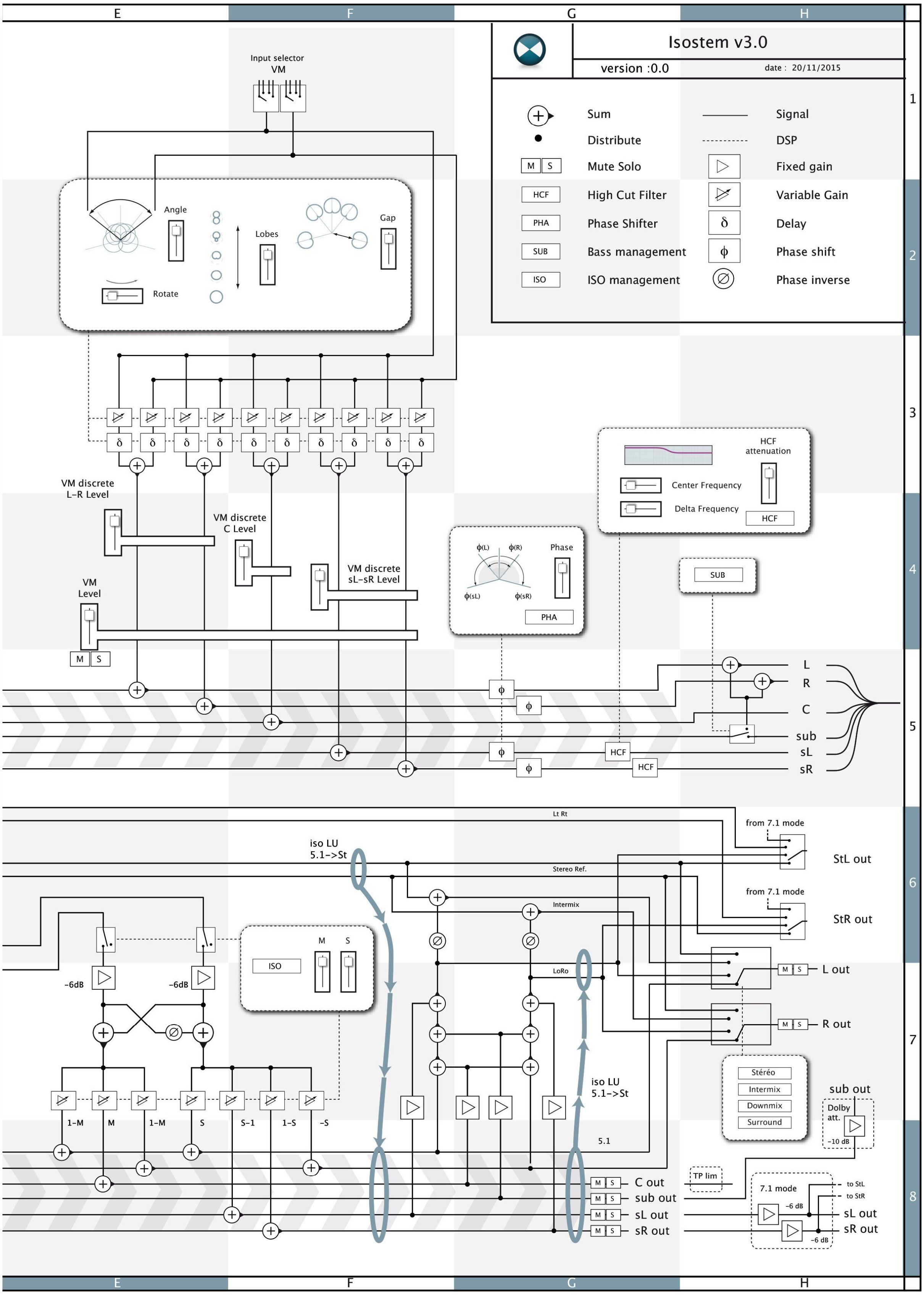
TASCAM - DIGIDESIGN
Sub-D 25 pining

AES In 1-2	hot(+)	pin 24
	cold(-)	pin 12
	GND	pin 25
AES In 3-4	hot(+)	pin 10
	cold(-)	pin 23
	GND	pin 11
AES In 5-6	hot(+)	pin 21
	cold(-)	pin 9
	GND	pin 22
AES In 7-8	hot(+)	pin 7
	cold(-)	pin 20
	GND	pin 8

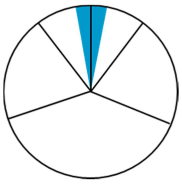
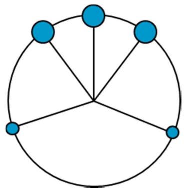
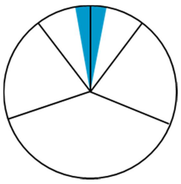
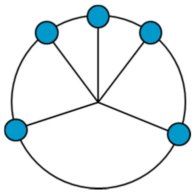
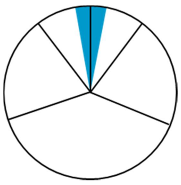
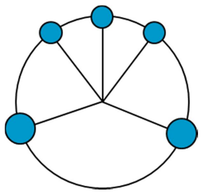
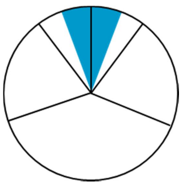
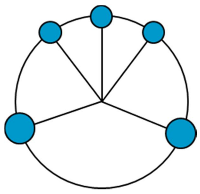
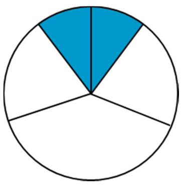
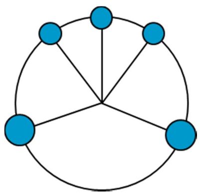
AES Out 1-2	hot(+)	pin 18
	cold(-)	pin 6
	GND	pin 19
AES Out 3-4	hot(+)	pin 4
	cold(-)	pin 17
	GND	pin 5
AES Out 5-6	hot(+)	pin 15
	cold(-)	pin 3
	GND	pin 16
AES Out 7-8	hot(+)	pin 1
	cold(-)	pin 14
	GND	pin 2

Appendix B: Schematic Block Diagram




E
F
G
H
Isostem v3.0
version :0.0
date : 20/11/2015
Sum
Distribute
Mute Solo
High Cut Filter
Phase Shifter
Bass management
ISO management
Signal
DSP
Fixed gain
Variable Gain
Delay
Phase shift
Phase inverse
HCF
PHA
SUB
ISO
1
2
3
4
5
6
7
8
Input selector
VM
Angle
Rotate
Lobes
Gap
VM discrete
L-R Level
VM discrete
C Level
VM discrete
sL-sR Level
VM
Level
M S
HCF
attenuation
Center Frequency
Delta Frequency
Phase
L
R
C
sub
sL
sR
Lt Rt
iso LU
5.1->St
Stereo Ref.
Intermix
LoRo
from 7.1 mode
StL out
StR out
L out
R out
-6dB
1-M
M
S
S-1
1-S
-S
Stéréo
Intermix
Downmix
Surround
sub out
Dolby
att.
-10 dB
5.1
C out
sub out
sL out
sR out
TP lim
7.1 mode
-6 dB
to StL
to StR

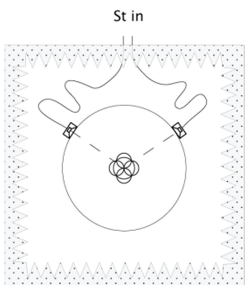
Appendix C: Factory Presets

Preset	Type	Diversity	Rear Level	Input / Ouput
1	Upmix	 narrow	 discreet	Stereo In: AES 4 In Upmix L/R: AES 1 Out Upmix C/LFE: AES 2 Out Upmix sL/sR: AES 3 Out Stereo out: AES 4 Out
2	Upmix	 narrow	 middle	Stereo In: AES 4 In Upmix L/R: AES 1 Out Upmix C/LFE: AES 2 Out Upmix sL/sR: AES 3 Out Stereo out: AES 4 Out
3	Upmix	 narrow	 strong	Stereo In: AES 4 In Upmix L/R: AES 1 Out Upmix C/LFE: AES 2 Out Upmix sL/sR: AES 3 Out Stereo out: AES 4 Out
4	Upmix	 middle	 strong	Stereo In: AES 4 In Upmix L/R: AES 1 Out Upmix C/LFE: AES 2 Out Upmix sL/sR: AES 3 Out Stereo out: AES 4 Out
5	Upmix	 wide	 strong	Stereo In: AES 4 In Upmix L/R: AES 1 Out Upmix C/LFE: AES 2 Out Upmix sL/sR: AES 3 Out Stereo out: AES 4 Out
6	Bypass	-/-	-/-	AES 1..4 In : AES 1..4 Out

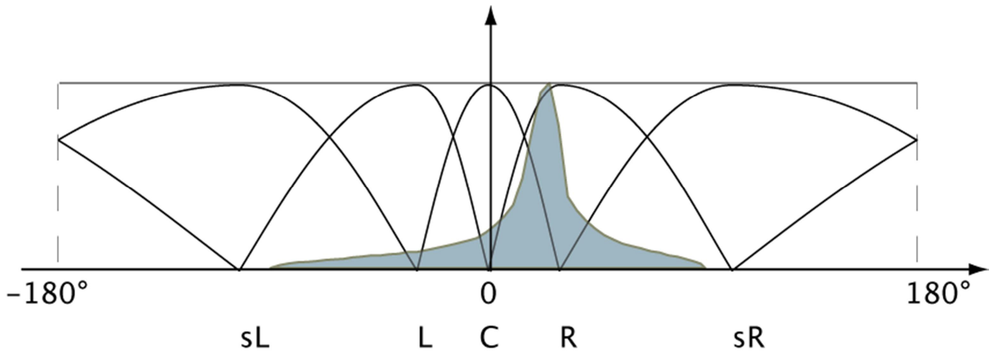
Please note: These presets represent the factory settings of firmware v3.0.

APPENDIX D: PANORAMIC ANALYZER PHYSICS

Acoustic modeling has the goal of extracting the ghost sources of the stereo signal by their positioning in the stereo mix. The signal is treated by acoustic simulation and the pressure and pressure gradient data establish the distribution of the energy fluxes in the frequency domain at the listening point.



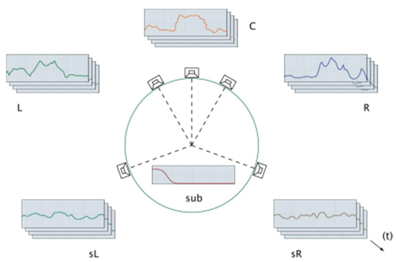
Acoustic Simulation



Energy Distribution and Panoramic Law

This distribution is modeled, then projected on the panoramic law of the reproduction system (here five channels in the ITU-R BS.1770 standard), resulting in a matrix of coefficients normalized for each channel and every frequency.

$$
\begin{bmatrix}
\alpha_{sL_0} & \alpha_{L_0} & \alpha_{C_0} & \alpha_{R_0} & \alpha_{sR_0} \\
\alpha_{sL_1} & \cdot & \alpha_{C_1} & \cdot & \alpha_{sR_1} \\
\cdot & \cdot & \cdot & \cdot & \cdot \\
\cdot & \cdot & \cdot & \cdot & \cdot \\
\cdot & \cdot & \cdot & \cdot & \cdot \\
\cdot & \cdot & \cdot & \cdot & \cdot \\
\alpha_{sL_{1022}} & \cdot & \alpha_{C_{1022}} & \cdot & \alpha_{sR_{1022}} \\
\alpha_{sL_{1023}} & \alpha_{L_{1023}} & \alpha_{C_{1023}} & \alpha_{R_{1023}} & \alpha_{sR_{1023}}
\end{bmatrix}
$$



Matrix of coefficients

Frequency Envelopes

A document itemizing the entire calculation is available in the AES (Audio Engineering Society) papers library:

AES Paper 6548; AES Convention 119; October 2005.