Application Note

Automatic Audio Synchronization & Channel Assignment (ASCA)

Revision 1.0
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Note. *This application note is a supplement to the PVD 56xx/58xx manuals until the current versions of these manuals are updated.*

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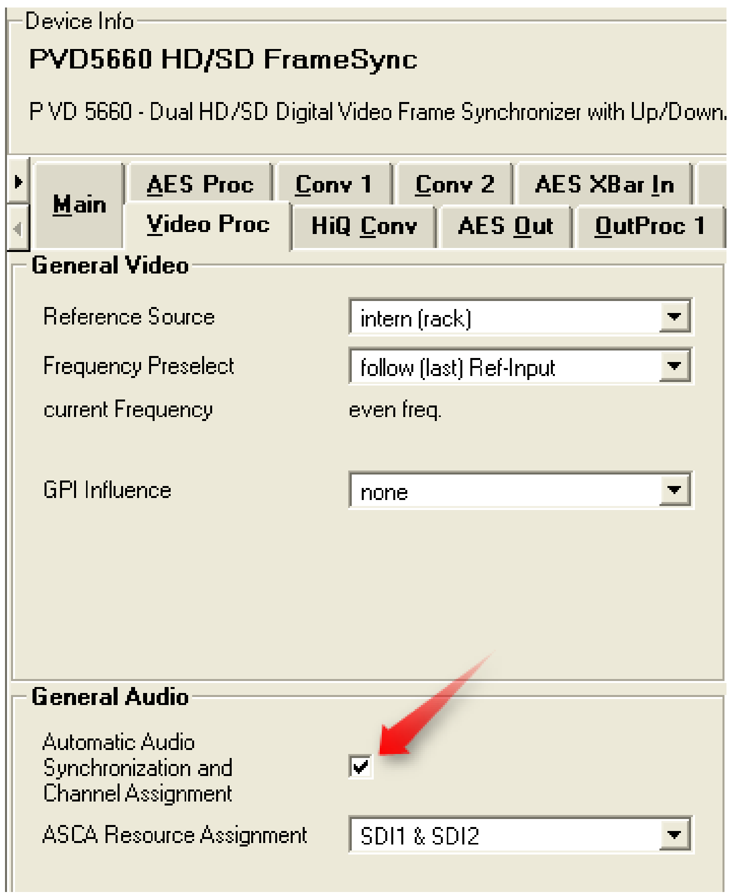
Introduction

The Automatic Audio Synchronization and Channel Assignment (ASCA) function of the product PVD56xx/PVD58xx can be used to make sure that incoming embedded audio streams are automatically synchronized and then routed correctly to the appropriate output (e.g. embedded into the same group of the same video program). External AES inputs are not usable while the ASCA function is switched on.

Note: The ASCA function will be applied only if the user has explicitly enabled it for the individual device (PVD56xx/PVD58xx). By default, the ASCA function is NOT enabled (i.e. all crossbars have to be configured manually).



Working Principle

Depending on the type of audio content (PCM, DolbyE, other data ...), different synchronization methods and therefore different internal audio channels have to be used for each audio input stream. The ASCA function will automatically select the appropriate pathways by configuring the various internal audio crossbars.

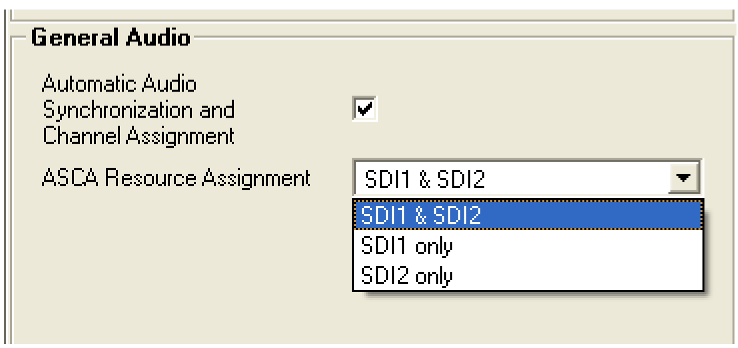
Limited Sync Resources

The amount of synchronization resources is limited. There are a total of 8 sample-rate-converters (SRC) and 4 DolbyE frame-synchronizers (DE-FS) available. In the case of a dual-channel processing device (e.g. PVD5660 or PVD5812 with 2ND_INPUT-Option), the user has to specify how these synchronization resources are assigned to the input SDI signals. The available choices are the following:

1. SDI1 + SDI2 *(this is the default)*
- 4 SRCs and 2 DE-FSs are available to synchronize the audio-content of SDI1
- 4 SRCs and 2 DE-FSs are available to synchronize the audio-content of SDI2

2. SDI1 only
- all 8 SRCs and 4 DE-FSs are available to synchronize the audio-content of SDI1
- audio-content of SDI2 cannot be synchronized and can only be passed through as is (embedded)

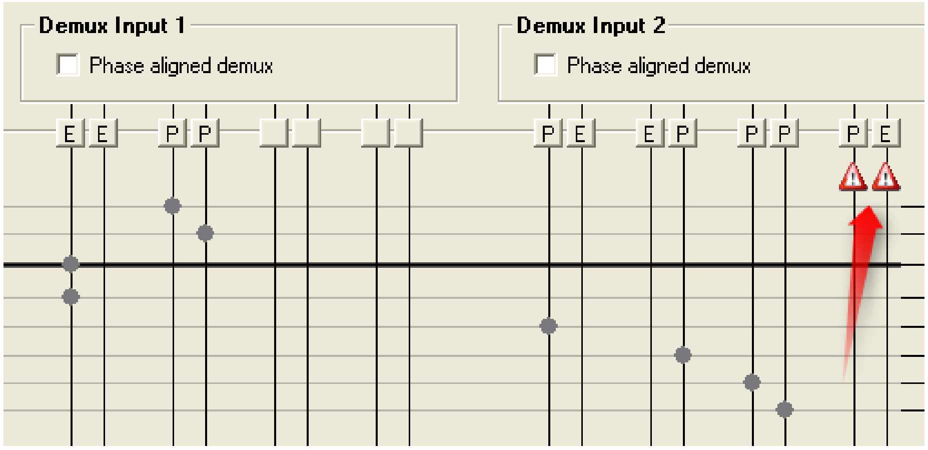
3. SDI2 only
- audio-content of SDI1 cannot be synchronized and can only be passed through as is (embedded)
- all 8 SRCs and 4 DE-FSs are available to synchronize the audio-content of SDI2



NOTES:

- *As mentioned above, this user-decision is relevant only in products with two inputs and when both of these inputs are active.*
  If only one input is usable in the product, the "SDI1 only" mode is automatically active.
- *In any of the non-shared modes ("SDI1 only" or "SDI2 only"), the other SDI input can only be used with signals that are either*
  (a) synchronous -OR-
  (b) do not contain any embedded audio of any format.

In the event that the available synchronization resources should be exceeded, the remaining audio content will be passed through **un-synchronized**. Accordingly, an appropriate warning will be visible in the GUI. This situation can occur even in a single-channel product, and independently of the above user decision on resource allocation per channel. As an example, this situation can occur if there are more than 4 DolbyE streams embedded in the incoming SDI signal. Or, as another example, when a dual-channel device is operated in "SDI1 + SDI2" mode and more than 4 'AES' streams containing PCM-audio are embedded in one of the incoming SDI signals.



The available resources per SDI channel are applied to the embedded audio streams in the following order of priority (if one of the mentioned audio-streams is not available, it will not be assigned any resources):

1. audio streams deembedded from group 1
2. audio streams deembedded from group 2
3. audio streams deembedded from group 3
4. audio streams deembedded from group 4
5. in the event of unavailable synchronization resouces, a warning will be issued and the remaining audio-channels will be process un-synchronized. All audio signals will be delivered on the output in any case. So the limitation applies only in an asynchronous environment.

Limitations:

1. audio disturbances
Whenever the ASCA function is re-configuring the audio-channels, the configuration process will possibly generate audible disturbances in some of the audio output channels (embedded or AES) of the same video program. Such re-configuration will be triggered by any change of the appropriate input configuration (video, embedded audio). Therefore this function is recommended to be used in environments, in which the incoming signal configuration does not change while a programming stream is being processed. I.e. it can be used for automatic pre-setup only.

2. flexibility: crossbars, processing
Using the ASCA function imposes the following limitations to the audio infrastructure:

- It will make the internal Audio Processing (mute, gain, invert, …) unavailable (set to neutral)
- It will take control over all internal audio-crossbars (input and output), except the crossbar configuring the external AES output channel assignment.

Accordingly, the appropriate audio-crossbars and processing parameters will be greyed out and set to read-only.

3. persistence of user settings
After turning the ASCA function ON, audio-infrastructure settings (crossbars, SRCs, Embedders) are modified by an automatic process. When the ASCA function is then turned OFF again, previous settings are *not* automatically restored. As a consequence, turning the ASCA ON and OFF will possibly result in a modified audio-infrastructure (crossbars, processing).

4. external AES input not usable
Turning the ASCA function ON will allocate all available audio-synchronization resources to the signals deembedded from video inputs SDI1 and SDI2. The external AES inputs cannot be used at all.
On the other hand, the external AES outputs are not controlled by ASCA. I.e. the "AES output" crossbar will still be usable (not greyed out). However, please note that the automatic ASCA process can re-assign individual audio streams to different internal audio channels. So, if an external AES output is connected to a particular internal audio stream, the content of that stream can change spontaneously, because ASCA has modified the AES input crossbar, following a change of the input configuration.

5. DolbyE Framesync timing assignment
The ASCA function will use the available DE-FS channels to provide frame-synchronization and guard-band alignment of the DolbyE frame structure. If the same video signal is assigned to multiple video outputs (using the video crossbar on the main page of the GUI), and if those video outputs use a different timing offset (relative to the current sync. input), then the correct audio/video timing of the DolbyE stream can only be guaranteed for the first of those SDI outputs. For details refer to section "Maintaining DolbyE Transparency" / "Pathway 3 – Dolby E" of the product user manual.

Required Firmware/Software Version

PVD 56xx/58xx Firmware Version 402 or higher

Lynx Desktop Controller Version 4.8.2 or higher

