Adjusting and Normalizing Output Level of SL NEO Server Audio Signal

Functionality of SL NEO server software allows adjusting the output level and normalizing the output audio signal. The functions are useful while forming SDI-signals with embedded sound, and especially when forming IP/ASI transport streams, so as in the last case installing de-embedders/embedders and hardware devices for audio processing is unavailable or inappropriate. Let's look at the details of configuring the functional at the server platform.

Adjustment of Output Audio Level

Adjusting parameters is implemented from the control panel - Administrator Control Panel. Login to the console is implemented locally from the server or from any computer in the network at http://server_ip:7901. Login to the control web console as an administrator.

After the login to the control console, select Manage at the menu. Find the Video IO Boards tab and start editing parameters of a corresponding Playout service. Find the Base Audio Level field in the General tab and set the value in decibels that will be used to increase or decrease the output audio level.

The function is applied to all audio tracks in the output audio signal of the selected Playout service, the default value is 0 dB. The level regulator is placed in the final chain of the server audio-path.

	Configure Service	
Ethernet port: (Add Service Clear Services)	-Mode	
Playout	Capture Playout	
X Playout_2(Edit) MPEG2TS/IP/Ember		RC Audio Channel Map MPEG2 TS Parameters IP Paramete
	Video Output:	MPEG2TS/IP •
	Audio Output:	Embedded 💌
	Video Mode:	1080i50
	Keyer Mode:	None -
	Base Audio Level:	0 Db
	Preview	+11 Db 1
	Force Deinterlace	+10 Db +9 Db
		+9 Db
		+7 Db
		+6 Db +5 Db
		+4 Db
		+3 Db
		+2 Db = +1 Db
		0 Db
		-1 Db
	Ok Cancel	-2 Db -3 Db
		-4 Db
		-5 Db -6 Db
		-6 Db -7 Db

At the end of editing, press Apply in the top of the control console window to apply changes.

Normalization of Output Audio Level

Compressor

The function of normalizing the audio level is provided for manual and automatic import of media files to the SL NEO server storage. Besides, you can activate the built-in compressor/limiter (the function DRC - Dynamic Range Compression) for on-line post-processing of audio signals formed by the server.

With different audio levels in played back files and signals that come to server inputs, the Audio DRC software module (placed at the output part of the server audio-path) increases or decreases the transfer ratio. Thus, the average output audio level is maintained unchangeable. The Audio DRC module works both with monophonic signals and multi-channel audio, simultaneously processing up to 16 audio channels within one server output playout channel.

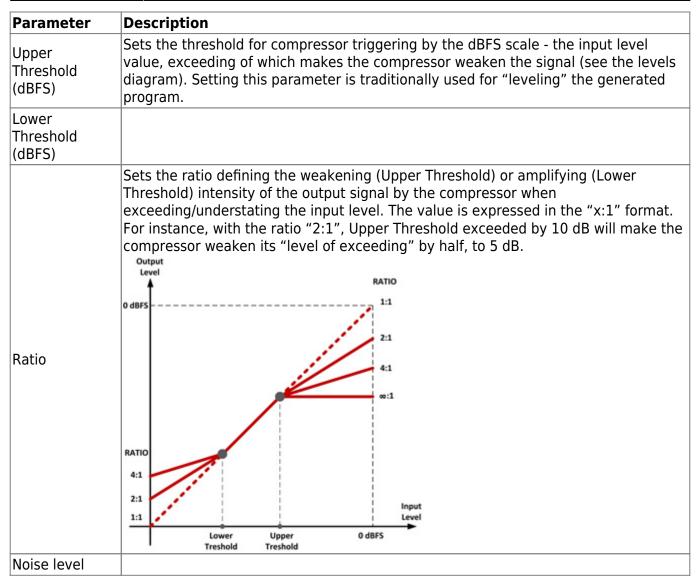
Audio DRC works with all types of output signals and interfaces supported by SL NEO servers: SD/HD SDI/HDMI, Audio Embedded, AES EBU, Analog Audio and DVB IP/ASI transport streams. Parameters of program compressors/limiters of every output server channel can be individually configured.

Configuring parameters of level automatic adjustment is implemented from the control panel -Administrator Control Panel. Select Manage in the console menu, then pick the output device for configuring: find the Video IO Boards tab and edit parameters of the corresponding Playout service.

Configure Service			
Mode			
Genlock Capture	Playout		
General Audio DRC	Audio Channel Map		
Enable DRC			
Level detection:	RMS -		
Upper threshold (dBFS):	-18 -		
Lower threshold (dBFS):	Not set 💌		
Ratio:	2 -		
Noise level (dbFS):	Not set 🔹		

The Audio DRC tab contains settings for normalization of output audio level.

Parameter	Description	
Enable DRC	The option launches the level processor	
Defines the compressor triggering type: •		
	Peak - triggering by the level peak values. This way is used to limit the signal peak values to a certain threshold.	
	RMS - triggering by the level average value. The RMS-compressor can miss short signal peaks with practically no decrease to the transfer ratio. The RMS-method is mostly used in compressors for volume leveling, so as loudness is more connected to mean-square capacity rather than peak capacity.	



The compressor doesn't allow to adjust the **Attack time** and **Release time** parameters.

Attack time displays the speed necessary for scanning the audio signal so to consider changes in the sound level. The set value of 200 msec is selected as the most optimal one.

The parameter **Release time** defines the time needed for the compressor to return to the condition of regular gain after the signal falling lower the inflection point. The set value about 5 msec is the minimum one.

Limiter

The limiter is the same compressor with the set Upper Threshold value, configured to limit the output level when exceeding the input one: input to output signal ratio starts from the 8:1 value. When using harder limitation, undesirable nonlinear distortions may appear in the output signal.



At the end of editing the Audio DRC module parameters, press Apply at the top of the server control console, to save the implemented settings.

Download

To see the Audio DRC module work in the RMS mode, use the avi-file containing fragments of a 1kHz signal with different levels (from -42 to 0 dBFS). The video contains information on the current levels values in a file - measuring the audio level from the server output, you can test the compression degree and correctness of settings.

From: https://wiki.skylark.tv/ - wiki.skylark.tv

Permanent link: https://wiki.skylark.tv/howto/output_audio_level_settings



Last update: 2019/06/11 09:12