



Cisco Linear Stream Manager

User Guide

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This guide contains information on getting started and getting the best results with the Cisco Linear Stream Manager (LSM), which provides centralized management of both Cisco AnyRes Live and Cisco AnyRes Packager products.

Introduction 1

About the Cisco Linear Stream Manager

Building on the success of the Cisco AnyRes Live family of live encoding solutions, Cisco offers a comprehensive management solution for multi-encoder and multi-packager streaming operations: the Cisco Linear Stream Manager (LSM). The LSM provides a consolidated management, configuration, and monitoring platform for both multi-encoder and multi-packager deployments. The LSM is a scalable solution that can meet demands for large deployments while providing additional value for any size installation. The LSM automates standard tasks and makes it easy to monitor multiple devices at a glance. In addition, features like global scheduling, multiple levels of redundancy, and live monitoring make the LSM an indispensable ally in any installation.

Cisco LSM Features:

Consolidated Management

- Ability to manage multiple AnyRes Live encoders and AnyRes Packagers from a single intuitive interface

Configuration

- Enhanced encoder and packager configurations using templates and overrides
- Optimized configurations for 24x7 Channel deployments
- Bulk packager Channel configuration interface

Redundancy

- Encoder Teaming, 1+N, and N+M encoder redundancy options
- Encoder SDI Router switching on failover
- Packager Channel High Availability (N:M redundancy, and failover resiliency)
- Dynamic Channel load balancing across multiple packagers

- Configurable packager failover thresholds

Monitoring

- Status monitoring of alarms and statistics during live streaming
- Customized reports for events, alarms, and logging
- Global calendar to optimize resource allocation for scheduled events
- Scheduled encoder software updates and maintenance events
- Scheduled encoder Mid-Events for slates and SDI router switching
- SDI Router Management for additional automation and redundancy

The LSM implementation consists of two components:

1. Cisco Linear Stream Manager Server—This is the application that manages all communication to the AnyRes Live encoders and AnyRes Packagers. It is normally installed on a dedicated server.
2. Cisco Linear Stream Manager Client—This Java-based application is the user interface, which can be installed on multiple systems. It communicates with the LSM Server to display information about all connected encoders and packagers, and provides access to all features of the LSM Server.

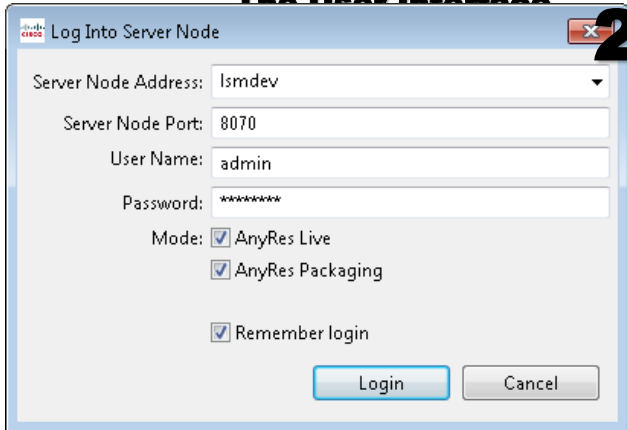
The LSM can manage the following resources:

1. Cisco AnyRes Live—These are the live encoding/transcoding nodes responsible for receiving live video and audio sources and encoding/transcoding them into adaptive transport stream (ATS) formats for downstream ABR packaging. Optionally, these encoders may also package them directly into the appropriate streaming format. AnyRes Live uses a messaging service to communicate with the LSM Server.
2. Cisco AnyRes Packager—These are the dense standalone servers responsible for receiving adaptive transport streams (ATS) from an upstream encoder and packaging them into various adaptive bitrate streaming formats. AnyRes Packager uses a RESTful API service for communication to the LSM Server.

Note: The LSM is required for all AnyRes Packager deployments.

Logging In

The User Interface



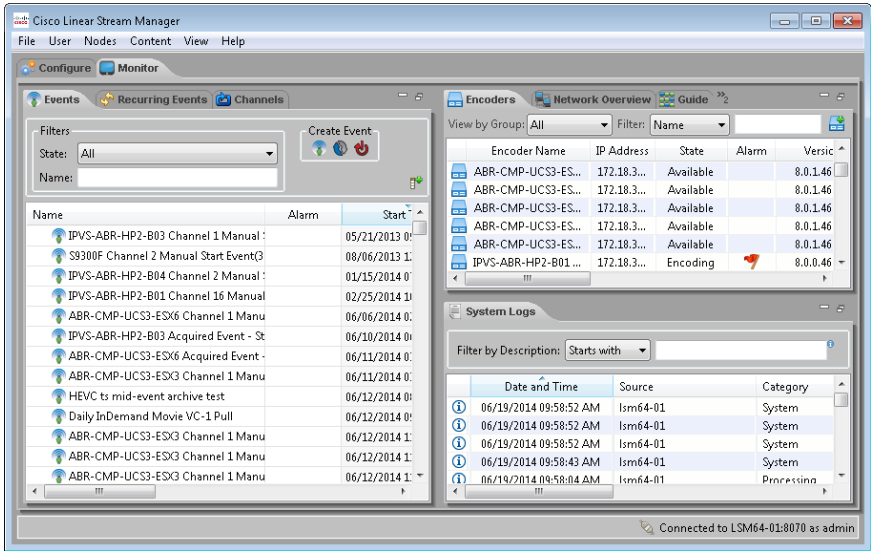
When you open the Cisco Linear Stream Manager (LSM), the Log into Server Node dialog box appears. Enter the address and port for the Cisco Linear Stream Manager (LSM) Server Node, and your username and password on that server. The default port for an LSM Server Node is 8070. The LSM ships with two default usernames, admin and user1. We recommend that you change the password immediately the first time you log in. See “Changing a Password” on page 263.

The Mode check boxes determine whether the LSM displays panes that apply to management of AnyRes Live encoders, AnyRes Packagers, or both. Check the **AnyRes Live** check box to only display encoder-related areas of the LSM. Check the **AnyRes Packaging** check box to only display packager-related areas of the LSM. Check both check boxes to display all areas of the LSM.

Optionally check the **Remember Login** check box to display this username and password in these fields for subsequent logins, and click **Login** to log in to the LSM Server.

See “Logging In” on page 262 for more details on logging in to the LSM server.

Navigating the User Interface



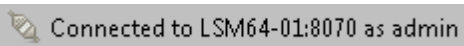
There are two major tabs for the LSM:

- **Configure Tab**—This portion of the interface is where Encoder Presets, Channel Templates, Linear Service Templates, Encoder Groups, Packagers, and Packager Groups can be created and edited. Also, reports can be edited and run. All users can view these items; administrator authority is required to create or edit.
- **Monitor Tab**—This portion of the user interface is where the details of Channels, events, encoders, packagers, and system logs are displayed. It is also where encoders are added or removed.

In the bottom right corner of the user interface, the LSM displays the connection status of the server. Before successful login, the LSM displays the following status:



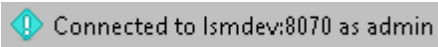
After successful login, the LSM displays the following status, identifying the server name or IP that is connected:



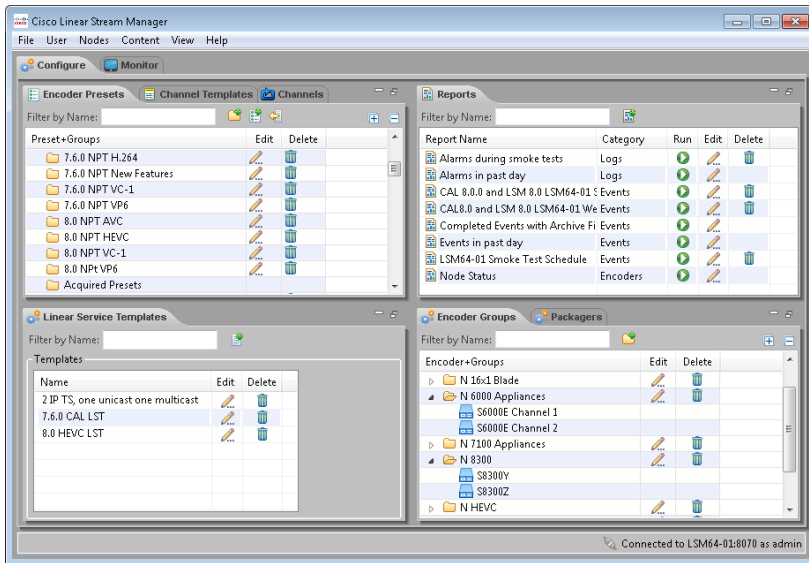
If the current versions of the LSM Server and client are incompatible, a dialog box will provide a warning. Click **OK** to continue. We strongly recommend that you update the

CISCO LINEAR STREAM MANAGER

software version. If you continue, the connection status will display the following warning icon:



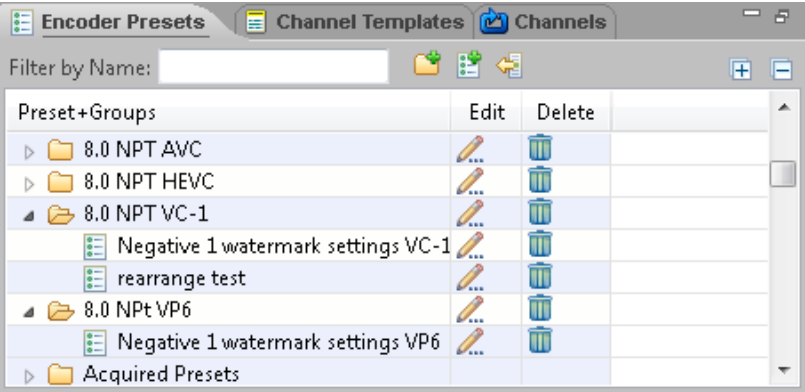
Configuration




If you choose the AnyRes Live or AnyRes Packaging only views at login or from the View menu, only the tabs that apply to that view are displayed.

Encoder Presets

Encoder Presets are used to store encode settings for AnyRes Live encoders. Several default presets and preset groups are included with the LSM. Preset groups can be used to organize presets for ease of use.



Click the **New preset**  icon to open the preset editor.

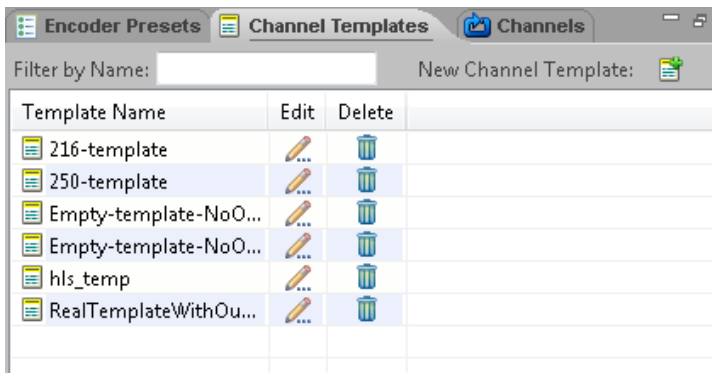
The preset editor is divided into nine areas of configuration, each with its own tab:

- Input Video—Provides Video input configuration fields.
- Input Audio—Provides Audio input configuration fields.
- VC-1—Provides Video, Audio, ASF Output, and HSS Output configuration fields for VC-1 streaming.
- VP6—Provides Video, Audio and Output Configurations for VP6 streaming.
- H.265—Provides Video, Audio, TS Output, HLS Output, and DASH Output Configurations for H.265 streaming.
- H.264—Provides Video, Audio, RTMP Output, TS Output, HLS Output, HSS Output, HDS Output, DASH Output, and RTP/3GPP Output Configurations for H.264 streaming.
- H.263—Provides Video, Audio, and RTP/3GPP Output Configurations for H.263 streaming.
- Summary—Provides the summary of specified encodings and its values.
- Rearrange Streams—Provides options to rearrange, copy, and delete streams.

For more detailed information on Encoder Presets and Preset Groups, see the “Presets and Linear Service Templates for Encoders” chapter on page 32.

Channel Templates

Channel Templates are used to store settings for creating Channels for AnyRes Packagers.



To create a new template, click the **New Channel Template** icon to display the Channel template editor.

This editor is divided into five areas of configuration, each with its own tab:

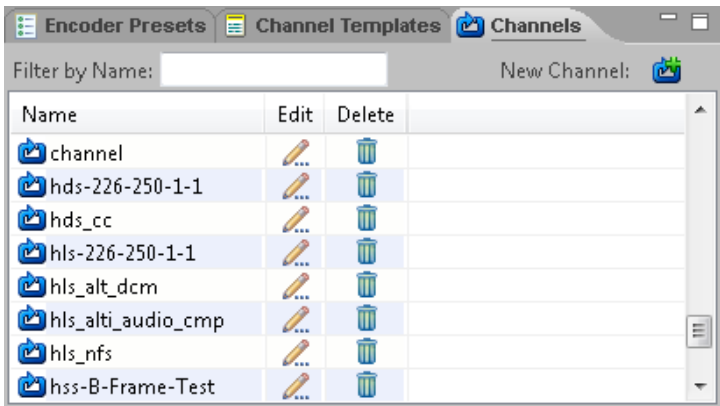
- Input—Provides input configuration fields.
- Output—Provides output configuration fields.
- Mapping—Provides mapping of inputs to outputs.
- Presentation—Provides configuration of master index files.
- Global—Provides ESAM parameters.

The Channel editor, available from within the Channel configuration, includes an additional Channel Redundancy tab that provides configuration for redundant instances of a Channel. Some other settings, such as PlayReady settings, are only available in the Channel editor and will not appear in the Channel template editor.

For more detailed information on Channel Templates, see the “Channel Templates for Packagers” chapter on page 158.

Channels

A Channel for packagers can be based on an existing Channel template or can be created without a template in the New Channel wizard. On the Channels tab on the Configure tab, click the **New Channel** icon to open the New Channel wizard.

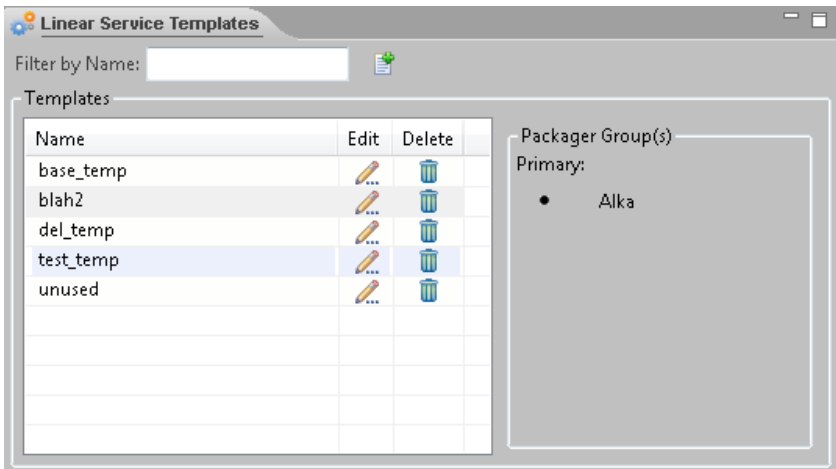


For more detailed information on configuring Channels for encoders, see “New Channel” on page 206.

For more detailed information on configuring Channels for packagers, see the “Channels for Packagers” chapter on page 210.

Linear Service Templates

For encoders, an LST specifies the essential makeup of a Live Streaming Event or Channel.

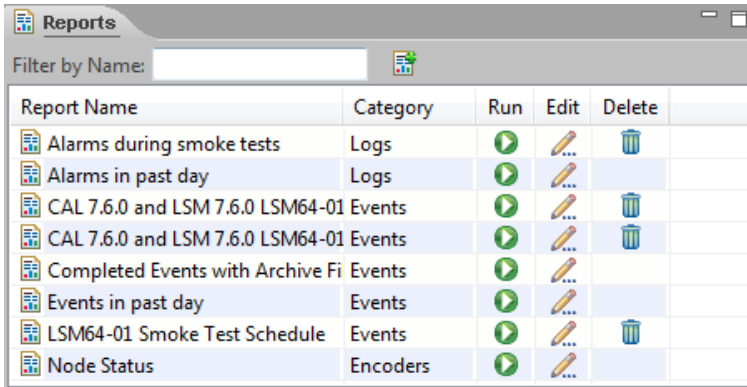


Click the **New Linear Service Template** icon and the New Linear Service Template wizard appears.

For more detailed information on LSTs for encoders, see “Linear Service Template Creation” on page 152.

Reports

For encoders, the Reports tab allows the creation, editing, and generation of reports on encoder nodes, events, and logs.



Report Name	Category	Run	Edit	Delete
Alarms during smoke tests	Logs			
Alarms in past day	Logs			
CAL 7.6.0 and LSM 7.6.0 LSM64-01 Events	Events			
CAL 7.6.0 and LSM 7.6.0 LSM64-01 Events	Events			
Completed Events with Archive Fi	Events			
Events in past day	Events			
LSM64-01 Smoke Test Schedule	Events			
Node Status	Encoders			

A report can be run to view immediately, saved, or scheduled to generate a file that can be emailed or stored in a network location.

Standard Reports

Standard reports are provided by the LSM for commonly useful report data. Standard reports cannot be edited or scheduled. To schedule a standard report, you will need to save a copy of the standard report and schedule the copy.

Custom Reports

Custom reports are user-defined reports that produce results according to search criteria. Custom reports can be run immediately or scheduled to automatically run on a regular basis.

Create a Report

Click the **New Report**  icon to define custom reports.

For more detailed information on reports, see “Reports” on page 254.

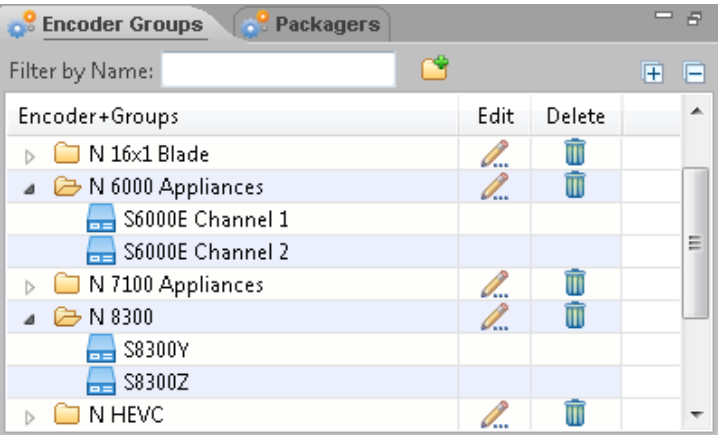
Encoder Groups


The Encoder Groups tab allows the creation and editing of groups of encoder nodes.

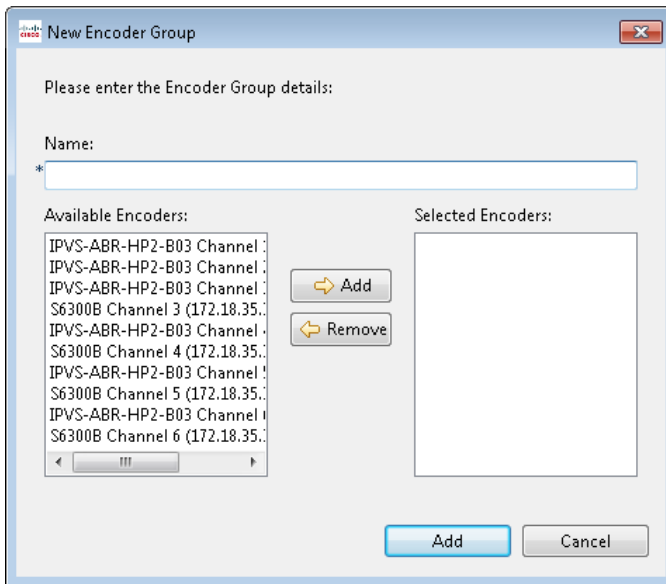
An encoder group consists of a set of encoders available to the LSM, all of which have the same license. The LSM will accept the first encoder added to the group, and then will restrict further additions to encoders with the same license. The encoder group will be in an offline state if all encoders in the group are offline, unavailable, or change licenses. If an encoder group is offline, it can only be edited to remove encoders.

If you define encoder groups, you can use them for redundancy or to restrict the list of encoders to view in other panes and dialog boxes.


Create a New Encoder Group

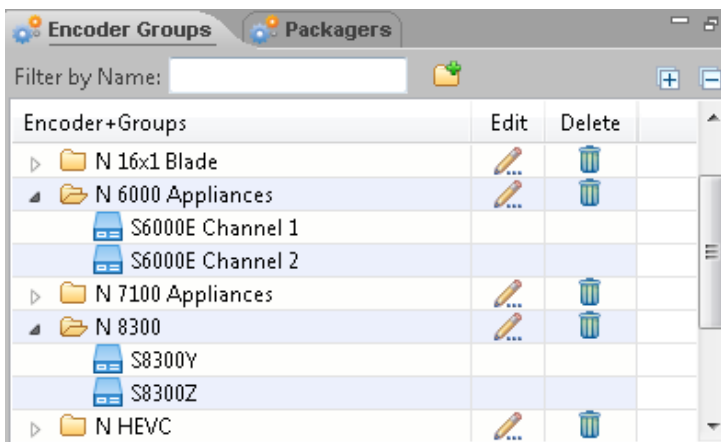


Click the **New Encoder Group**  icon and the New Encoder Group dialog box appears.



Edit an Encoder Group

To edit an existing encoder group, click the **Edit**  icon next to the group name.



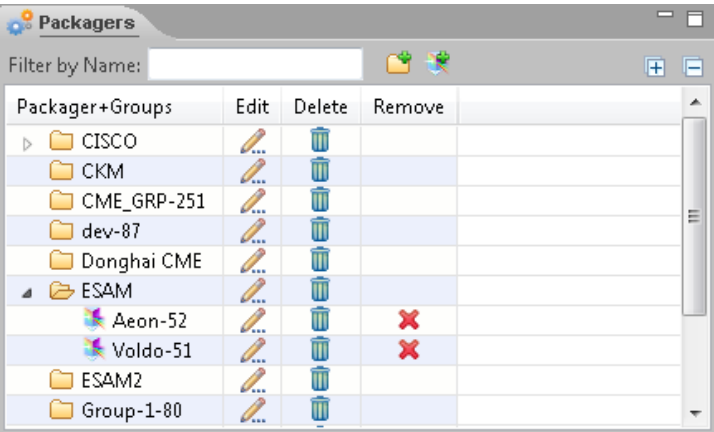
Packagers

The Packagers tab allows the addition of a packager as a managed resource, as well as the creation and editing of groups of AnyRes Packager nodes.

A packager group consists of a set of AnyRes Packagers available to the LSM. Any of the packagers in the group may be chosen by the LSM when a Channel is started, based on available capacity.

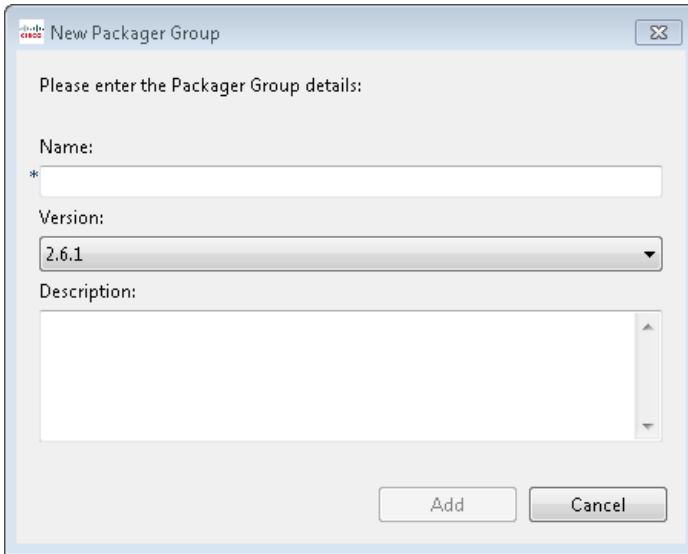
You must first create a packager group, and then add one or more packagers to that group. All packagers must belong to a group to be used for Channels. Packagers may only belong to one group.

Create a New Packager Group



Click the **New Packager Group** icon and the New Packager Group dialog box appears.

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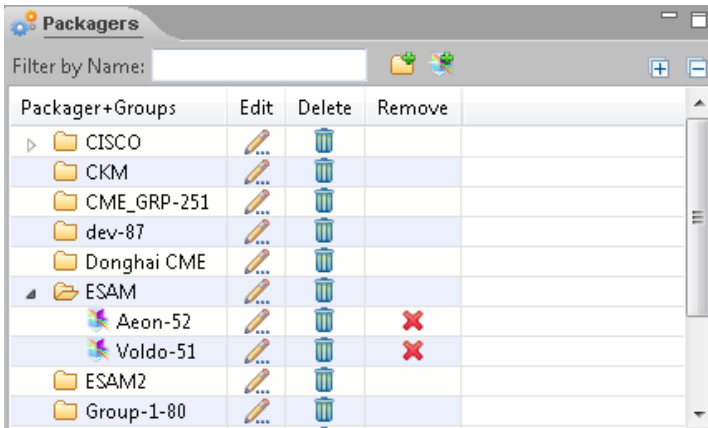


The 'New Packager Group' dialog box is shown. It has a title bar with the Cisco logo and the text 'New Packager Group'. The main area contains the text 'Please enter the Packager Group details:'. Below this are three fields: 'Name:' with a text input field, 'Version:' with a dropdown menu showing '2.6.1', and 'Description:' with a large text area. At the bottom are two buttons: 'Add' and 'Cancel'.

Enter the packager group name, the lowest software version of the packagers that will comprise the group, and an optional description, and then click **Add** to create the group.


If you are upgrading to LSM 9.0, the version of existing packager groups is unaffected. Packagers in this group may be upgraded to higher versions, but new features will not be enabled until the group is updated to the version that supports the new features. Before raising the group version to a higher software version, each packager in the group must be upgraded to that version or later.

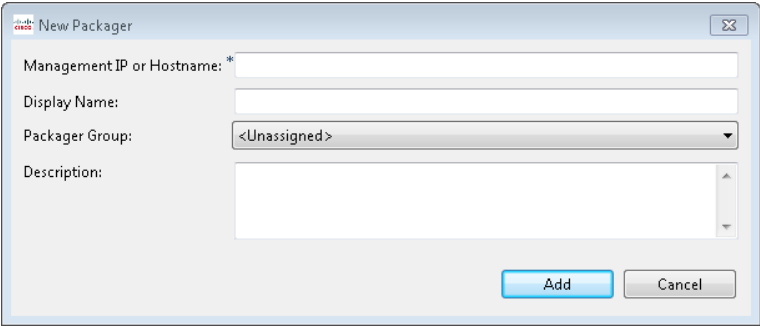
Add a Packager



The 'Packagers' window shows a list of packager groups. The title bar says 'Packagers'. Below the title bar is a 'Filter by Name:' text box. The main area is a table with columns: 'Packager+Groups', 'Edit', 'Delete', and 'Remove'. The table lists several groups, including CISCO, CKM, CME_GRP-251, dev-87, Donghai CME, ESAM, Aeon-52, Voldo-51, ESAM2, and Group-1-80. The 'Remove' column has red 'X' marks for Aeon-52 and Voldo-51.

Packager+Groups	Edit	Delete	Remove
▶ CISCO			
▶ CKM			
▶ CME_GRP-251			
▶ dev-87			
▶ Donghai CME			
▶ ESAM			
▶ Aeon-52			
▶ Voldo-51			
▶ ESAM2			
▶ Group-1-80			

Click the **New Packager**  icon and the New Packager dialog box appears.




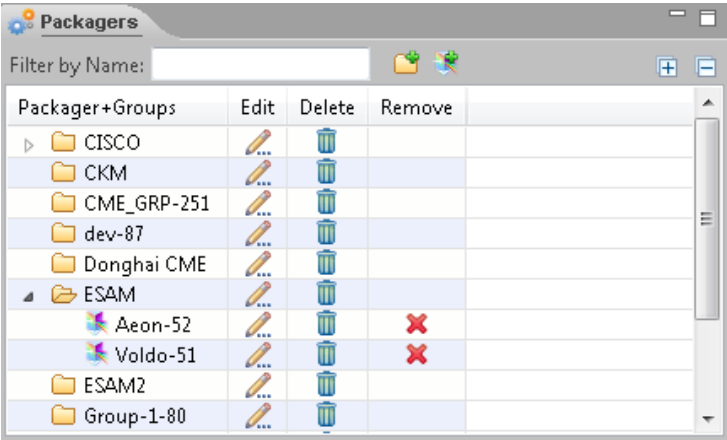
The 'New Packager' dialog box contains the following fields and controls:

- Management IP or Hostname: *
- Display Name:
- Packager Group: <Unassigned>
- Description:
- Buttons: Add, Cancel























Enter the packager management IP or hostname, optional display name, and optional description. You can choose an existing packager group, or leave the packager in the Unassigned group to be moved to a group at a later time. Click **Add** to add the packager.



Edit a Packager or Packager Group

To edit an existing packager or packager group, click the **Edit**  icon next to the group or packager name.

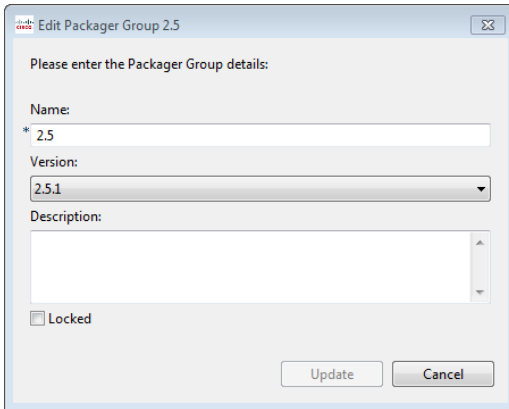


The 'Packagers' window displays a list of packagers and groups with the following columns:

Packager+Groups	Edit	Delete	Remove
▶ CISCO			
▶ CKM			
▶ CME_GRP-251			
▶ dev-87			
▶ Donghai CME			
▶ ESAM			
▶ Aeon-52			
▶ Voldo-51			
▶ ESAM2			
▶ Group-1-80			

You can edit any of the packager or group information, and you can choose a different group for a packager. To remove a packager from a specific group, click the **Remove**  icon. A removed packager will be placed in the Unassigned group. The **Delete**  icon will delete the packager from being managed by the LSM, but does not uninstall the packager from its host machine.

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Please enter the Packager Group details:

Name:
* 2.5

Version:
2.5.1

Description:

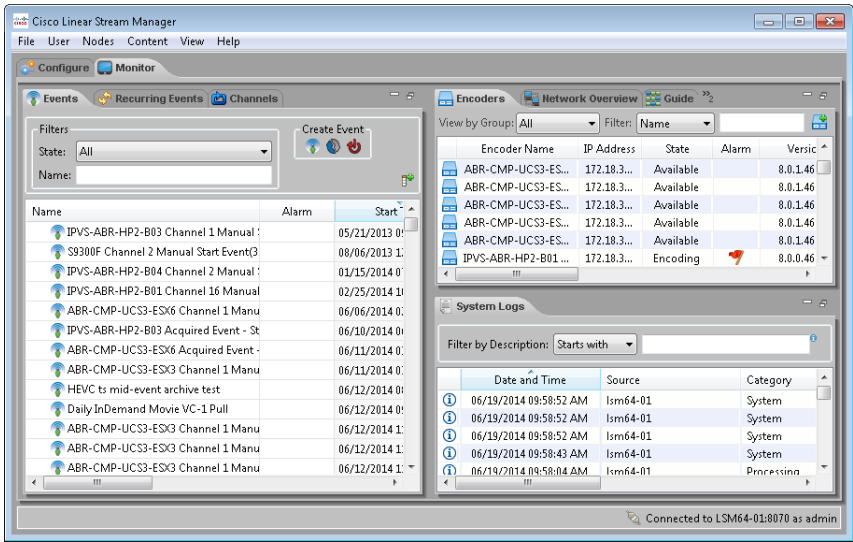
☐ Locked

Update Cancel

When editing a packager group, you can check the **Locked** check box to prevent the relocation of Channels in the group. Check the **Locked** check box, and then click **Update** to lock the group. On the Packagers tab, the group will now display a lock icon to indicate that it is locked.

Note: If you are upgrading to LSM 9.0, the version of existing packager groups is unaffected. Packagers in this group may be upgraded to higher versions, but new features will not be enabled until the group is updated to the version that supports the new features. Before raising the group version to a higher software version, each packager in the group must be upgraded to that version or later.

Monitoring



The Monitor tab displays information about Channels, Events, Encoders, Packagers, and System Logs. Detailed information on the functions of this tab is found in the “Monitoring” chapter on page 218.

Channels

The Channels tab contains a table of configured Channels. See “Channels for Packagers” on page 210, or “Event Creation and Channels for Encoders” on page 191 for more information on creating Channels.

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Name	Start/Stop	Resource/Group	State	Input State
all-publishers				
ats-esam				
ats-esam		Aeon-52		Running Ok
att_hd				
channel				
channel		ESAM		Not Running
hds-226-250-1-1				
hds_cc				
hds_cc		Group-1-80		Not Running
hls-226-250-1-1				
hls_alt_dcm				
hls_alti_audio_cmp				
hls_nfs				

Events

The Events tab contains a table of information. The columns of the tables are movable and sortable. To move a column, drag the column heading to the desired location. To sort the information in the table by a particular column, click the column heading.

Name	Alarm	Start Time	End Time	State
IPVS-ABR-HP2-B03 Channel 1 Manual Star		05/21/2013 09:25:09 AM	TBD	Completed
S9300F Channel 2 Manual Start Event(334 N		08/06/2013 12:21:45 PM	TBD	Completed
daily indemand movie from atl twc feed		08/30/2013 10:00:00 AM	09/04/2013 03:00:00 AM	Failed
S7100B Preset of Death Long Run via LSM		08/30/2013 01:34:37 PM	09/03/2013 08:12:22 AM	Completed
HP G7 B03		08/30/2013 02:16:41 PM	09/03/2013 11:00:00 AM	Failed
HP G7 B03 WSJ like preset long run via LSM		08/30/2013 02:18:31 PM	09/03/2013 11:00:00 AM	Failed
HP G7 B03 ch 3 WSJ like preset long run via		08/30/2013 02:22:32 PM	09/03/2013 11:00:00 AM	Failed
S6000e VC-1 HSS Long run via LSM		08/30/2013 03:13:03 PM	09/03/2013 11:00:00 AM	Failed

This tab can filter events to display only those events in the following states:

- Active
- Completed
- Draft

- Failed
- Scheduled

Encoders

The Encoders tab lists the encoder nodes and information specific to each. You can use the View by Group drop-down list to choose an encoder group to restrict the list of encoders displayed, or filter the list by the encoder name or IP address.

Encoder Name	IP Address	State	Alarm	Version
IPVS-ABR-HP2-B04 ...	172.18.36....	Offline		7.2.4.42290
IPVS-ABR-HP2-B04 ...	172.18.36....	Offline		7.2.4.42290
S7100B	172.18.35....	Available		7.6.1.44338
S7100C	172.18.35....	Available		7.6.1.44338
S8300Y	172.18.35....	Available		7.6.1.44338
S8300Z	172.18.35....	Available		7.6.1.44338
S9300Z Channel 1	172.18.35....	Available		7.6.1.44338
S9300Z Channel 2	172.18.35....	Available		7.6.1.44338

Packagers

The Packagers tab lists the packager nodes and information specific to each. Use the View by drop-down list to view the list of packagers by group or view all packagers.

Name	State	Alarm	Version
mlague-cmegrp1			2.6.1
Alka			2.6.1
cme-210	✓		2.6.1
hls_multi_aud	✓		
cme-22	⊖		2.6.1
Alka-old			2.6.1
2.5			2.5.1
<Unassigned>			

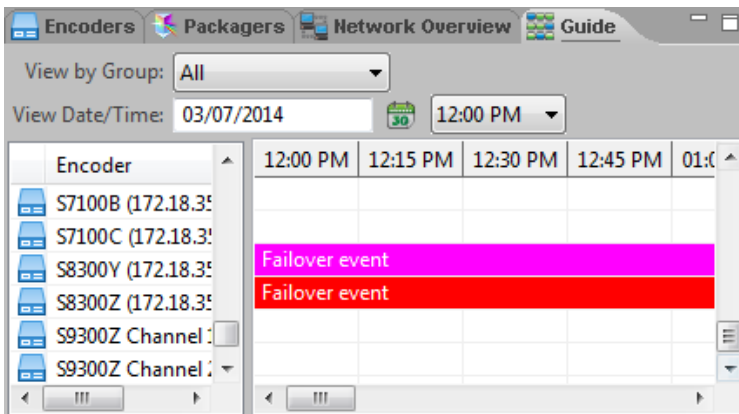
Network Overview

The Network Overview tab provides a pictorial view of the encoder nodes and their status. Use the View by drop-down list to restrict the list of encoders that are displayed.



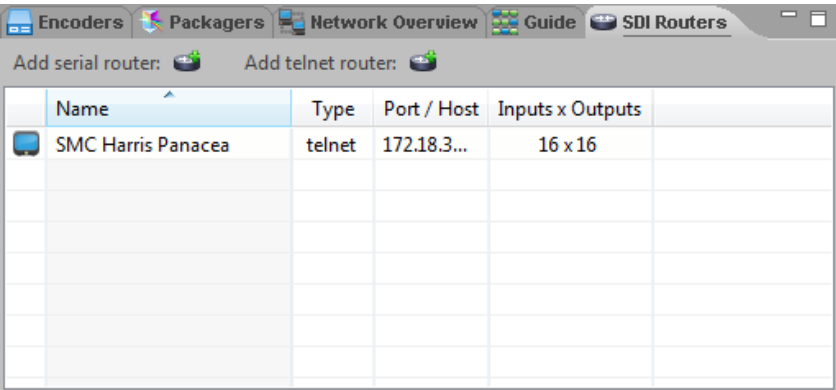
Guide

The Guide tab provides a pictorial view of the encoder node schedules.



SDI Routers

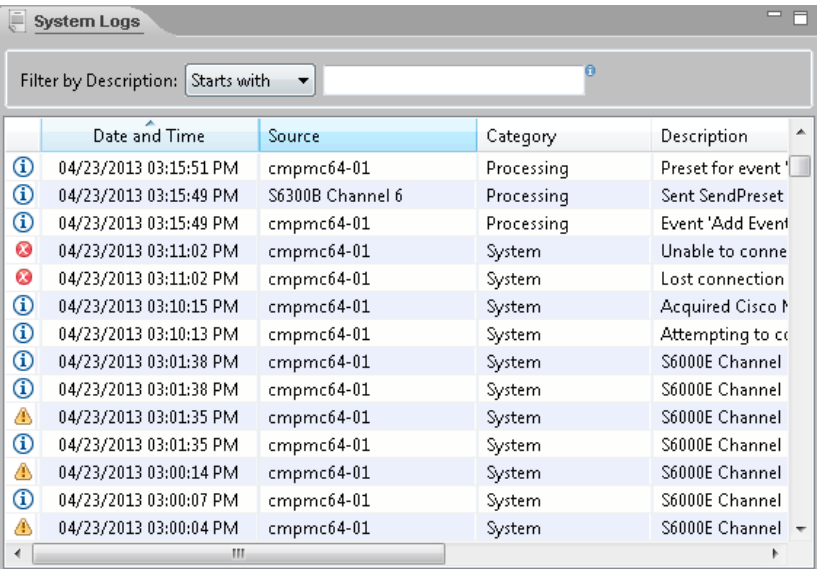
The SDI Routers tab lists the SDI routers and information specific to each.



Name	Type	Port / Host	Inputs x Outputs
SMC Harris Panacea	telnet	172.18.3...	16 x 16

System Logs

The System Logs tab displays information, warning, and error events for encoders and packagers managed by the LSM. The Filter by Description field provides several filtering options.



	Date and Time	Source	Category	Description
i	04/23/2013 03:15:51 PM	cmpmc64-01	Processing	Preset for event '...
i	04/23/2013 03:15:49 PM	S6300B Channel 6	Processing	Sent SendPreset...
i	04/23/2013 03:15:49 PM	cmpmc64-01	Processing	Event 'Add Event...
x	04/23/2013 03:11:02 PM	cmpmc64-01	System	Unable to connect...
x	04/23/2013 03:11:02 PM	cmpmc64-01	System	Lost connection...
i	04/23/2013 03:10:15 PM	cmpmc64-01	System	Acquired Cisco M...
i	04/23/2013 03:10:13 PM	cmpmc64-01	System	Attempting to co...
i	04/23/2013 03:01:38 PM	cmpmc64-01	System	S6000E Channel...
i	04/23/2013 03:01:38 PM	cmpmc64-01	System	S6000E Channel...
w	04/23/2013 03:01:35 PM	cmpmc64-01	System	S6000E Channel...
i	04/23/2013 03:01:35 PM	cmpmc64-01	System	S6000E Channel...
w	04/23/2013 03:00:14 PM	cmpmc64-01	System	S6000E Channel...
i	04/23/2013 03:00:07 PM	cmpmc64-01	System	S6000E Channel...
w	04/23/2013 03:00:04 PM	cmpmc64-01	System	S6000E Channel...

For more detailed information on monitoring, see the “Monitoring” chapter on page 218.

Menus

File Menu	
New	Creates a new Channel, event, or preset
Generate Sample	Generates a sample packager-only Channel lineup spreadsheet file
Import	Imports a Channel lineup, Channel templates, an encoder preset, an LSM Server Node license, or an encoder update file
Export	Exports a Channel lineup, Channel templates, or an LSM Server Node license
Exit	Exits the LSM application

User Menu	
Log In (CTRL+L)	Logs in to the LSM Server Node
Log Out	Logs out of the LSM Server Node
Change Password	Changes the user password
Manage Users	Adds, modifies, or removes users

Nodes Menu	
Get System Logs	Requests log files for specified encoders and/or LSM Server logs

Content Menu	
Manage Slates	Imports, previews, and deletes slates for use in slate events

View Menu	
AnyRes Live	Displays only tabs that configure and monitor AnyRes Live encoders
AnyRes Packaging	Displays only tabs that configure and monitor AnyRes Packagers
AnyRes Live and Packaging	Displays all configuration and monitoring tabs
Monitor	Displays panes within the Monitor tab
Configure	Displays panes within the Configure tab
Refresh (F5)	Refreshes the information in the LSM user interface
Restore Layout	Restores the original layout
Properties	Sets certain LSM Server Node or packager properties

Help Menu	
User Guide	Displays the <i>Cisco Linear Stream Manager User Guide</i> document
README	Displays the readme.txt file
EULA	Displays the End User License Agreement
About Cisco Linear Stream Manager	Views information about this version of the LSM

File Menu

New

Use the New option to choose a shortcut to create a new Channel, a new Live Streaming Event, or new Preset.

Choose **New > Channel**, or press **Ctrl-S**, to open the New Channel dialog box. For more information on Channels for encoders, see “Event Creation and Channels for Encoders” on page 191. For more information on Channels for packagers, see “Channels for Packagers” on page 210.

Choose **New > Live Streaming Event**, or press **Ctrl-E**, to open the New Live Streaming Event dialog box. For more information on event creation, see “New Streaming Event” on page 192.

Choose **New > Preset**, or press **Ctrl-P**, to open the Preset Editor dialog box. For more information on presets, see “Preset Editor” on page 33.

Generate Sample

Choose **Generate Sample** to generate a sample .xlsx spreadsheet as a guide for importing a packager-only Channel lineup. This file should be saved in a location with access to spreadsheet software that can edit an .xlsx file. Edit this file with the spreadsheet software to define your particular Channel lineup, and then use **Import > Channel Lineup** to create the Channel lineup.

Import

Choose **Import > Channel Lineup** to import a packager-only Channel lineup into the LSM database. For more information on the import procedure for Channels, see “Bulk Import of a Channel Lineup” on page 215.

Choose **Import > Channel Templates** to import a packager-only Channel Template XML file into the LSM database. Enter or browse to your XML file and click **Open**. New templates will be added to the LSM database, and templates that are not in use by a Channel will be updated. If a particular template in the XML file is currently in use by a Channel in the LSM, has a duplicate name in the file, or has an invalid definition, it will fail to import. See “Channel Templates for Packagers” on page 158.

Choose **Import > Preset** to import an encoder preset file from an encoder to the LSM. After importing, this preset is added to the Unassigned preset group. The imported preset can be edited or added to a group from the Encoder Presets tab.

Choose **Import > License** to import a new LSM Server Node license file. This file must have been previously received and placed in a location accessible from the LSM user

interface. After browsing to the license file and clicking **Import**, you must restart the LSM Server Node service by logging in to the server node locally or using remote desktop, then following these steps:

1. From the Windows Start Menu, enter “services” in the search field, or choose **Administrative Tools > Services**.
2. In the list of services, scroll down and right-click **Cisco LSM Server Node**.
3. Click **Restart**.

Choose **Import > Encoder Update File** to import a new file to the LSM that is used for update events. Enter or browse to your .exe file and click **Open**. The imported file replaces the current file and will be used for all scheduled update events.

Export

Choose **Export > Channel Lineup** to export a packager-only Channel lineup from the LSM database to a spreadsheet.

Choose **Export > Channel Templates** to export all Channel templates in the LSM database to an XML file. Browse to the file name and location for the exported template file and click **Save**.

Choose **Export > License** to export a license file. In the Export LSM Server Node License dialog box, enter or browse to the file name and location for the exported license data and save, and then click **Export**.

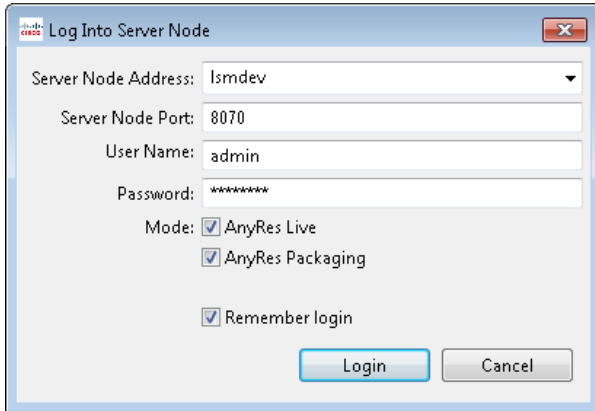
Exit

The Exit option will close the LSM client application.

User Menu

Log In

Log in from the User menu to manually log into the LSM Server.



The image shows a Windows-style dialog box titled "Log Into Server Node" with a Cisco logo in the top-left corner. The dialog contains the following fields and options:

- Server Node Address:** A dropdown menu with "lsmdev" selected.
- Server Node Port:** A text field containing "8070".
- User Name:** A text field containing "admin".
- Password:** A text field containing "*****".
- Mode:** A section with three checked checkboxes:
 - ☒ AnyRes Live
 - ☒ AnyRes Packaging
 - ☒ Remember login
- Buttons:** "Login" and "Cancel" buttons at the bottom right.

See "Logging In" on page 262 for details on logging in to the LSM server.

Automatic Connectivity Monitoring

If the LSM client to server communication is interrupted for longer than four seconds, the client will automatically be logged out from the LSM Server. Once a connection has been re-established, the LSM client will automatically reconnect using the last known good credentials.

Logout

Choose **Logout** from the User menu to log out of the LSM Server.

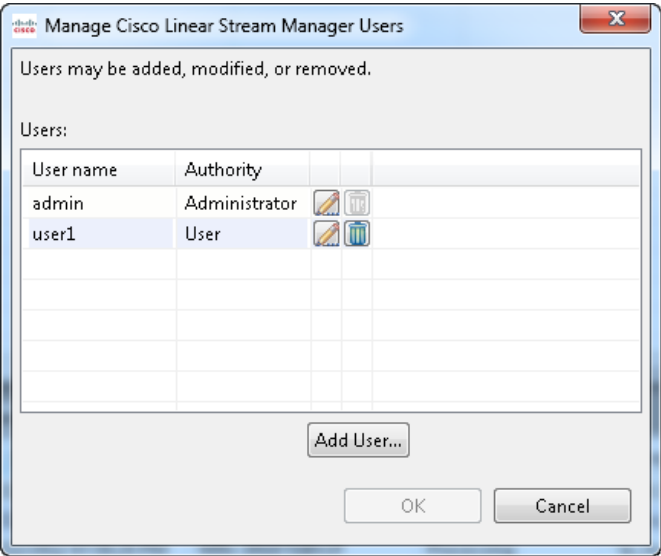
Change Password

Choose **Change Password** to display the Change Password dialog box.

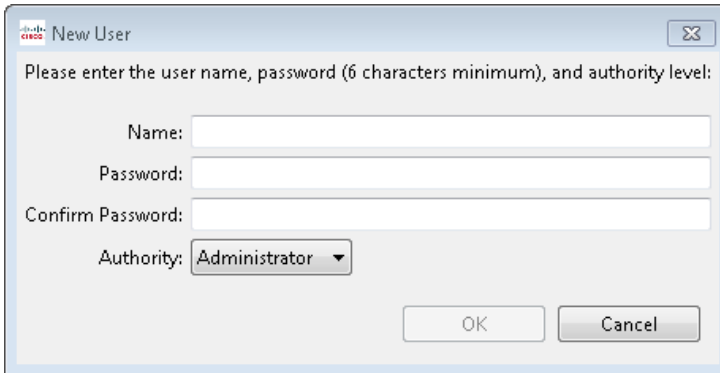


Manage Users

Choose **Manage Users** to add, modify, or remove users.



To add a user, click **Add User** to display the New User dialog box.

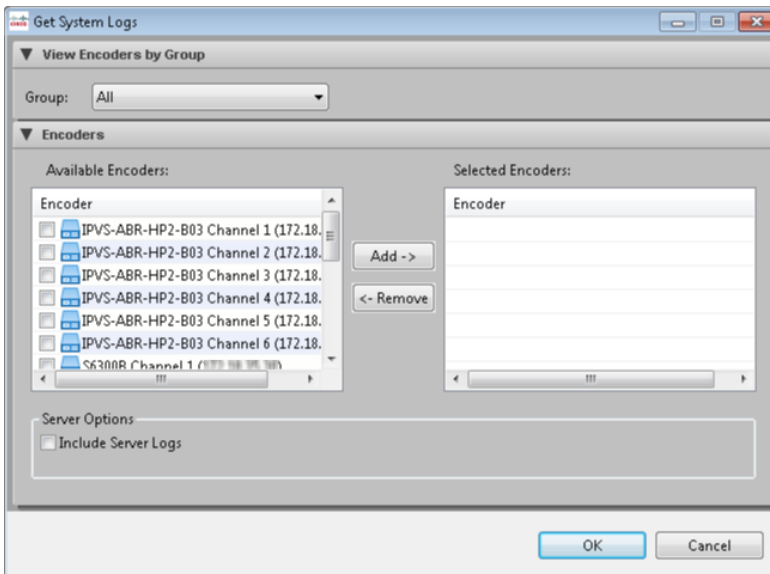


The 'New User' dialog box is a standard Windows-style window with a title bar containing the Cisco logo and the text 'New User'. It has a close button in the top right corner. The main text reads: 'Please enter the user name, password (6 characters minimum), and authority level:'. Below this text are four input fields: 'Name:', 'Password:', 'Confirm Password:', and 'Authority:'. The 'Authority' field is a dropdown menu currently showing 'Administrator'. At the bottom right are 'OK' and 'Cancel' buttons.

Nodes Menu

Get System Logs

Use the Get System Logs dialog box to request system logs for selected encoders.



The 'Get System Logs' dialog box is a standard Windows-style window with a title bar containing the Cisco logo and the text 'Get System Logs'. It has minimize, maximize, and close buttons in the top right corner. The main area is divided into several sections. At the top is a 'View Encoders by Group' section with a 'Group:' dropdown menu set to 'All'. Below this is an 'Encoders' section. It contains two list boxes: 'Available Encoders:' on the left and 'Selected Encoders:' on the right. The 'Available Encoders' list contains several entries, each with a checkbox and a description: 'IPVS-ABR-HP2-B03 Channel 1 (172.18.172.18.)', 'IPVS-ABR-HP2-B03 Channel 2 (172.18.172.18.)', 'IPVS-ABR-HP2-B03 Channel 3 (172.18.172.18.)', 'IPVS-ABR-HP2-B03 Channel 4 (172.18.172.18.)', 'IPVS-ABR-HP2-B03 Channel 5 (172.18.172.18.)', 'IPVS-ABR-HP2-B03 Channel 6 (172.18.172.18.)', and 'S6300R Channel 1 (172.18.172.18.)'. Between the two list boxes are 'Add ->' and '<- Remove' buttons. Below the list boxes is a 'Server Options' section with a checkbox labeled 'Include Server Logs'. At the bottom right are 'OK' and 'Cancel' buttons.

You can request logs for encoders, for the LSM Server, or for both.

For each encoder log to be requested, check the check box next to the encoder entry, and then click **Add** to include the log for that encoder.

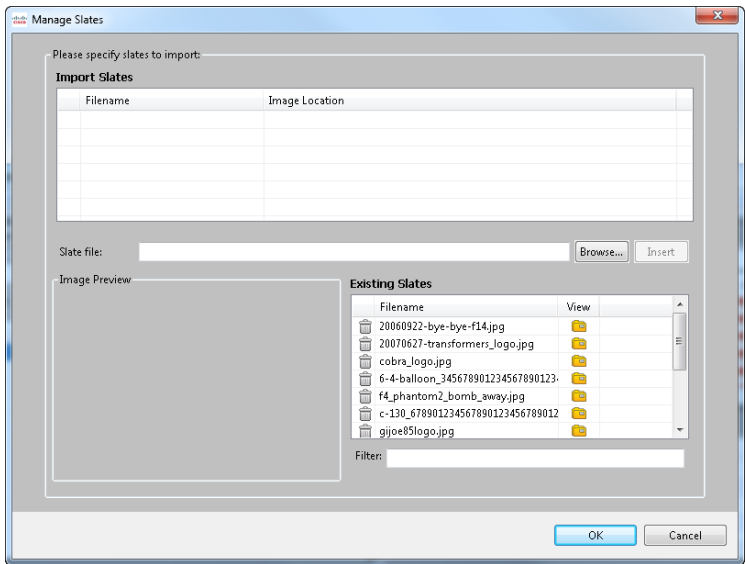
In the Server Options area, check the **Include Server Logs** check box if you are providing the LSM Server logs to Cisco support.

Click **OK** to submit the log request.

Content Menu

Manage Slates

Use the Manage Slates dialog box to manage slates.



The Manage Slates dialog box consists of two main areas, the Import Slates area and the Existing Slates area.

In the Import Slates area, browse the local file system, or any network share location, to find a valid image file. Valid image formats are files ending in extensions .jpg, .jpeg, .gif, .bmp, or .png. Note that no image validation apart from the file extension is performed. Once an image file has been specified, a preview of the image will appear in the Image Preview area of the dialog box. Next, click **Insert** to add the image file to the list of files to import. Multiple files can be selected to import at the same time. Next, click **OK** to begin the upload process in the background after the Manage Slates dialog box is closed. All slate files are stored on the LSM Server and downloaded to the encoder at the time of a slate mid event launched from the Manage Slate Events option on the streaming event context menu.

The Existing Slates image resource area provides a list of the image filenames currently uploaded to the LSM Server. Click the View folder icon for an image file to display a preview of that image in the Image Preview area.

View Menu

AnyRes Live

Choose **AnyRes Live** to display only tabs that configure and monitor AnyRes Live encoders.

AnyRes Packaging

Choose **AnyRes Packaging** to display only tabs that configure and monitor AnyRes Packagers.

AnyRes Live and Packaging

Choose **AnyRes Live and Packaging** to display all configuration and monitoring tabs.

Monitor

Choose **Monitor** to display panes within the Monitor tab.

Configure

Choose **Configure** to display panes within the Configure tab.

Refresh

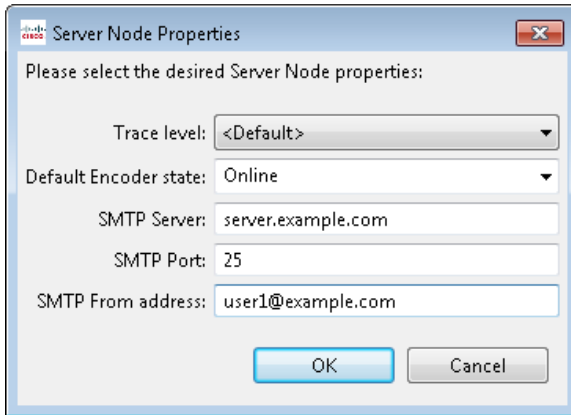
Refresh (F5) will re-sync all data with the LSM Server Node. This option may be used in case any data appears stale or out of date.

Restore Layout

The LSM remembers the layout of the user interface. The Restore Layout option reverses these changes and restores the size and location of the panes back to the original installation settings.

Properties

Choose **Properties > Server Node** to edit the LSM Server Node properties.



Server Node Properties

Please select the desired Server Node properties:

Trace level: <Default>

Default Encoder state: Online

SMTP Server: server.example.com

SMTP Port: 25

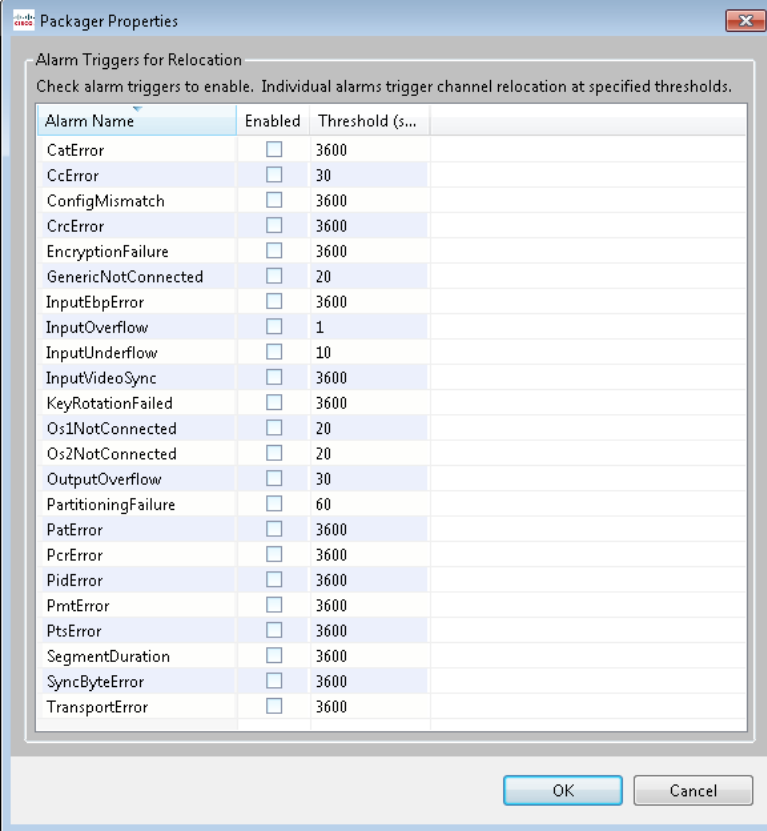
SMTP From address: user1@example.com

OK Cancel

For more information on the LSM Server Node property settings, see “Server Options” on page 267.

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Choose **Properties > Packager Properties** to set global properties for packagers.



The Packager Properties dialog box features a section titled "Alarm Triggers for Relocation" with the instruction: "Check alarm triggers to enable. Individual alarms trigger channel relocation at specified thresholds." Below this is a table with three columns: "Alarm Name", "Enabled", and "Threshold (s...)". The table lists 20 different alarm types, each with an unchecked checkbox and a default threshold value.

Alarm Name	Enabled	Threshold (s...)
CatError	<input type="checkbox"/>	3600
CcError	<input type="checkbox"/>	30
ConfigMismatch	<input type="checkbox"/>	3600
CrcError	<input type="checkbox"/>	3600
EncryptionFailure	<input type="checkbox"/>	3600
GenericNotConnected	<input type="checkbox"/>	20
InputEbpError	<input type="checkbox"/>	3600
InputOverflow	<input type="checkbox"/>	1
InputUnderflow	<input type="checkbox"/>	10
InputVideoSync	<input type="checkbox"/>	3600
KeyRotationFailed	<input type="checkbox"/>	3600
Os1NotConnected	<input type="checkbox"/>	20
Os2NotConnected	<input type="checkbox"/>	20
OutputOverflow	<input type="checkbox"/>	30
PartitioningFailure	<input type="checkbox"/>	60
PatError	<input type="checkbox"/>	3600
PcrError	<input type="checkbox"/>	3600
PidError	<input type="checkbox"/>	3600
PmtError	<input type="checkbox"/>	3600
PtsError	<input type="checkbox"/>	3600
SegmentDuration	<input type="checkbox"/>	3600
SyncByteError	<input type="checkbox"/>	3600
TransportError	<input type="checkbox"/>	3600

At the bottom of the dialog are "OK" and "Cancel" buttons.

For a description of the packager alarms enabled as triggers for channel relocation in this Packager Properties dialog box, see "Packager Alarms" on page 279.

When a packager alarm that has been enabled is triggered, the LSM will relocate the Channel to another packager. These alarms are global, therefore these triggers apply to all packagers managed by the LSM.

To enable an alarm trigger, check the check box next to the alarm name. The threshold is the number of seconds that the alarm condition must be active before Channel relocation action is taken. Default thresholds are displayed. To edit a threshold, click the value, and then modify it. You can enable any combination of triggers. Click **OK** when finished enabling alarm triggers.

Encoder Presets

3

Linear Service Templates

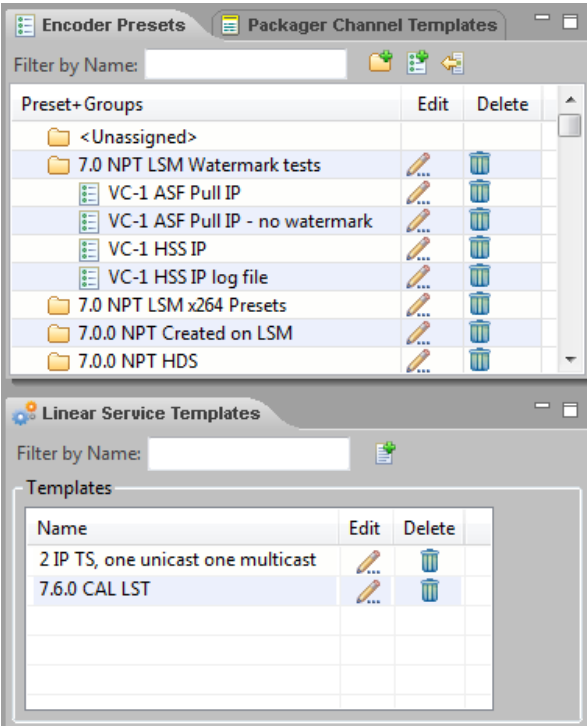
Presets and Linear Service

Templates for Encoders

The Encoder Presets and Linear Service Templates tabs are located on the Configure tab.

Presets are used to store encode settings, and Linear Service Templates (LSTs) associate one or more encoders with one or more presets. Several default presets and preset groups are included with the Cisco Linear Stream Manager (LSM). Preset groups can be used to organize presets for ease of use.

Enter text in the Filter by Name field to restrict the display of groups, presets, and templates to those that begin with a particular character string.



Preset Editor

To create a new preset, click the **New Preset**  icon to display the preset editor.

The preset editor is divided into nine areas of configuration, each with its own tab:

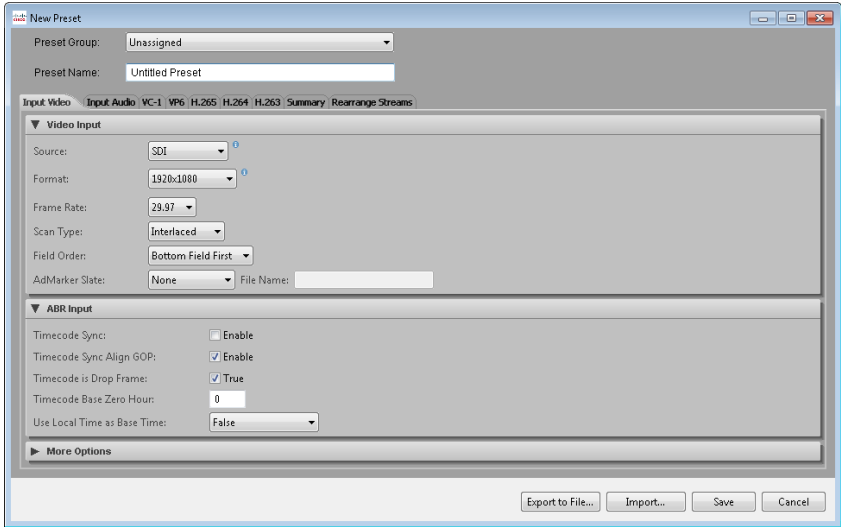
- Input Video—Provides Video input configuration fields.
- Input Audio—Provides Audio input configuration fields.
- VC-1—Provides Video, Audio, ASF Output, and HSS Output configuration fields for VC-1 streaming.
- VP6—Provides Video, Audio and Output Configurations for VP6 streaming.
- H.265—Provides Video, Audio, TS Output, HLS Output, and DASH Output Configurations for H.265 streaming.
- H.264—Provides Video, Audio, RTMP Output, TS Output, HLS Output, HSS Output, HDS Output, DASH Output, and RTP/3GPP Output Configurations for H.264 streaming.
- H.263—Provides Video, Audio, and RTP/3GPP Output Configurations for H.263 streaming.
- Summary—Provides the summary of specified encodings and its values.
- Rearrange Streams—Provides options to rearrange, copy, and delete streams.

At the top of the preset editor, optionally assign the preset to a preset group or modify the preset name.

The tab labels will change from black to green when that tab is enabled.

Each area within the tabs can be collapsed or expanded by clicking the down arrow next to the area label.

Input Video Tab



The video source chosen on the Input Video tab determines the available settings on the tab.

SDI, Component, Composite, S-Video, or HDMI Source

Source

Choose from the following video sources: SDI, Component, S-Video, Composite, and HDMI. The AnyRes 9300 only has an SDI source, and an HDMI source is only available for the AnyRes 8300. HD for the Cisco Media Processor AS7100/AS8100 is only supported via the SDI source.

Format/Frame Rate/Scan Type

Choose the format, frame rate, and scan type of your input video. HD for the AnyRes 5300 is only supported with the HD Upgrade Option. For the AnyRes 9300, all encode channels must have the same format, frame rate, and scan type, and these settings can only be modified on Encode Channel 1. For non-IP sources, the following formats are supported (frame rate in the table is *frame* not field rate).

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	Format	Frame Rate	Scan Type
HD	1920x1080	29.97/25	Interlaced
HD	1920x1080	25/23.976	Progressive
HD	1920x1080	HD 23.976	PsF
HD	1280x720	59.94	Progressive
NTSC	720x480/ 720x486	29.97	Interlaced
PAL	720x576	25	Interlaced

Field Order

By default, the captured video is assumed to be Bottom Field First. Change the field order by choosing Top Field First from the drop-down list.

AdMarker Slate

To substitute a black slate for input video when a marker is detected, choose **Black** from the drop-down list. To substitute a file, choose **File** and enter the slate file name. The File Name field must include the path to the existing slate file, either on AnyRes Live (recommended) or on a network share accessible by AnyRes Live.

ABR Input Area

▼ ABR Input

Timecode Sync:

☐ Enable

Timecode Sync Align GOP:

☒ Enable

Timecode is Drop Frame:

☒ True

Timecode Base Zero Hour:

Use Local Time as Base Time:

For ABR, view or modify the following Adaptive Streaming properties:

Timecode Sync

Check the **Enable** check box to start the Adaptive Streaming timeline based on the input timecode instead of 0. This option must be checked when distributing encodes across multiple encoders.

Timecode Sync Align GOP

Check the **Enable** check box to start the Adaptive Streaming timeline on a multiple of the GOP duration. This option must be checked when distributing encodes across multiple encoders.

Timecode is Drop Frame

If your NTSC timecode is not Drop Frame, then uncheck the **True** check box (this is unusual and not typical). The default is True.

Timecode Base Zero Hour

Timecode of 0 is assumed to be midnight. Enter a value in the Timecode Base Zero Hour field to indicate that Timecode 0 is not midnight. For example, entering 1 indicates that Timecode of 0 is 1:00 a.m. This setting is usually left at 0 even if your timecode has no relation to clock time.

Use Local Time as Base Time

If using a single system and not using timecode, then the time of samples in the encode stream defaults to start at 0. Optionally, use this field to set the base start time. You can choose to set the base start time to use the current system time measured since midnight today, since midnight on the first day of January 1970, or since midnight on the first day of the specified base year and month.

More Options Area

▼ More Options

SCTE-104:

☐ Enable

Line Number: 0

DTMF Duration Override:

☐ Enable

Use Input Channel: Input 1

Timecode Source:

Use Global Configuration on Encoder

Fail on Start if External Timecode Unavailable:

☐ Enable

Teletext Track Assignments

Track	Language	Magazine	Page	Line Number	Track ID	
1		0	0	0		Clear
2		0	0	0		Clear
3		0	0	0		Clear
4		0	0	0		Clear
5		0	0	0		Clear
6		0	0	0		Clear
7		0	0	0		Clear
8		0	0	0		Clear

SCTE-104

Check the **Enable** check box and enter a line number to capture SCTE-104 standalone splice requests, as well as splice requests or time signal requests containing a segmentation descriptor, and convert to SCTE-35 messages, which will then be passed through to the output if SCTE-35 is enabled. Splice request messages are supported on all outputs, but time signal and segmentation descriptor commands are only supported on TS outputs.

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Check the **Enable** check box and choose an input channel to allow an incoming DTMF duration tone to override the subsequent SCTE-104 splice insert message break duration. This feature is only used in certain specialized cases, and should not be enabled unless the input channel is expected to contain only these DTMF duration tones.

Timecode Source

Choose the desired timecode source from the drop-down list. The default value is Use Global Configuration on Encoder, which uses the Timecode Source setting on the Global Configurations page of the encoder user interface. Choose Integrated VITC to use integrated VITC timecode. Choose Integrated LTC to use integrated LTC timecode. Choose External LTC to use external timecode.

Fail on Start if External Timecode Unavailable

Check the **Enable** check box to prevent the encoder from starting if no external timecode source is provided by an external device.

For teletext input, you can configure up to 8 tracks. The combination of magazine, page, and line number for each track must be unique.

Language

Enter the language identifier for the track. The language identifier is a unique 3-letter value. By default, the language identifier will also be the track ID.

Magazine

Enter the number of the teletext magazine. The range is 0 to 7.

Page

Enter the number of the page in the specified magazine. The range is 0 to 99.

Line Number

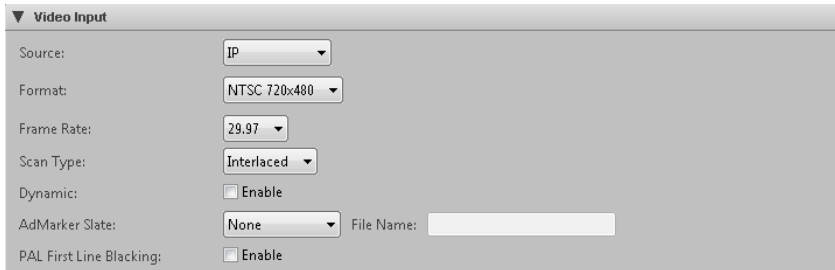
Enter the number of the line in the specified page. The range is 7 to 22 or 320 to 335.

Track ID

Enter the ID for the teletext track. The default track ID is the language identifier.

IP Source

Video Input Area



The screenshot shows a 'Video Input' configuration window. It contains several settings:

- Source: IP (selected in a dropdown)
- Format: NTSC 720x480 (selected in a dropdown)
- Frame Rate: 29.97 (selected in a dropdown)
- Scan Type: Interlaced (selected in a dropdown)
- Dynamic: ☐ Enable
- AdMarker Slate: None (selected in a dropdown)
- File Name: (empty text field)
- PAL First Line Blanking: ☐ Enable

Source

Choose **IP** from the Source drop-down list.

Format/Frame Rate/Scan Type

First choose the format of the input video from the drop-down list. The format will determine the options for frame rate and scan type. For IP source only, optionally specify a custom format by choosing Custom and entering a resolution. Next, choose the frame rate and scan type of the input video.

Dynamic

Check the **Enable** check box to dynamically adjust cropping, resizing, and deinterlacing operations to maintain the desired output resolution if the input source changes to be different from the initial values. The Aspect Ratio Control setting on the output Video tab specifies the method that is used to control how the output aspect ratio is maintained. If Dynamic is unchecked and the input differs from preset values, no automatic adjustment is done and the output may be incorrect. The default is unchecked.

AdMarker Slate

To substitute a black slate for input video when a marker is detected, choose Black from the drop-down list. To substitute a file, choose File and enter the slate file name. The File Name field must include the path to the existing slate file either on the AnyRes Live encoder (recommended) or a network share accessible by AnyRes Live.

PAL First Line Blanking

Check the **Enable** check box to black out line 23 video information for PAL sources. First line blanking is disabled when a video loss slate is active.

Input Streams Area

▼ Input Streams

Primary Source
Multicast Address: 0.0.0.0
Multicast Port: 0
Source Address (optional): 0.0.0.0

Secondary Source
Multicast Address: 0.0.0.0
Multicast Port: 0
Source Address (optional): 0.0.0.0

Video PID: 0

Audio Channel Assignments

Channel	PID	Audio Level
1	0	0 dB
2	0	0 dB
3	0	0 dB
4	0	0 dB
5	0	0 dB
6	0	0 dB
7	0	0 dB
8	0	0 dB

DPI (SCTE-35) PID: 0

ID3 Metadata PID: 0

Teletext Track Assignments

Track	PID	Language	Magazine	Page	Track ID	
1	0		0	0		Clear
2	0		0	0		Clear
3	0		0	0		Clear
4	0		0	0		Clear
5	0		0	0		Clear
6	0		0	0		Clear
7	0		0	0		Clear
8	0		0	0		Clear

DVB Subtitles
Burn-in Bitmap Subtitles: ☐ Enable PID: 16 Composition Page: 0 Ancillary Page: 0 Language:

For IP source, enter the multicast address, port, and optional source address that the encoder should use for IP input. Optionally, enter a secondary source carrying the same transport stream.

Video PID

Enter the PID for the video stream as a decimal number.

Audio PID

For each audio channel that is used, enter the PID for the audio stream as a decimal number. Enter a PID of 0 to disable ingest of audio.

Audio Level

Audio levels for all channels may be adjusted in a range from +60 dB to -60 dB for volume control. A dB change of 6 is a doubling of audio level. The default is 0, which means the audio level will not be adjusted.

DPI (SCTE-35) PID

For SCTE-35, enter the PID for the DPI (SCTE-35) stream as a decimal number to have SCTE-35 messages passed through. Enter a PID of 0 to disable ingest of SCTE-35.

ID3 Metadata PID

For ID3 Metadata, enter the PID containing the ID3 elementary stream. This PID is required for configuring HLS timed metadata pass-through.

Teletext Track Assignments

Enter the PID, language, magazine, page, and track ID for up to 8 teletext tracks. Track ID defaults to the language, or may be modified.

DVB Subtitles

Check the **Enable** check box to burn in the DVB Bitmap Subtitles from the subtitle track specified by the PID, Composition Page, optional Ancillary Page, and optional Language fields. If no Ancillary Page is specified, it is the same as the Composition page.

ESAM Area

▼ ESAM

Signaling URL:

☐ Enable

Out-of-Band Address:

☐ Enable

Interface: Management 1

Port: 50000

Acquisition Point Identity:

The ESAM area provides settings to configure Event Signaling and Management (ESAM). ESAM is used to prepare the output for Alternate Content Substitution (ACS). ACS allows downstream entities such as Manifest Manipulators to provide alternate content, such as targeted or zone-based advertising, blackouts, and the possibility of replacing national content with regionalized content.

To configure ESAM, you must enter a nonzero PID for SCTE-35 in the Input Streams area.

Signaling URL

Check **Enable** and enter the URL address of the destination Position Opportunity Information Server (POIS). The encoder will send SignalProcessingEvent messages to this URL.

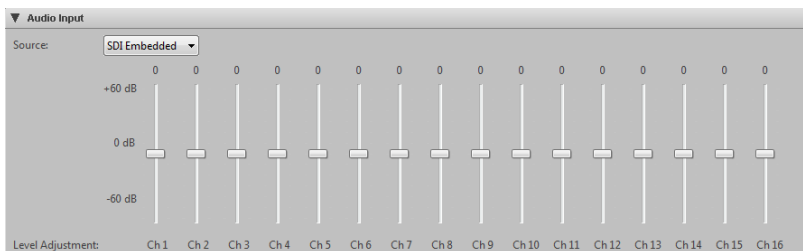
Out-of-Band Address

Check **Enable**, choose the Interface from the drop-down list, and enter the Port. The physical interface and port are used when the POIS sends SignalProcessingNotification messages to the encoder. The port must be unique on multiple encode channels of the same encoder.

Acquisition Point Identity

Enter the string which identifies the name of the channel or transcoder ID. This ID is typically configured on the POIS and is included in SignalProcessingEvent messages.

Input Audio Tab



For audio input, when the video source is SDI, balanced, AES and embedded audio are available. When the video source is any of the analog inputs, balanced and AES audio are available. When the video source is HDMI (8300 only), only HDMI audio is available. For IP source, audio input level is adjusted on the Input Video tab.

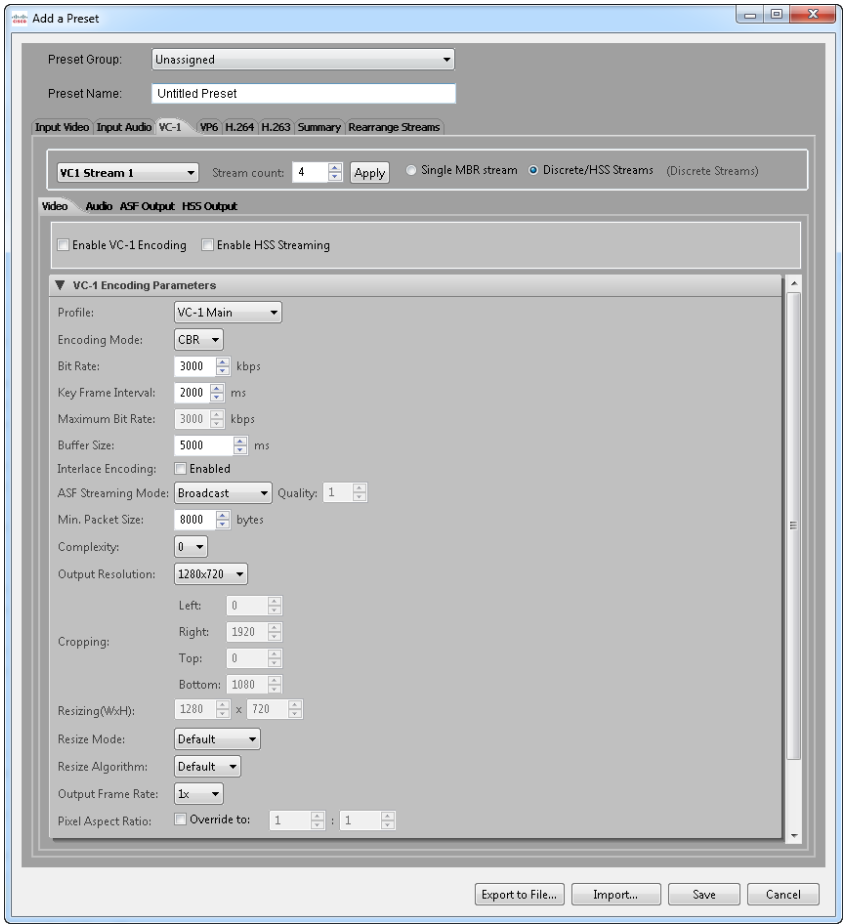
Source

Choose the audio source from the Source drop-down list. Available options are SDI Embedded, AES, or Balanced audio. For the 8300, the audio source option is HDMI if the video source is HDMI. Only SDI sources can use embedded audio. For SDI, eight input channels are available (16 for the 9300). For AES, balanced audio, or HDMI, two input channels are available.

Level Adjust

Use the sliders for each audio channel to adjust audio levels in a range from +60 dB to -60 dB for volume control. A dB change of 6 is a doubling of audio level. The default is 0, which means the audio level will not be adjusted.

VC-1 Tab



On the VC-1 tab, first enter the Stream count value and click **Apply**. You can choose to enable up to the number of VC-1 streams licensed on the encoder. Click the **Single MBR stream** radio button for these streams to be output within the same stream as the main stream (a single MBR stream), or click the **Discrete/HSS Streams** radio button for these streams to be discrete/HSS streams. Using the drop-down list at the top of the tab, choose each encode stream to view its settings.

Single stream MBR is a method of streaming where different bit rates of the same video are packaged together into a single stream. When playing back an MBR stream, a user will receive one of the substreams based on the network bandwidth. When encoding to an MBR Single Stream, each additional MBR stream should be composed of a lower

video/audio data rate than preceding streams. For example, if your main VC-1 encode is set up for 1 Mb video and 192 kb audio, your first additional MBR stream should have a lower video and audio rate (for example, 500 kb video and 128 kb audio). If you add another stream, its video and audio rates should be set up less than the previous one (for example, 100 kb video and 64 kb audio). If the primary stream has an audio stream, the child MBRs must also have audio streams. The user interface enforces this rule when setting up an MBR stream.

Separate streams can produce multiple independent streams of different bit rates. These streams are on separate ports, and are transported individually. Currently, all streams will share the same advanced settings of the main stream. When encoding to separate streams, it is not necessary for each successive stream to have a lower bit rate than its predecessor; the streams are independent and may be configured as needed.

VC-1 Video Tab

In the Video tab, check the **Enable VC-1 Encoding** check box. Also, if streams are discrete/HSS, you can optionally check the **Enable HSS Streaming** check box. If HSS Streaming is enabled, all VC-1 streams are HSS streams, and GOP parameters (key frame interval, GOP structure) are automatically set the same for all streams of a given HSS presentation. HSS output also requires Advanced Profile and no more than one audio stream with one audio track.

VC-1 Encoding Parameters Area

Profile

Profiles for encoding include VC-1 Advanced Profile, VC-1 Main Profile, and VC-1 Simple Profile. HSS output requires Advanced profile. Choose from the profiles available from the drop-down list.

Encoding Mode

Choose CBR (Constant Bit Rate) or VBR (Variable Bit Rate) encoding. If the mode is VBR, a field to enter Maximum Bit Rate in kbps appears and the Buffer Size field is dimmed. The VBR mode automatically sets the buffer size based on the target and peak bit rate.

Bit Rate

This setting indicates bits per second in units of kbps. For example, a value of 1500 is 1.5 Mbps.

Key Frame Interval

This value (in milliseconds) sets the maximum distance between key frames. The encoder may output key frames sooner than this interval (if it detects a scene

change, for example). If using HSS output, all streams must have the same key frame interval.

Buffer Size

The buffer size value (in milliseconds) sets the encoder buffer size. Two seconds (2000 milliseconds) is a typical buffer size.

Interlace Encoding

Check the **Enabled** check box to perform interlaced based encoding. If unchecked, the encoder will process video as progressive. Interlace encoding is only available for VC-1 Advanced Profile. VC-1 Main and Simple profiles do not support interlace encoding. Also, interlace encoding is not available when using HSS output. If you are not encoding as interlaced, then you should either choose to deinterlace your video (for video content) or perform an inverse telecine (IVT) on your video (for film-based content), using settings in the Pre-Processing area.

ASF Streaming Mode

Choose between Broadcast and Web for the ASF streaming mode. For Web mode, enter a value for Quality. The quality setting, from 0 to 100, is a tradeoff between smoother video (Quality 0) and better quality video (Quality 100). With a higher quality setting, the encoder may choose to drop more frames to achieve a target bit rate, versus keeping the frames but encoding them at a lower quality. If you prefer smoother video (less playback stutter), then choose a lower quality setting (25 for example). If you prefer slower motion video (more stutter) with higher quality, choose a high quality setting (80 for example).

For Broadcast streaming mode used with Main Profile, the encoder emphasizes smoother video over quality video, equivalent to Web streaming mode with Quality 0. In this mode, the encoder may drop (not encode) frames as needed. When Broadcast streaming mode is used with Advanced Profile, the encoder can encode skip frames for frames the Main Profile encoder would drop. This allows for the preservation of critical metadata, such as captioning and timecode.

Min. Packet Size

Enter the minimum size of the packets in the ASF stream. The actual packet size may be larger based on audio and video compression requirements. For streaming applications, a value no more than approximately 1/15th of your compressed video data *byte* rate is suggested. For example, 8000 is a good setting for a 1 Mbps stream.

Complexity

Enter a complexity setting from 0 to 4 to trade off between the needs of Better Performance (0) versus Better Quality (4) encoding. Encoder complexity tells the encoder how hard to work to gain maximum compression efficiency. For complex content, higher encoder complexities will yield better quality at the same data rate. Simple, static content will not show nearly as much of a difference. For Live

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Streams, we recommend a complexity setting of 2 or less. Setting complexity to 3 or 4 could result in dropped frames.

Output Resolution

Choose from the available resolutions or choose Custom, which makes the Cropping and Resizing settings available. The pixel aspect ratio is auto set for each of these presets.

Cropping

The cropping parameters apply a crop to the input image. Note that if an odd number of lines are cropped from the top, the sense of which field is first (top or bottom) will change and you must set the Field Order setting on the Input Video tab accordingly.

Resizing

Enter the output resolution to be applied to the cropped image. If the resolution is different than the original, scaling is performed. The minimum supported resolution is 64x64.

Resize Mode

Choose Progressive mode, Interlaced mode, or Single Field mode to manually specify how the resizer will process the source video. Single field mode, which should only be used for interlace source material, scales to the destination image using only a single field of the source video instead of performing a deinterlace operation.

Note: We highly recommend that you use single field mode for producing progressive frames from a 1080i source. For example, to stream a 1280x720p from a 1920x1080i source, we recommend that you choose the single field mode and resize to 1280x720 instead of performing a deinterlace operation.

Resize Algorithm

Choose Nearest, Linear, Cubic, or Super to choose the algorithm the resize operation uses. By default, Linear is used. Nearest (also referred to as Nearest neighbor) is the worst quality and has the lowest CPU requirements. Cubic is a higher quality scaler than Linear, and Super is a higher quality algorithm than Cubic. However, Super is very processor intensive and is only used for downscaling, and only when the resize is less than about half the source size. For example, if scaling single field mode from 1920x1080i, the field size is 1920x540, therefore, use Super if the output size is no more than 270 in height.

Output Frame Rate

Choose 1x to match the input frame rate, or reduce the input frame rate for telecine purposes or frame rate decimation. Available options listed are the input frame rate and 1/2, 1/3, 1/4, 1/5, and 1/6 the input rate. For example, if your input frame rate is 30 fps and you choose 1/2, then your output frame rate is 15 fps.

Pixel Aspect Ratio

To override the automatically set pixel aspect ratio, check the **Override** check box and modify the calculated ratio to the new ratio. If this check box is unchecked, the pixel aspect ratio is calculated based on the output size of the video for the preset sizes. Overriding is useful in cases where source video has a non-standard aspect ratio. For example, a 720x480 input source may have been produced anamorphically, and by indicating the aspect ratio here, players can correctly resize video on playback to the non-anamorphic size.

Aspect Ratio Control (IP Input only)

When Dynamic is enabled on the Input Video tab, choose the method that is used to maintain the output aspect ratio when the input resolution or aspect ratio changes during encoding. Manual specifies not to automatically adjust for the aspect ratio. Letter/Pillar will add letterboxing or pillarboxing to achieve the specified output aspect ratio. Crop will modify the crop settings to achieve the specified output aspect ratio. The default is currently set to Crop.

VC-1 Video Pre-Processing Area

▼ Pre-Processing

Interlacing Options:

None

Deinterlace Mode:

<default>

Noise Reduction Filter:

None

☐ Edge Enhance

Edge Threshold: 8

Median Filter:

☐ Enable

Watermark:

☐ Enable

File Name:

Browse...

Left: 0

Width: 0

Top: 0

Height: 0

Opacity: 100 %

☐ No Chroma

☐ Banner

Interlacing Options

If your video input is interlaced and your resolution height is greater than the field height, choose deinterlace to convert interlaced video to progressive. For resolutions less than or equal to the field height (320x240 for example for an NTSC source or 960x540 for HD 1080 sources), no deinterlacing is required and none should be specified.

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Both Main and Simple profiles *only* support progressive encoding and therefore, unless the resolution is less than or equal to the field height, deinterlacing should be chosen if not performing inverse telecine.

Choose inverse telecine (IVT) to convert film-based interlaced 30 fps video to progressive 24 fps. If IVT is specified and the encoding mode is interlaced, the encoder will produce IVT flags (top field first, bottom field first, repeat first field) in the bit stream; decoders will then know how to display the 24 fps progressive video on an interlaced display.

Deinterlace Mode

If deinterlacing is chosen, the encoder provides multiple methods for deinterlacing. The default, a Motion Adaptive deinterlace, attempts to preserve spatial information in areas of motion while removing interlace artifacts in areas of motion. The Advanced option provides a higher quality deinterlace than the other options, but also uses significantly more CPU cycles. The Blend option blends two fields, maintaining temporal information through motion blur. The Interpolate option removes temporal information and interpolates even fields from odd fields, while Interpolate Denoise applies a noise reduction filter after the deinterlace. The Line Double option simply creates even field lines as direct copies of the odd field lines. The CAVT option uses content adaptive vertical temporal filtering.

Noise Reduction Filter

You can choose between various noise reducing filters, or choose None for no filter. The Light filter uses less noticeable filtering that should have less impact on picture quality. The Normal filter provides for moderate noise reduction. The Smooth filter provides for large noise reduction at the expense of softer images.

If you choose one of the above filters, you can also check the **Edge Enhance** check box and enter a threshold to add the edge enhance mode to the filter. This mode detects edges in the image, and the detected edges are not processed by the filtering. The intent of this mode is to preserve edges and sharpness while still applying noise reduction to smoother areas. The edge threshold specifies the sensitivity of the edge detection. Lower values increase sensitivity, while higher values may detect few if any edges. The range is 8 to 128.

Median Filter

Check the **Enable** check box to use the Median filter. This filter is a standard image processing filter best used on noisy images.

Watermark

To add a watermark image, Check the **Enable** check box and browse or provide the path to the watermark image file. The File Name field must include the path to the existing image file either on the AnyRes Live encoder (recommended) or a network share accessible by AnyRes Live. The watermark image file may be of type .bmp, .jpg, .jpeg, .png, .tif, or .tiff. You can use the same image file for different streams.

Left and Top specify the pixel location where the upper left pixel of the watermark will be placed. Default Left 0, Top 0 is the upper left of the image. Enter Left -1, Top -1 to specify alignment to the bottom right location.

Width and Height cause the watermark image to be resized (enlarged or reduced) to be width pixels wide and height pixels high. Resizing is not supported for bitmaps that have an alpha channel. Both the original and resized watermark must be no larger than the encoded image size. Default Width 0, Height 0 specifies no resizing. You may want to use different resolution watermark images for different streams to preserve observed watermark size, or the same image could be scaled to different sizes.

Optionally, specify the opacity of the watermark. The default is 100% opaque.

Check the **No Chroma** check box to remove color information from the watermark image to display it in black and white.

Check the **Banner** check box to cause the watermark image to move two pixels per frame across the output image from right to left.

Advanced Compression Settings Area

▼ Advanced Compression Settings

Reset To Defaults

Filtering

In-loop Filter (Deblocking): <default>

Noise Filter: <default>

Median Filtering: <default>

Noise Edge Removal: <default>

Overlap Smoothing: <default>

Quantization

Adaptive Quantization: <default>

DQuant Option: <default>

P DQuant Strength: <default>

B DQuant Strength: <default>

B Frame Delta QP: <default>

GOP Structure

Number of B Frames: None

Variable GOP: <default>

Closed GOP: <default>

Miscellaneous

Letter Box Present: <default>

Key Pop Reduction: <default>

Lookahead (Flash Detect): <default>

CBR Lookahead: Off

No Drop Frame: <default>

Motion Estimation

MB Mode Cost: <default>

Match Method: <default>

Search Level: <default>

MV Range: <default>

MV Range Index: <default>

MV Cost: <default>

Video Encode Type: <default>

Number of Threads: <default>

Affinity Mask: 0

The Advanced Compression Settings area gives you control over many of the VC-1 encoding tools. This is important because, depending on your decoding solution (whether it be a mobile device, an STB or a PC), you may need to limit or want to expand on the tools used for encoding. Most options have a <default> setting where the actual setting used internally depends on multiple encoding setup factors, such as encode profile, resolution and bit rate. Click **Reset to Defaults** to reset all advanced encoding settings to their default value.

Filtering Options

In-Loop Deblocking

This setting enables an adaptive in-loop deblocking filter to help smooth the boundaries of encoded macroblocks when using an aggressive bit rate on complex material. This filter reduces blocking artifacts during encoding to improve the quality of P and B frames. It is not supported for the Simple profile.

Noise Filter

This setting enables a noise filter to attempt to detect and remove noise. It can improve the quality of noisy video sources, such as film containing visible grain or video that contains noise as a result of low-light conditions.

Median Filtering

This setting enables a median filter that improves motion estimation processing by factoring out noise artifacts. This can improve the quality of very noisy video and may reduce the encoded size, but may also introduce compression artifacts such as motion trails behind moving objects in an image.

Noise Edge Removal

This setting will attempt to detect noisy frame edges and remove them by duplicating adjacent lines to fill in the frame.

Overlap Smoothing

This setting helps to reduce blocking artifacts by smoothing the borders between adjacent macroblocks. This smoothing tends to make the image appear softer, but can improve the appearance of low bit rate video that contains many blocking artifacts.

GOP Structure

Number of B Frames

This setting specifies the number of B frames to use between other types of frames, up to 7. It is not supported for the Simple profile. The Lookahead (flash detect) option can also cause insertion of B frames even if the number of B frames set here is 0. To turn off all B frames, make sure that this parameter is set to 0 and

also set Lookahead (flash detect) to Off. HSS output requires that all streams have the same number of B frames.

Variable GOP

This setting (Group of Pictures), when set to Off, will force a key frame exactly at the maximum key frame distance interval. So, if the key frame interval is 2 seconds, you will get a key frame (I frame) at time 0, time 2, time 4, and so forth. In the default On mode, the forced placement of a key frame as defined by the key frame distance resets with each key frame inserted in the stream. For example, if the key frame interval is 2 seconds, but a scene change happens at time 1 that causes the output of a key frame, you will get a key frame at time 0, time 1, time 3, time 5, and so forth. HSS output requires fixed GOP.

Closed GOP

Choose **True** to specify that a GOP does not contain frames that depend on adjacent GOPs. HSS output requires Closed GOP.

Miscellaneous

Letterbox Present

Choose True to enable letterbox detection.

Key Pop Reduction

Choose On to use key pop reduction.

Lookahead (flash detect)

Choose On (available only in CBR encoding mode) for the encoder to evaluate future frames before encoding the current frame. Specifically, this mode looks for ideal places to insert B frames. This occurs when a single frame stands out as being very different from the frames around it. If this different frame were to be used for prediction, quality could suffer because there would be little to predict from. Therefore, the encoder may choose to change the frame type of this frame to B so that it will not be used for prediction. A camera flash is one common example, which is the basis for the name “flash detect.”

CBR Lookahead

Choose Mode 1 or Mode 2 (available only in CBR encoding mode) to enable lookahead rate control. This mode attempts to provide smoother/tighter CBR bit rates by analyzing future frames before encoding current frames. We recommend Mode 2 over Mode 1.

No Drop Frame

Choose True (available only in CBR encoding mode) to indicate that the encoder should not drop frames. In the cases where the encoder would normally drop frames, the encoder will output skip frames instead. This allows for the preservation

of metadata (such as timecode and embedded captions) while still allowing a high quality setting.

Quantization Properties

Adaptive Quantization

Choose the type of adaptive quantization to use. Higher values mean stronger quantization. Choose Adaptive Deadzone 1 to 15 for the conservative range, or Aggressive Adaptive Deadzone (ADZ) 1 to 5 for the strongest quantization.

DQuant Option

Choose **Off** to disable DQuant perceptual optimization. When this optimization is used, it can improve quality in smooth areas of video, at the cost of increased encoding computation. This option is not available when using Simple Profile. If this optimization is enabled, you can choose to optimize only I frames, only I and P frames, or I, P, and B frames.

P DQuant Strength and B DQuant Strength

If DQuant Optimization is enabled, these settings specify the strength of optimization to use for P and B frames respectively. Greater strength indicates that the encoding is more CPU-intensive.

B Frame Delta QP

This value, from 1 to 30, specifies the amount of the increase in QP for B-frames relative to the anchor frame QP on a per-macroblock basis. A higher QP value means a higher compression ratio. Increasing B-frame delta QP can sometimes result in better video quality because this can free up bits to better compress the key frames from which the B-frames are temporally predicted. This option is only available when using VC-1 Advanced Profile.

Motion Estimation Properties

MB Mode Cost

This setting specifies the cost method used by the codec to determine which macroblock mode to use. SAD/Hadamard only uses distortion to compute cost, while RD accounts for both rate and distortion in the computation.

Match Method

This setting specifies the method to use for motion searching. Choose SAD (best performance), Hadamard (best quality), or Macroblock-adaptive, which allows the codec to determine which method to use on each macroblock. This can potentially reduce overall computation required for encoding by performing the computationally-intensive Hadamard transform only when appropriate.

Search Level

This setting specifies which types of video information that motion search operations will use. Choose Luma only (best performance), Luma with nearest-integer chroma, Luma with true chroma (best quality), Macroblock-adaptive with true chroma, or Macroblock-adaptive with nearest-integer chroma.

MV Range

This setting specifies the range, in pixels, that will be used in motion searches. This option is not available when using Simple Profile. Choose from the horizontal (H) and vertical (V) ranges or Macroblock-adaptive.

MV Range Index

This setting specifies the method used to code the motion vector information in field pictures. This setting is only available when using VC-1 Advanced Profile. Choose encoder defaults, or choose to improve encoding efficiency for highly spread-out horizontal delta motion vector distributions, vertical distributions, or both.

MV Cost

This setting specifies the method used to estimate the cost (amount of processing needed) of motion vector coding. The codec uses the cost to determine which features will be used in encoding. Choose Static or Dynamic motion vector cost, which is varied between blocks to achieve optimal visual quality.

Video Encode Type

This setting specifies the method used to encode progressive or interlaced source video. This setting is only available when using VC-1 Advanced Profile. Choose from the following values.

Setting Value	Function
<default>	Default based on encoder setting
Progressive	Progressive Video
Interlace Frame	The encoder treats all source video frames as interlaced frames. This method is suitable for content that does not contain fast motion.
Interlace Field	The encoder treats each source video frame as two fields of interlaced video. This is usually the most efficient method, especially if the content contains fast motion.
Interlace Auto	The encoder automatically determines whether input video frames are interlaced frames or fields of

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	interlaced video. This method is suitable for content that does not contain fast motion.
Interlace/Progressive Auto	The encoder automatically determines the most efficient encoding method. This is the best method for content that contains a mixture of frame and field types

Number of Threads

This setting specifies 1, 2, 4, or 8 threads to use for encoding. This setting is intended to take advantage of multiple processors.

Affinity Mask

This setting specifies, in integer form, the hexadecimal bit mask that indicates which processors to use for the number of threads defined above. The affinity mask allows the user to run multiple concurrent encodes on different processors. Each bit in the affinity mask represents a processor. First determine the hexadecimal mask bits to be set, such as F for four threads on a four processor machine, and then convert the value to an integer for the setting. If the affinity mask is set to 0, the encoder will auto set the mask, such that the N least significant bits are set, where N is the number of threads assigned to the encode.

Captioning Area

▼ Captioning

Source: Line 21

608: ☐ CC1/CC2
☐ CC2/CC3

708: ☐ Service 1
☐ Service 2
☐ Service 3
☐ Service 4

SAMI Output: ☐ Enabled

Open Captions: ☐ Enabled (Note: Open Captioning applies to all output streams)

Script Output: ☐ Enabled (Note: Script Output applies to all Script-enabled ASF output streams)

Teletext Subtitles: ☐ Enabled

Teletext Mode: Word by Word ▼

Note: Closed Captions cannot be enabled with either Open Captions or Teletext Subtitles.

608

Check the appropriate check box(es) to process field 1 (CC1/CC2) and/or field 2 (CC3/CC4) captioning on the input source and to either have CEA-608 data inserted into the video elementary stream user data area (VC-1 Advanced Profile

only) or to enable SAMI output. At least one of these options must also be checked if you will be inserting captioning data into Script streams or HSS metadata.

708

For VC-1 Advanced Profile only, if you have checked the appropriate boxes under 608 to process field 1 (CC1/CC2) and field 2 (CC3/CC4) captioning on the input source, use the corresponding Service 1 through Service 4 check boxes to have up-converted CEA-708 data inserted into the video elementary stream user data area.

SAMI Output

If 608 Closed Captioning is enabled, check the **Enable** check box to generate a .smi output file in the same directory as the WMV file specified on the Output tab. To output SAMI, at least one of the 608 check boxes must be enabled.

Open Captions

When Open Captions is enabled, Line21 CC1 captions are decoded and rendered on top of the incoming video. Thus, unlike 608, 708 or SAMI captioning, the rendering of open captions is done on top of video in the encoder instead of in the decoder. This processing is independent of 608, 708, and SAMI captioning and none of these options need to be enabled for open captions.

Open captions requires NTSC 720x480 SD input sources.

Note: When open captions is enabled, it is enabled for all output streams, including any MBR, VP6 Flash, H.265, H.264, or H.263 streams.

Script Output

If you have enabled Script Streams, you can also enable Script Output for captioning. With this enabled, closed captions will be automatically inserted into all Script-enabled VC-1 ASF streams. For Script output, at least one of the 608 check boxes must be enabled.

Teletext Subtitles

If teletext subtitle data is present in the input video and the source is PAL, Check the **Enable** check box to insert the teletext subtitle data in the output stream. The data will be sent as metadata.

Teletext Mode

If Teletext Subtitles is enabled, choose the Mode. Line by Line mode tries to collect complete lines before sending the metadata. In this mode, any leading white space is removed from all the subtitle text. Word by Word mode sends out subtitle text as soon as it is decoded from the video frame.

VC-1 Audio Tab

The screenshot shows the 'VC-1 Audio Configuration' window. At the top, there are tabs: 'Video', 'Audio' (selected), 'ASF Output', and 'HSS Output'. Below the tabs is the 'VC-1 Audio Configuration' panel. It contains two track configurations:

- Track 1:**
 - ☒ Enabled
 - Output Format: WMA Pro
 - Bit Rate: 96 kbps
 - Sample Rate: 48000 Hz
 - Language: eng
 - ID: track_1
 - Offset (ms): 0
 - Left: Input 1
 - Right: Input 2
- Track 2:**
 - ☐ Enabled
 - Output Format: WMA
 - Bit Rate: 48 kbps
 - Sample Rate: 44100 Hz
 - Language: eng
 - ID: track_2

For VC-1 audio configuration, when the video source is SDI, audio source is embedded SDI, and discrete streams are specified, the encoder may optionally encode up to three WMA audio streams from an SDI source. These three streams are stereo tracks (unless defined as mono) and you can choose between any of the audio inputs for the left and right outputs of each track. Check the **Enabled** check box to enable encoding of a stereo track. These streams will be mutually exclusive from a player perspective, and the player will be able to choose the stream based on the language of the stream.

When the audio source is AES or Balanced, only a single stereo analog track may be encoded.

For HSS output, currently AnyRes Live only allows a single audio track per video stream. Therefore, if the multiple audio use case requires 3 audio tracks, at least 3 video streams are required.

Note: Currently, for HSS output, audio compression settings must be the same for all audio tracks. They must have the same bit rate, sample rate, complexity, offset and number of channels. The encoder web interface does not currently enforce this rule, so care must be taken to ensure this.

Output Format

Choose whether to encode audio using WMA, WMA Pro, or to pass through AC-3 audio.

Bit Rate

Choose from the list of available audio bit rates. The odd value bit rates (such as 191) indicate the use of the low delay WMA codec. If the output format is AC-3 pass-through, you must choose the bit rate of input AC-3 audio from the list of available rates.

Sample Rate

If the output format is AC-3 pass-through, you must choose the sample rate of input audio from the list of available rates, which varies based on the bit rate chosen above.

Language

Choose the ISO-639-2/T language label to apply to each stream.

ID

Enter the ID for the audio track. Each HSS audio track must have a unique ID. The client player should allow the user to choose between various audio IDs.

Offset

Enter an offset (in milliseconds) between audio and video. The offset can be a positive or negative value. Typically, this value is 0 unless you know that your source audio and video are not in sync. If not in sync, use this setting to adjust timing such that they are re-synced prior to encoding.

Left

Choose from the available audio inputs to specify the Left stereo output. For Mono audio, populate only the Left channel box and choose None for Right.

Right

Choose from the available audio inputs to specify the Right stereo output.

Input Channel (IP Input only)

For presets using IP input, choose from the available audio input channels to specify the audio stream to encode.

VC-1 ASF Output Tab

Video Audio **ASF Output** H264 Output

▼ ASF Network Push

☐ Enable ASF Push ☐ Auto Remove Publishing Point Stream 1 Values

Server Socket Address: (Server:Port)

Publishing Point:

Username:

Password: Password will not be displayed

ASF Network Push Area

Check the **Enable ASF Push** check box to push ASF content to a local network port on the encoder. This creates and starts publishing points on a Windows Media Server. You can also use an additional Auxiliary server.

Server Socket Address

Enter the name of the Windows Media server. This can be an alias, a fully qualified domain name, or an IP address of the Windows Media Server. The port is the UDP port that the stream should be sent on.

Publishing Point

Enter the name of the publishing point on the server through which the encoded stream will be broadcast. Check the **Auto Remove Publishing Point** check box to automatically remove the publishing point once the encoder is stopped.

If your Windows Media Server requires authentication, enter the username and password. The password characters will not be displayed.

ASF Network Pull Area

▼ ASF Network Pull

☐ Enable ASF Pull

Network Port:

Maximum Clients:

Check the **Enable ASF Pull** check box to enable Windows Media Servers and Windows Media Players to retrieve (“pull”) content as it is being encoded from the encoder UDP port.

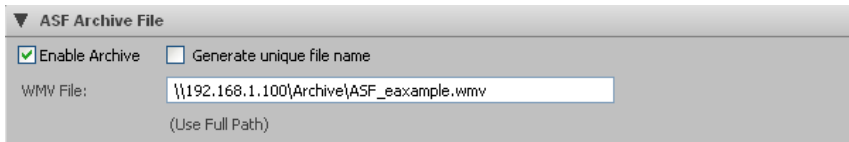
Network Port

Enter the UDP port that the Windows Media Server will pull the stream from.

Maximum Clients

Enter the maximum number of servers that can connect to the UDP port/stream.

ASF Archive File Area



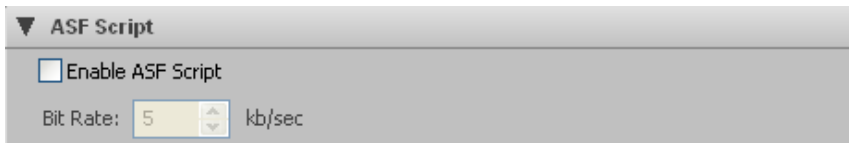
The screenshot shows a configuration window titled "ASF Archive File". It contains two checkboxes: "Enable Archive" (checked) and "Generate unique file name" (unchecked). Below these is a text field labeled "WMV File:" containing the path "\\192.168.1.100\Archive\ASF_example.wmv". A note below the text field says "(Use Full Path)".

The encoder can optionally save the ASF stream to disk. Check the **Enable Archive** check box and supply a path and file name to save to a user-specified WMV file. When using a network drive, a fully qualified UNC path is required and the AnyRes Live Encoding Service must have permission to create directories and copy files to the destination (for instructions see "Setting Up AnyRes Live to Write to a Network Drive" on page 265).

Note: If no path information is given, the file output is stored at C:\inetpub\wwwroot\encadmin\output on the encoder.

Check the **Generate unique file name** check box to append the current date and time to the file name, which will ensure that previous archive files will not be overwritten.

ASF Script Area



The screenshot shows a configuration window titled "ASF Script". It contains a checkbox labeled "Enable ASF Script" which is unchecked. Below this is a "Bit Rate:" label followed by a numeric input field containing the value "5" and a unit label "kb/sec".

The ASF Script area allows the use of an ASF script stream. An ASF script stream can deliver URL, text and caption data/commands to players. To have captions delivered, you must enable Script Output in the Captioning area of the Video tab. The script commands are delivered in presentation time order. Players such as the Windows Media Player can be optionally configured to process these commands as they encounter them.

The ASF script area may be used to independently enable an ASF script stream for each ASF stream by checking the appropriate box. Also, set the bit rate to be used for the script commands in kbps. A typical bit rate is 5.

ASF Attributes Area

▼ ASF Attributes

Title	<input type="text"/>
Author	<input type="text"/>
Copyright	<input type="text"/>
Description	<input type="text"/>

The ASF Attributes area defines information that is associated with the encoded media. These attributes are placed in the ASF header for Windows Media/VC-1 files, and players such as Windows Media Player display the attributes. You can edit the contents of the following attributes:

Title

Author

Copyright

Description

VC-1 HSS Output Tab

Video Audio ASF Output **HSS Output**

☐ Enable HSS Output Copy Primary HSS Stream Values

▼ Streaming to Origin Server

Server:	<input type="text"/>	
Publishing Point:	<input type="text"/>	
Stream Manifest File:	<input type="text"/>	Leave blank to have LSM auto-generate manifest
Username:	<input type="text"/>	
Password:	<input type="text"/>	Password will not be displayed
Publisher Retry Count:	<input type="text" value="5"/>	
Publisher Retry Delay:	<input type="text" value="20"/> sec	

▼ Streaming to Redundant Origin Server

Server:	<input type="text"/>	
Publishing Point:	<input type="text"/>	
Stream Manifest File:	<input type="text"/>	Leave blank to have LSM auto-generate manifest

Streaming to Origin Server Areas

Check the **Enable HSS Output** check box to enable VC-1 HSS output to an IIS Server with IIS Media Services installed. You can also use an additional redundant server. Click **Copy** to populate the stream settings from the settings of the primary stream.

Server

Enter the name of the IIS server. This can be an alias, a fully qualified domain name, or an IP address of the server. The default port is 80.

Publishing Point

Enter the stream ID to identify the video profile for this publishing point. The format for this field is: *publishingPointName.isml/stream(id)*.

Stream Manifest File

Enter the name and location of the stream manifest file to be used. Leave this field blank to have the LSM auto-generate the manifest file.

Username and Password

If your IIS Server requires authentication, enter the username and password. The password characters will not be displayed.

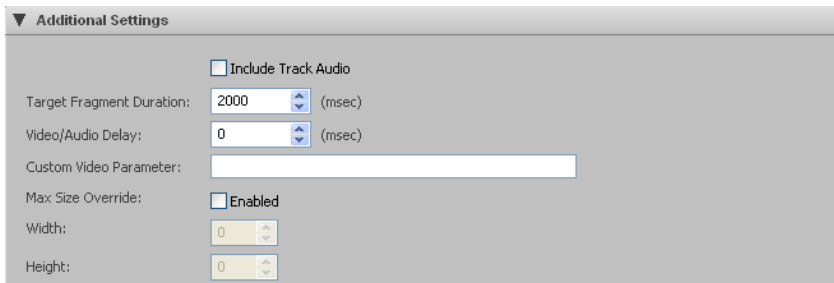
Publisher Retry Count

Set to the number of times the encoder will retry connecting to the configured server.

Publisher Retry Delay

Set to the delay in seconds between retry attempts.

Additional Settings Area



The screenshot shows a configuration window titled "Additional Settings" with a dropdown arrow on the left. The settings include:

- ☐ Include Track Audio
- Target Fragment Duration: 2000 (msec) with up/down arrows
- Video/Audio Delay: 0 (msec) with up/down arrows
- Custom Video Parameter: [Empty text box]
- Max Size Override: ☐ Enabled
- Width: 0 with up/down arrows
- Height: 0 with up/down arrows

Include Track Audio

Check the Include Track Audio box to include audio for a given encode stream in the HSS output. This check box only has meaning if audio has been enabled for the corresponding encode stream on the VC-1 Audio tab.

Target Fragment Duration

Enter (in milliseconds) the target duration for HSS fragments. This value should normally match the key frame interval value on the VC-1 Video tab. Although this

duration setting appears on the HSS output tab for each encode stream, there is only one global setting and it affects all VC-1 HSS streams.

Video/Audio Delay

Enter the delay in milliseconds for the output streams. Although this delay setting appears on the HSS output tab for each encode stream, there is only one global setting and it affects all VC-1 HSS streams.

Custom Video Parameter

The text in this field will be inserted as a custom attribute in the live server manifest. An example use for this field is for multiple camera angles. HSS streams are uniquely identified by a quality level, which is essentially the bit rate. Two streams of the same bit rate cannot be distinguished without custom attributes. This field provides the custom attribute to identify this stream. For multiple camera angles, this attribute could identify the specific video input.

An example format for this field is:

```
<param name ="cameraAngle" value="coach-cam" valuetype="data" />
```

Thus, two different encoders could provide two different camera angles. Each encoder has the same number of streams and bit rates for those streams. However, one encoder has for each stream, for example, a custom video parameter of:

```
<param name ="cameraAngle" value="main" valuetype="data" />
```

and the other encoder has for each stream a custom video parameter of:

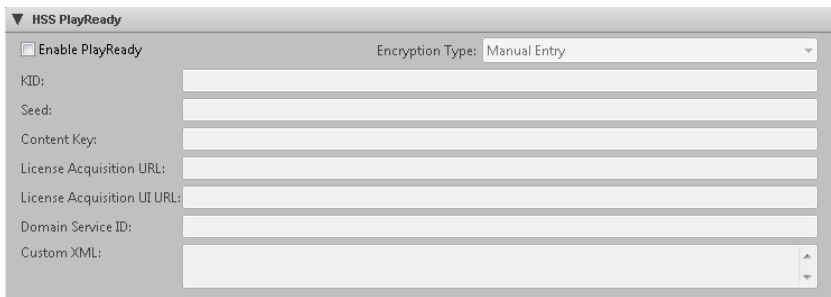
```
<param name ="cameraAngle" value="alternate" valuetype="data" />
```

Max Size Override

Check the **Enabled** check box to override the maximum output video size, and enter the new maximum width and height for this stream.

Use this setting to handle anamorphic/widescreen video. For example, 16x9 content is often anamorphically resized to 720x480 for delivery over SDI. Instead of resizing this 720x480 video back to 848x480 prior to compression, simply compress the content as is and set the max size override to 848x480. You should only need to do this to your top resolution stream. On playback, the Silverlight player will handle reversing the anamorphic processing and will display the content resized back to its original 16x9 848x480 size.

HSS PlayReady Area



PlayReady provides encryption support for HSS output.

Check the **Enable PlayReady** check box to use Smooth PlayReady. To complete the rest of the PlayReady setup you will need a third-party PlayReady License provider. This provider may provide you with setup information which you can manually enter in the fields provided. Some third-party PlayReady Platforms have been integrated to automatically retrieve the setup information. To use the third-party approach, you will also need a business relationship with the third party.

If you choose an Encryption Type for a third-party provider such as BuyDRM, additional settings will be available at the time of event creation.

If you choose Manual Entry, provide values for the following settings:

KID

Enter the Key ID. This value is required. The KID must be supplied as a GUID.

GUID should be entered in the form of: **{cab50618-c7e1-4e84-b683-c086d15170eb}**

Seed

The encryption seed. **Either** a seed or a content key is required.

Content Key

The encryption content key. **Either** a seed or a content key is required.

License Acquisition URL

Enter the URL. A License acquisition URL is required.

License Acquisition UI URL

This is an optional field. If supplied, enter the URL.

Domain Service ID

This is an optional field. If supplied, enter a Domain Service ID. The ID may be supplied as either a GUID or Base64 string.

GUID should be entered in the form of: **{cab50618-c7e1-4e84-b683-c086d15170eb}**

An example Base64 string is: **GAa1yuHHhE62g8CG0VFw6w==**

Custom XML

This is an optional field. The PlayReady license provider may specify custom data for the PlayReady header. If supplied use this custom XML field to include it.

HSS Metadata Area

The screenshot displays the 'HSS Metadata' configuration panel with three sections, each representing a different type of metadata track. Each section has a set of controls for enabling/disabling the track, naming it, and configuring its properties.

Track Type	Enabled	Name	Sparse track	Subtype	FourCC	Manifest output	Bit rate
User Track 1:	<input checked="" type="checkbox"/>	Metadata	<input checked="" type="checkbox"/>	DATA	plain text	<input type="checkbox"/>	1000
SCTE-35 Track:	<input checked="" type="checkbox"/>	AdMarkers	<input checked="" type="checkbox"/>	SCTE35	plain text	<input checked="" type="checkbox"/>	1000
CC/Subt... Track:	<input checked="" type="checkbox"/>	captions	<input type="checkbox"/>	CAPT	TTML/DFXP	<input type="checkbox"/>	1000

HSS offers the ability to have three metadata tracks that may be enabled independently. One track is available for user-supplied data, one track is used for Markers (SCTE-35) and the final track is used for Closed Captioning/Subtitles. For captions, a source for captions must be specified on the VC-1 Video tab. For subtitles, teletext subtitles must be enabled on the VC-1 Video tab.

All tracks are HSS "Text" tracks.

For each track, accept or modify the following settings:

Name

Each track can be given a name. For CC/Subtitles, there is special processing in the Silverlight Smooth Streaming Media Element (SSME) player for handling text tracks with the name captions, so we recommend leaving this track name as captions.

Sparse Track

The sparse track check box specifies that samples into these tracks do not happen at regular intervals. The User and SCTE-35 tracks are Sparse, and the CC/Subtitle track should not be Sparse. Currently, all Sparse tracks have "video" as their parent track.

Subtype

Accept or modify the pre-defined subtype that will be passed through to the manifest. The CC/Subtitle track should always be CAPT regardless if the text is captions or subtitles.

FourCC

The optional FourCC field is simply passed through to the manifest. For the CC/Subtitle track, FourCC should be set to TTML/DFXP for the encoder to generate DFXP-based metadata. This track may also be set to plain text. The default is TTML/DFXP.

Manifest Output

The manifest output check box specifies whether the metadata track samples are to be reflected in the manifest. For the CC/Subtitle track, this option generally should not be enabled unless some debugging is desired. With the Manifest Output checked, the sample data for the track is base64 encoded and included in the manifest. Over time this can create an excessively large manifest and thus in general it is best to not enable this option.

Bit rate

The Bit rate entry specifies an estimate of how much bandwidth in bits per second the metadata stream will use. If unknown, leave this at the default. The range is 1 to 10000.

VP6 Tab

Note: VP6 is not included with a standard AnyRes Live purchase. VP6 can be added to any AnyRes Live encoder as an additional add-on.

Conversely, if you create a streaming event and choose an AnyRes Live that is not enabled with VP6, the drop-down list to choose presets for the event will not include any VP6 presets.

VP6 Video Tab

The screenshot shows the 'VP6 Video Tab' interface. At the top, there are tabs for 'Video', 'Audio', and 'Output'. The 'VP6 Encoding Parameters' section is expanded, showing the following settings:

- ☐ VP6 Encoding Enabled
- Bit rate: 2000
- Key Frame Interval: 120
- Sharpness: 7
- Output Resolution: Custom
- Cropping:
 - Left: 0
 - Right: 1920
 - Top: 0
 - Bottom: 1080
- Resizing(WxH): 960 x 540
- Resize Mode: Default
- Resize Algorithm: Default
- Output Frame Rate: 1x

Below the 'VP6 Encoding Parameters' section are three expandable sections:

- Pre-Processing
- Advanced Compression Settings
- Captioning

In the VP6 Encoding Parameters area, you can enable VP6 Flash encoding, and then view or modify the following settings:

Bit Rate

This setting indicates bits per second in units of kilobits per second. For example, a value of 1500 is 1.5 Mbps.

Key Frame Interval

This value (in milliseconds) sets the maximum distance between key frames. The encoder may output key frames sooner than this interval (if it detects a scene change, for example).

Sharpness

This setting defaults to 7, or enter a setting between 0 and 10. A higher sharpness will result in a sharper image but may also result in more visible artifacts.

Output Resolution

Choose from the available resolutions or choose Custom, which will make the Cropping and Resizing settings available for modification. The pixel aspect ratio is auto set for each of these presets.

Cropping

The cropping parameters apply a crop to the input image. Note that if an odd number of lines are cropped from the top, the sense of which field is first (top or bottom) will change and you must set the Field Order setting on the Input Video tab accordingly.

Resizing

Enter the output resolution to be applied to the cropped image. If the resolution is different than the original, scaling will be performed. The minimum supported resolution is 64x64.

Resize Mode

Choose Progressive mode, Interlaced mode, or Single Field mode to manually specify how the resizer will process the source video. Single field mode, which should only be used for interlace source material, scales to the destination image using only a single field of the source video instead of performing a deinterlace operation.

Note: We highly recommend that you use single field mode for producing progressive frames from a 1080i source. For example, to stream a 1280x720p from a 1920x1080i source, we recommend that you choose the single field mode and resize to 1280x720 instead of performing a deinterlace operation.

Resize Algorithm

Choose Nearest, Linear, Cubic, or Super to specify the algorithm to choose the algorithm the resize operation uses. By default, Linear is used. Nearest (also referred to as Nearest neighbor) is the worst quality and has the lowest CPU requirements. Cubic is a higher quality scaler than Linear, and Super is a higher quality algorithm than Cubic. However, Super is very processor intensive and is only used for downscaling, and only when the resize is less than about ½ the source size. For example, if scaling single field mode from 1920x1080i, the field size is 1920x540, so use Super if the output size is no more than 270 in height.

Output Frame Rate

Choose 1x to match the input frame rate, or reduce the input frame rate for telecine purposes or frame rate decimation. Available options listed are the input frame rate and 1/2, 1/3, 1/4, 1/5, and 1/6 the input rate. For example, if your input frame rate is 30 fps and you choose 1/2, then your output frame rate will be 15 fps.

VP6 Pre-Processing Area

Interlacing Options

If your video input is interlaced *and* your resolution height is greater than the field height, choose deinterlace to convert interlaced video to progressive. For resolutions less than or equal to the field height (320x240 for example for an NTSC source or 960x540 for HD 1080 sources), no deinterlacing is required and none should be specified.

Choose inverse telecine (IVT) to convert film-based interlaced 30 fps video to progressive 24 fps. If IVT is specified and the encoding mode is interlaced, the encoder will produce IVT flags (top field first, bottom field first, repeat first field) in the bitstream; decoders will then know how to display the 24 fps progressive video on an interlaced display.

Deinterlace Mode

If deinterlacing is chosen, the encoder provides multiple methods for deinterlacing. The default, a motion adaptive deinterlace, attempts to preserve spatial information in areas of motion while removing interlace artifacts in areas of motion. The Advanced option provides a higher quality deinterlace than the other options, but also uses significantly more CPU cycles. The Blend option blends two fields, maintaining temporal information through motion blur. The interpolate mode removes temporal information and interpolates even fields from odd fields, while interpolate denoise applies a noise reduction filter after the deinterlace. The line double option simply creates even field lines as direct copies of the odd field lines.

Miscellaneous Filters

The encoder provides for various noise reducing filters. The normal filter provides for moderate noise reduction. The smooth filter provides for large noise reduction at the expense of softer images.

Watermark

To add a watermark image, Check the **Enable** check box and browse or provide the path to the watermark image file. The File Name field must include the path to the existing image file either on the AnyRes Live encoder (recommended) or a network share accessible by AnyRes Live. The watermark image file may be of type .bmp, .jpg, .jpeg, .png, .tif, or .tiff. You can use the same image file for different streams.

Left and Top specify the pixel location where the upper left pixel of the watermark will be placed. Default Left 0, Top 0 is the upper left of the image. Enter Left -1, Top -1 to specify alignment to the bottom right location.

Width and Height cause the watermark image to be resized (enlarged or reduced) to be width pixels wide and height pixels high. Resizing is not supported for bitmaps that have an alpha channel. Both the original and resized watermark must be no larger than the encoded image size. Default Width 0, Height 0 specifies no resizing. You may want to use different resolution watermark images for different streams to preserve observed watermark size, or the same image could be scaled to different sizes.

Optionally, specify the opacity of the watermark. The default is 100% opaque.

Check the **No Chroma** check box to remove color information from the watermark image to display it in black and white.

Check the **Banner** check box to cause the watermark image to move two pixels per frame across the output image from right to left.

Advanced Compression Settings Area

▼ Advanced Compression Settings

Quantizer Maximum:	45 (0 to 63)	RealTime Mode Speed:	7
Key Frame Data Target:	250 \backslash (width+height)/4	Noise Sensitivity:	0
Auto key Frame:	<input checked="" type="checkbox"/> Enabled	Start Buffer Level(ms):	500
Auto key Frame Threshold:	70 (0 to 100)	Optimal Buffer Level(ms):	500
Minimum Key Frame Distance:	30	Max Buffer Level(ms):	500
Allow Drop Frames:	<input type="checkbox"/> Enabled	Undershoot Percent:	95
Drop frame Watermark:	20 (0 to 100)		

The advanced compression settings area gives you control over many of the VP6 Flash encoding tools. This is important because, depending on your decoding solution

(whether it be a mobile device, an STB or a PC), you may need to limit or want to expand on the tools used for encoding.

Quantizer Maximum

This setting is the worst frame quality allowed (higher is worse), and ranges from 0 to 63 with a default of 45. Lower numbers may lead to dropped frames during compression to achieve the target bit rate if the allow drop frames option is enabled. The combination of quantizer maximum and allow drop frames below is equivalent to the notion of the smoothness/quality tradeoff of VC-1.

Key Frame Data Target

This setting is the bit rate in Kbps to use when compressing key frames.

Auto Key Frame

Check the **Enabled** check box to enable the encoder to determine where to place key frames.

Auto Key Frame Threshold

This setting specifies the threshold of how different frames must be before a key frame is generated. A higher number means fewer key frames. The range is 0 to 100 with a default of 70.

Minimum Key Frame Distance

This setting is the minimum number of frames allowed between key frames. The default is 8 frames.

Allow Drop Frames

Check the **Enabled** check box to allow the encoder to drop frames to achieve the desired bit rate. If this box is unchecked, the encoder will not drop frames and may output a higher bit rate than desired.

Drop Frame Watermark

This setting is the percentage of the buffer below which the encoder will start dropping frames. The default is 20%.

RealTime Mode Speed

This setting determines what proportion of the available processor cycles the encoder will use to compress the video. At speed 0, the encoder will use all available cycles; at 8, it will use half; at 16, it will use none. The default is 7.

Noise Sensitivity

This setting is the level of noise pre-processing to apply, with 0 being no preprocessing and 6 being the highest level of temporal pre-processing. The default is 0.

Start Buffer Level

This setting is the amount of data in milliseconds preloaded by the media player before starting playback. The range is 0 to 300000, with a default of 120.

Optimal Buffer Level

This setting is the buffer size the encoder will attempt to reach and maintain, in milliseconds. The range is 0 to 300000, with a default of 200.

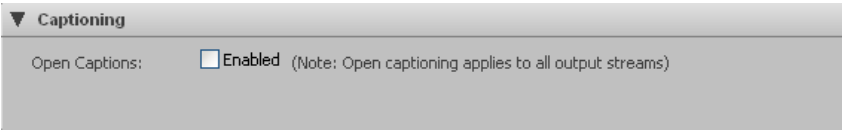
Max Buffer Level

This setting determines the maximum size of the buffer, in milliseconds. The range is 0 to 300000, with a default of 200.

Undershoot Percent

This setting is the percentage of the specified bit rate the encoder will target. Targeting a slightly lower bit rate results in bits being available in the buffer to improve difficult sections. The default is 95%.

Captioning Area

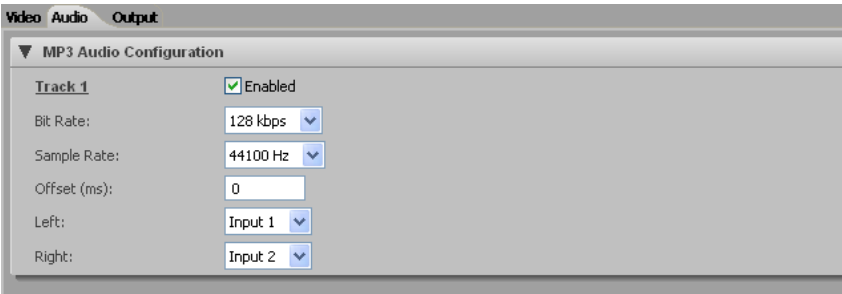


Open Captions

When Open Captions is enabled, Line21 CC1 captions are decoded and rendered on top of the incoming video.

Note: When open captions is enabled, it is enabled for all output streams, including any VC-1, H.265, H.264, or H.263 streams.

VP6 Audio Tab



Check the Enabled box to enable MP3 encoding of a stereo channel with the following parameters:

Bit Rate

Choose from the list of available audio bit rates in kbps.

Sample Rate

Specify the sample rate of input audio from the list of available rates in hertz, which varies based on the bit rate chosen above.

Offset

Enter an offset (in milliseconds) between audio and video. The offset can be a positive or negative value. Typically, this value is 0 unless you know that your source audio and video are not in sync. If not in sync, use this setting to adjust timing such that they are re-synced prior to encoding.

Left

Choose from the available audio inputs to specify the Left stereo output. For Mono audio, populate only the Left channel box and choose None for Right.

Right

Choose from the available audio inputs to specify the Right stereo output.

VP6 Output Tab

The screenshot shows the 'VP6 Output Tab' configuration window. It has three tabs: 'Video', 'Audio', and 'Output', with 'Output' being the active tab. The window is divided into two main sections. The top section is titled 'Streaming To Flash Media Server' and contains the following controls: an 'Enable Streaming' checkbox (unchecked), a 'Server Location URL:' text box with the value 'rtmp://192.168.1.31/live', a 'Stream Name:' text box with the value 'vp6stream', an 'Enable Authentication For' checkbox (unchecked) with a dropdown menu showing 'Akamai', and 'Username' and 'Password' text boxes. A note 'Password will not be displayed' is next to the password box. The bottom section is titled 'FLV Archive File' and contains: a 'Stream: Main' checkbox (checked) and a 'Generate unique file name' checkbox (unchecked), and an 'FLV File:' text box with a placeholder '(Use Full Path)'.

Streaming to Flash Media Server Area

Check the **Enable Streaming** check box to push content to a Flash Media server.

Server Location URL

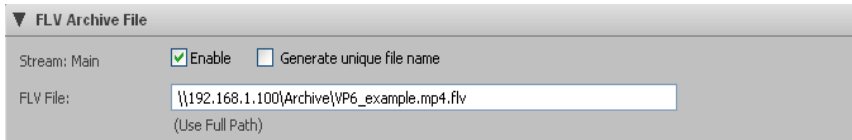
Enter the URL for the Flash Media server with the format of rtmp://serverip/application-name.

Stream Name

Enter the name of the stream.

Check the **Enable Authentication** check box and choose the CDN (content distribution network) from the drop-down list to enable authentication with a Flash Media Server before the stream can be published. Enter the username and password. The password characters will not be displayed.

FLV Archive File Area



▼ FLV Archive File

Stream: Main ☒ Enable ☐ Generate unique file name

FLV File:
(Use Full Path)

The encoder can optionally save the VP6 Flash stream to disk. Check the **Enable** check box and supply a path and file name to save to a user-specified FLV file.

When using a network drive, a fully qualified UNC path is required and the AnyRes Live Encoding Service must have permission to create directories and copy files to the destination (for instructions see “Setting Up AnyRes Live to Write to a Network Drive” on page 265).

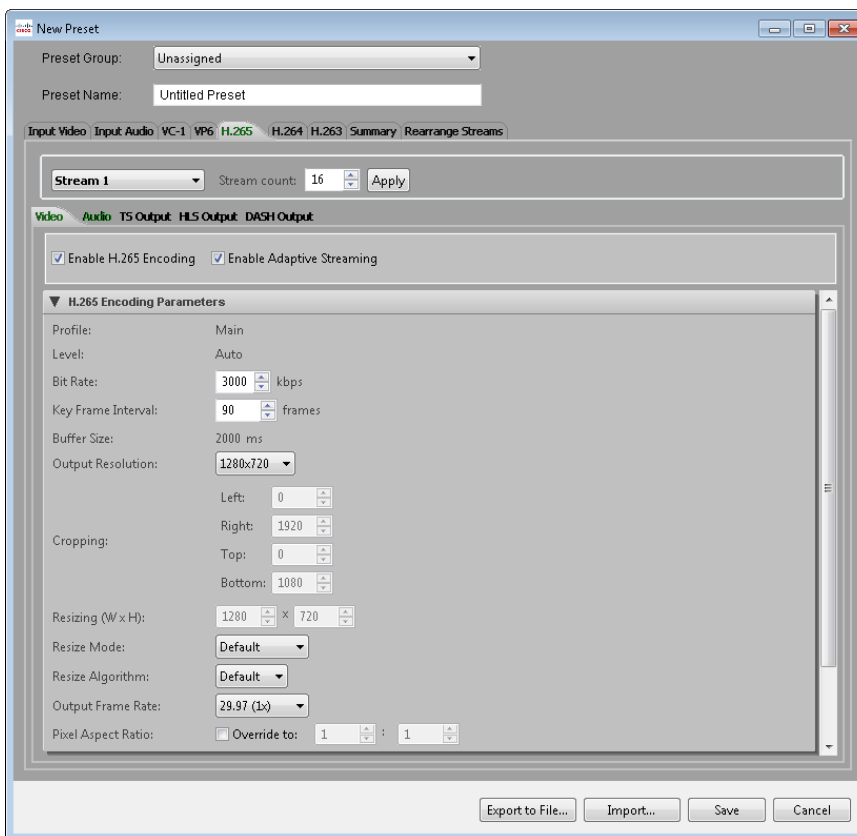
Note: If no path information is given, the file output is stored at C:\inetpub\wwwroot\encadmin\output on the encoder.

Check the **Generate unique file name** check box to append the current date and time to the file name, which will ensure that previous archive files will not be overwritten.

H.265 Tab

Note: H.265 and H.264 outputs cannot be enabled simultaneously in the same preset. If you enable H.265 output while H.264 output is enabled in the preset, H.264 will be automatically disabled.

H.265 Video Tab



In the H.265 tab, first enter the Stream count value and click **Apply**. In the Video tab, enable H.265 encoding, and optionally enable Adaptive Streaming for one or more streams. Adaptive Streaming forces subsequent streams to have the same values for Key Frame Interval, and also for the Number of B Frames advanced setting. To change these settings on other streams, you must check the **Unlocked** check box that appears on stream 2 and above when Adaptive Streaming is enabled. Finally, view or modify the settings in each area of the Video tab for each stream.

H.265 Encoding Parameters Area

Profile

Display only. Only Main profile is supported.

Level

Display only. Only Auto is supported.

Bit Rate

This setting indicates bits per second in units of kilobits per second. For example, a value of 1500 is 1.5 Mbps. The valid range depends on frame rate and output size. The default is 3000 kbps. The bit rate must be unique for each H.265 output stream.

Key Frame Interval

This value (in frames) sets the maximum distance between key frames. The default is 60 frames for non-PAL video and 50 frames for PAL video, which is 2 seconds of video. Adaptive Streaming requires this interval to be the same for all adaptive streams, and should be a multiple of the frame rate. Segment Duration and Fragment Duration settings on the TS Output tab must each be a multiple of the key frame interval.

Buffer Size

Display only. The buffer size value (in milliseconds) sets the encoder buffer size. AnyRes Live uses a buffer size of 2000 ms, and this value is not configurable.

Output Resolution

Choose from the available resolutions or choose Custom, which will make the Cropping and Resizing settings available for modification. The pixel aspect ratio is auto set for each of these presets.

Cropping

The cropping parameters apply a crop to the input image. Note that if an odd number of lines are cropped from the top, the sense of which field is first (top or bottom) will change and you must set the Field Order setting on the Input Video tab accordingly.

Resizing

Specify the output resolution to be applied to the cropped image. If the resolution is different than the original, scaling will be performed. The minimum supported resolution is 64x64.

Resize Mode

Choose Progressive mode, Interlaced mode, or Single Field mode to manually specify how the resizer will process the source video. Single field mode, which

should only be used for interlace source material, scales to the destination image using only a single field of the source video instead of performing a deinterlace operation. We highly recommend that you use single field mode for producing progressive frames from a 1080i source. For example, to stream 1280x720p from a 1920x1080i source, we recommend that you choose the single field mode and resize to 1280x720 instead of performing a deinterlace operation.

Resize Algorithm

Choose Nearest, Linear, Cubic, or Super choose the algorithm the resize operation uses. By default, Linear is used. Nearest (also referred to as Nearest neighbor) is the worst quality and has the lowest CPU requirements. Cubic is a higher quality scaler than linear, and Super is a higher quality algorithm than Cubic. However, Super is very processor intensive and is only used for downscaling, and only when the resize is less than approximately half the source size. For example, if scaling single field mode from 1920x1080i, the field size is 1920x540, therefore, use Super if the output size is no more than 270 in height.

Output Frame Rate

Choose from the available frame rate options. For output heights of 720 or less, you can choose the 2x frame rate to encode each input field of interlaced source as a progressive picture, which converts 1080i sources to 720p outputs. Choose 1x to use the input frame rate. Choose fractional frame rates to reduce the input frame rate for telecine purposes or frame rate decimation. Available fractional frame rate options are 1/2, 1/3, 1/4, 1/5, and 1/6 the input rate.

Pixel Aspect Ratio

To override the automatically set pixel aspect ratio, check the **Override** check box and modify the calculated ratio to the new ratio. If this check box is unchecked, the pixel aspect ratio is calculated based on the output size of the video for the preset sizes. Overriding is useful in cases where source video has a non-standard aspect ratio. For example, a 720x480 input source may have been produced anamorphically, and by indicating the aspect ratio here players can correctly resize video on playback to the non-anamorphic size.

Aspect Ratio Control (IP Input only)

When Dynamic is enabled on the Input Video tab, choose the method that is used to maintain the output aspect ratio when the input resolution or aspect ratio changes during encoding. Manual specifies not to automatically adjust for the aspect ratio. Letter/Pillar will add letterboxing or pillarboxing to achieve the specified output aspect ratio. Crop will modify the crop settings to achieve the specified output aspect ratio. The default is currently set to Crop.

Active Format Description (AFD) (IP Input Only)

Check the **Enable** check box to pass the Active Format Description through to H.264 user data. Also, choose from the drop-down list to specify whether the encoder should not send AFD or repeat the last value if the AFD input is lost.

Pre-Processing Area

▼ Pre-Processing

Interlacing Options:

None

Deinterlace Mode:

<default>

Noise Reduction Filter:

None

☐ Edge Enhance

Edge Threshold:

8

Median Filter:

☐ Enable

Watermark:

☐ Enable

File Name:

Browse...

Left:

0

Width:

0

Top:

0

Height:

0

Opacity:

100

%

☐ No Chroma

☐ Banner

Interlacing Options

If your video input is interlaced *and* your resolution height is greater than the field height, choose deinterlace to convert interlaced video to progressive. For resolutions less than or equal to the field height (320x240 for example for and NTSC source or 960x540 for HD 1080 sources), no deinterlacing is required and none should be specified.

Choose inverse telecine (IVT) to convert film-based interlaced 30 fps video to progressive 24 fps. If IVT is specified and the encoding mode is interlaced, the encoder will produce IVT flags (top field first, bottom field first, repeat first field) in the bitstream; decoders will then know how to display the 24 fps progressive video on an interlaced display.

Deinterlace Mode

If deinterlacing is chosen, the encoder provides multiple methods for deinterlacing. The default, a Motion Adaptive deinterlace, attempts to preserve spatial information in areas of motion while removing interlace artifacts in areas of motion. The Advanced option provides a higher quality deinterlace than the other options, but also uses significantly more CPU cycles. The Blend option blends two fields, maintaining temporal information through motion blur. The Interpolate mode removes temporal information and interpolates even fields from odd fields, while Interpolate Denoise applies a noise reduction filter after the deinterlace. The Line Double option simply creates even field lines as direct copies of the odd field lines. The CAVT option uses content adaptive vertical temporal filtering.

Noise Reduction Filter

You can choose between various noise reducing filters, or choose None for no filter. The Light filter uses less noticeable filtering that should have less impact on

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picture quality. The Normal filter provides for moderate noise reduction. The Smooth filter provides for large noise reduction at the expense of softer images.

If you choose one of the above filters, you can also check the **Edge Enhance** check box and enter a threshold to add the edge enhance mode to the filter. This mode detects edges in the image, and the detected edges are not processed by the filtering. The intent of this mode is to preserve edges and sharpness while still applying noise reduction to smoother areas. The edge threshold specifies the sensitivity of the edge detection. Lower values increase sensitivity, while higher values may detect few if any edges. The range is 8 to 128.

Median Filter

Check the **Enable** check box to use the Median filter. This filter is a standard image processing filter best used on noisy images.

Watermark

To add a watermark image, check the **Enable** check box and browse or provide the path to the watermark image file. The File Name field must include the path to the existing image file either on the AnyRes Live encoder (recommended) or a network share accessible by AnyRes Live. The watermark image file may be of type .gif, .bmp, .jpg, .jpeg, .png, .tif, or .tiff. You can use the same image file for different streams.

Left and Top specify the pixel location where the upper left pixel of the watermark will be placed. Default Left 0, Top 0 is the upper left of the image. Enter Left -1, Top -1 to specify alignment to the bottom right location.



Width and Height cause the watermark image to be resized (enlarged or reduced) to be width pixels wide and height pixels high. Resizing is not supported for bitmaps that have an alpha channel. Both the original and resized watermark must be no larger than the encoded image size. Default Width 0, Height 0 specifies no resizing. You may want to use different resolution watermark images for different streams to preserve observed watermark size, or the same image can be scaled to different sizes.

Optionally specify the opacity of the watermark. The default is 100% opaque.

Check the **No Chroma** check box to remove color information from the watermark image to display it in black and white.

Check the **Banner** check box to cause the watermark image to move two pixels per frame across the output image from right to left.

Additional Compression Settings Area

▼ Additional Compression Settings	
Number of B Frames:	7
Lookahead Frames:	60  

The Additional Compression Settings area gives you control over some additional H.265 encoding tools.

Number of B Frames

Display only. This setting specifies the maximum number of B frames to use between other types of frames. Adaptive Streaming requires this setting to be the same on all streams. For H.265 video, this value is 7.

Lookahead Frames

Choose the maximum number of lookahead frames to use in rate control decisions. The default is 60 frames, and we recommend that you set this value equal to the key frame interval. Valid values range from 17 to 120 frames, unless the frame rate is less than the input rate. If the frame rate is reduced, then the lookahead frames maximum value is reduced by the same divisor, unless the result is less than the minimum of 17 lookahead frames.

Captioning Area

▼ Captioning

Source: Line 21

608: ☐ CC1/CC2 ☐ CC3/CC4

708: ☐ Service 1 ☐ Service 2 ☐ Service 3 ☐ Service 4

SAMI Output: ☐ Enable

Open Captions: ☐ Enable

Teletext Subtitles: ☐ Enable

Teletext Mode: Word by Word ▼

Note: Closed Captions cannot be enabled with either Open Captions or Teletext Subtitles.

608

Check the appropriate check box(es) to process field 1 (CC1/CC2) and/or field 2 (CC3/CC4) captioning on the input source and to either have CEA 608 data inserted into the video elementary stream user data area or to enable SAMI output. Closed Captions are not available on 25 fps sources.

708

If you have checked the appropriate check boxes under 608 to process field 1 (CC1/CC2) and field 2 (CC3/CC4) captioning on the input source, use the corresponding Service 1 through Service 4 check boxes to have up-converted CEA-708 data inserted into the video elementary stream user data area.

SAMI Output

If 608 Closed Captioning is enabled, check the **Enable** check box to generate a .smi output file in the same directory as the WMV file specified on the Output tab. To output SAMI, at least one of the 608 check boxes must be enabled.

Open Captions

When Open Captions is enabled, Line21 CC1 captions are decoded and rendered on top of the incoming video. Thus, unlike 608, 708 or SAMI captioning, the rendering of open captions is done on top of video in the encoder instead of in the decoder. This processing is independent of 608, 708, and SAMI captioning and none of these options need to be enabled for open captions.

Open captions requires NTSC 720x480 SD input sources.

Note: When open captions is enabled, it is enabled for all output streams, including any VC-1, VP6 Flash, H.264, or H.263 streams.

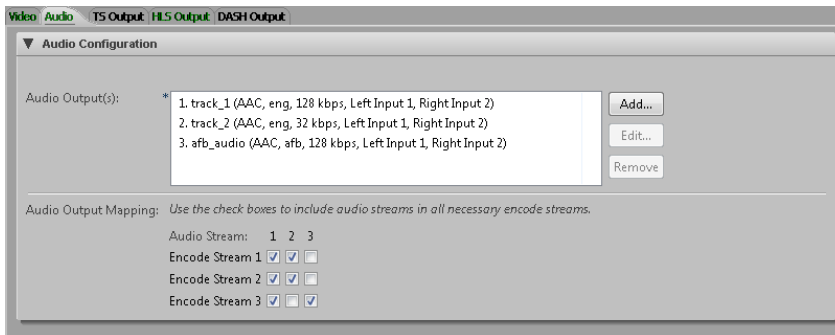
Teletext Subtitles

If teletext subtitle data is present in the input video and the source format is PAL (or, for IP input, if the source carries teletext data on a transport stream PID), and teletext has been configured on the Input Video tab, check the **Enable** check box to insert the teletext subtitle data in output streams that support teletext.

Teletext Mode

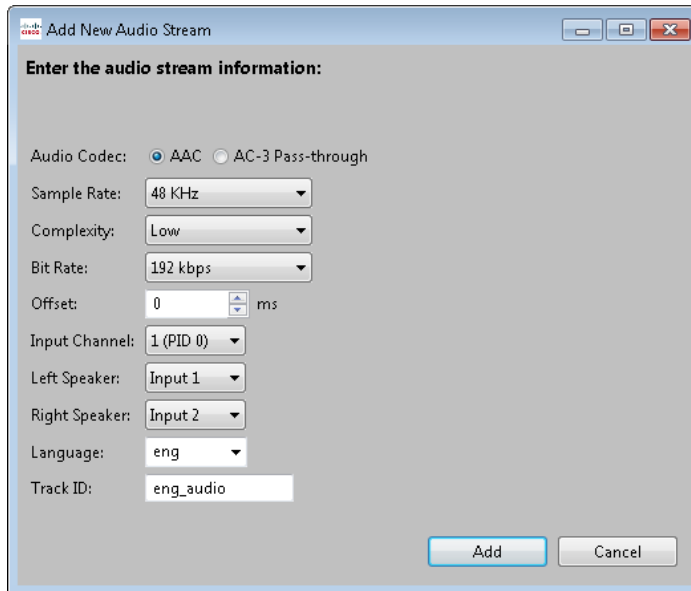
If Teletext Subtitles is enabled, choose the mode. Line by Line mode tries to collect complete lines before sending the cue point. In this mode, any leading white space is removed from all the subtitle text. Word by Word mode sends out subtitle text as soon as it is decoded from the video frame.

H.265 Audio Tab



For H.265 audio configuration, a list of currently configured audio outputs is displayed. Click **Add** to display the Add New Audio Stream dialog box to configure additional AAC or AC-3 Pass-through audio outputs:

AAC Audio



Add New Audio Stream

Enter the audio stream information:

Audio Codec: ☒ AAC ☐ AC-3 Pass-through

Sample Rate: 48 KHz

Complexity: Low

Bit Rate: 192 kbps

Offset: 0 ms

Input Channel: 1 (PID 0)

Left Speaker: Input 1

Right Speaker: Input 2

Language: eng

Track ID: eng_audio

Add Cancel

To configure an AAC audio stream, choose AAC for the Audio Codec. You can modify the following settings:

Sample Rate

Specify the sample rate of input audio from the list of available rates in hertz, which varies based on the bit rate chosen above. The input audio will be resampled as needed to this target sample rate prior to encoding the audio.

Complexity

The complexity defaults to Low complexity, or choose HE v1 or HE v2.

Bit Rate

Choose from the list of available audio bit rates.

Offset

Enter an offset (in milliseconds) between audio and video. The offset can be a positive or negative value. Typically, this value is 0 unless you know that your

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source audio and video are not in sync. If not in sync, use this setting to adjust timing such that they are re-synced prior to encoding.

Input Channel

For IP Input video, choose from the available audio inputs.

Left Speaker

Choose from the available audio inputs to specify the Left stereo output. For Mono audio, populate only this drop-down list, and choose None for Right Speaker.

Right Speaker

Choose from the available audio inputs to specify the Right stereo output. For Mono audio, choose None.

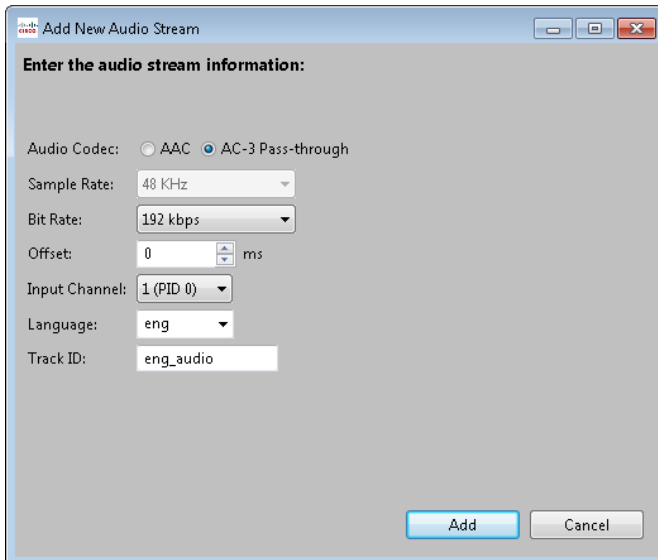
Language

Enter the ISO-639-2/T language label to apply to each stream.

Track ID

Specify the ID for the audio stream. Each HSS audio stream must have a unique ID. The client player should allow the user to choose between various audio IDs.

AC-3 Pass-through Audio



Add New Audio Stream

Enter the audio stream information:

Audio Codec: ☐ AAC ☒ AC-3 Pass-through

Sample Rate: 48 KHz

Bit Rate: 192 kbps

Offset: 0 ms

Input Channel: 1 (PID 0)

Language: eng

Track ID: eng_audio

Add Cancel

To configure an AC-3 Pass-through audio stream, choose AC-3 Pass-through for the Audio Codec. You can modify the following settings:

Sample Rate

Display only. The only supported AC-3 pass-through sample rate is 48 kHz.

Bit Rate

Choose from the list of available audio bit rates.

Offset

Enter an offset (in milliseconds) between audio and video. The offset can be a positive or negative value. Typically, this value is 0 unless you know that your source audio and video are not in sync. If not in sync, use this setting to adjust timing such that they are re-synced prior to encoding.

Input Channel

For IP Input video, choose from the available audio inputs.

Language

Enter the ISO-639-2/T language label to apply to each stream.

Track ID

Specify the ID for the audio stream. Each HSS audio stream must have a unique ID. The client player should allow the user to choose between various audio IDs.

Audio Output Mapping

For mapping audio outputs, each enabled encode stream has a check box for each enabled audio stream. Check the check box for any audio stream that should be encoded with a given encode stream. Each audio output must be mapped to at least one encode stream. The Media Processor AS5100, AS7100, and AS8100 are only licensed for one audio stream per encode stream. AnyRes Live 5300, 8300, and 9300 are licensed for up to 8 audio streams per encode stream.

H.265 Outputs

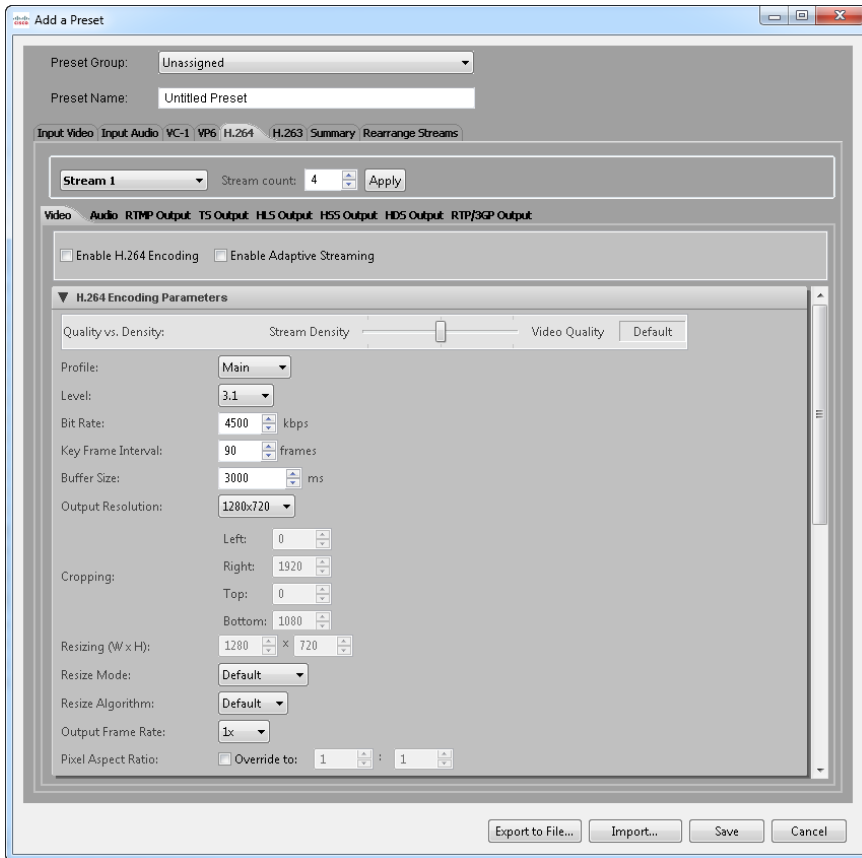
H.265 TS output, HLS output, and DASH output settings are identical to the corresponding H.264 output settings.

- For TS output settings, see “H.264 TS Output Tab” on page 100.
- For HLS output settings, see “H.264 HLS Output Tab” on page 108.
- For DASH output settings, see “H.264 DASH Output Tab” on page 128.

H.264 Tab

Note: H.265 and H.264 outputs cannot be enabled simultaneously in the same preset. If you enable H.264 output while H.265 output is enabled in the preset, H.265 will be automatically disabled.

H.264 Video Tab



In the H.264 tab, first enter the Stream count value and click **Apply**. In the Video tab, enable H.264 encoding, and optionally enable Adaptive Streaming for one or more streams. Adaptive Streaming forces subsequent streams to have the same values for Key Frame Interval, and also for the Number of B Frames and Scene Change Detection additional compression settings. To change these settings on other streams, you must check the **Unlocked** check box that appears on stream 2 and above when Adaptive

Streaming is enabled. Finally, view or modify the settings in each area of the Video tab for each stream.

H.264 Encoding Parameters Area

Quality vs. Density

This slider determines values for many of the Advanced Codec settings for the Default codec. The slider will not be visible if the Legacy codec is in use. Each slider setting from left to right increases CPU utilization as more video quality encoding tools are enabled, which may cause a high CPU alarm or dropped frames alarm on the encoder. If you experience these issues, simplify the preset by moving the slider setting to the left, reducing the number of output streams, or reducing output stream resolution.

Profile

The profile for video encoding specifies a subset of the H.264 syntax required for decoding the stream.

H.264 Baseline—Used primarily for lower-cost applications with limited computing resources, this profile is used widely in videoconferencing and mobile applications. It supports progressive video, uses I and P slices, and CAVLC entropy coding.

H.264 Main—Targeted mainly towards the broadcast market, this profile supports both interlaced and progressive video with macroblock or picture level field/frame mode selection. Uses I, P, and B slices, weighted prediction, and both CAVLC and CABAC for entropy coding.

H.264 High—An extension of the Main profile for effective coding of High Definition content, this profile uses adaptive 8x8 or 4x4 transform and enables perceptual quantization matrices.

Level

This setting specifies constraints for the encoding parameters.

Bit Rate

This setting indicates bits per second in units of kbps. For example, a value of 1500 is 1.5 Mbps. The default is 3000 kbps. The bit rate must be unique for each H.264 output stream.

Key Frame Interval

This value (in frames) sets the maximum distance between key frames. The default is 60 frames for non-PAL video and 50 frames for PAL video, which is 2 seconds of video. The encoder may output key frames sooner than this interval (if it detects a scene change, for example). Adaptive Streaming requires this interval to be the same for all adaptive streams, and should be a multiple of the frame rate. Segment

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Duration and Fragment Duration settings on the TS Output tab must each be a multiple of the key frame interval.

Buffer Size

The buffer size value (in milliseconds) sets the encoder buffer size.

Output Resolution

Choose from the available resolutions or choose Custom, which will make the Cropping and Resizing settings available for modification. The pixel aspect ratio is auto set for each of these presets.

Cropping

The cropping parameters apply a crop to the input image. Note that if an odd number of lines are cropped from the top, the sense of which field is first (top or bottom) will change and you must set the Field Order setting on the Input Video tab accordingly.

Resizing

Specify the output resolution to be applied to the cropped image. If the resolution is different than the original, scaling will be performed. The minimum supported resolution is 64x64.

Resize Mode

Choose Progressive mode, Interlaced mode, or Single Field mode to manually specify how the resizer will process the source video. Single field mode, which should only be used for interlace source material, scales to the destination image using only a single field of the source video instead of performing a deinterlace operation. We highly recommend that you use single field mode for producing progressive frames from a 1080i source. For example, to stream 1280x720p from a 1920x1080i source, we recommend that you choose the single field mode and resize to 1280x720 instead of performing a deinterlace operation.

Resize Algorithm

Choose Nearest, Linear, Cubic, or Super to choose the algorithm the resize operation uses. By default, Linear is used. Nearest (also referred to as Nearest neighbor) is the worst quality and has the lowest CPU requirements. Cubic is a higher quality scaler than linear, and Super is a higher quality algorithm than Cubic. However, Super is very processor intensive and is only used for downscaling, and only when the resize is less than approximately half the source size. For example, if scaling single field mode from 1920x1080i, the field size is 1920x540, therefore, use Super if the output size is no more than 270 in height.

Output Frame Rate

Choose from the available frame rate options. For output heights of 720 or less, you can choose the 2x frame rate to encode each input field of interlaced source as a progressive picture, which converts 1080i sources to 720p outputs. Choose 1x to

use the input frame rate. Choose fractional frame rates to reduce the input frame rate for telecine purposes or frame rate decimation. Available fractional frame rate options are 1/2, 1/3, 1/4, 1/5, and 1/6 the input rate.

Pixel Aspect Ratio

To override the automatically set pixel aspect ratio, check the **Override** check box and modify the calculated ratio to the new ratio. If this check box is unchecked, the pixel aspect ratio is calculated based on the output size of the video for the preset sizes. Overriding is useful in cases where source video has a non-standard aspect ratio. For example, a 720x480 input source may have been produced anamorphically, and by indicating the aspect ratio here players can correctly resize video on playback to the non-anamorphic size.

Aspect Ratio Control (IP Input only)

When Dynamic is enabled on the Input Video tab, choose the method that is used to maintain the output aspect ratio when the input resolution or aspect ratio changes during encoding. Manual specifies not to automatically adjust for the aspect ratio. Letter/Pillar will add letterboxing or pillarboxing to achieve the specified output aspect ratio. Crop will modify the crop settings to achieve the specified output aspect ratio. Variable will pass through changes in the input Sequence Parameter Set aspect ratio to the output Sequence Parameter Set, and no cropping, letterboxing, or pillarboxing is performed. The default is currently set to Crop.

Active Format Description (AFD) (IP Input Only)

Check the **Enable** check box to pass the Active Format Description through to H.264 user data. Also, choose from the drop-down list to specify whether the encoder should not send AFD or repeat the last value if the AFD input is lost.

Pre-Processing Area

▼ Pre-Processing

Interlacing Options:

None

Deinterlace Mode:

<default>

Noise Reduction Filter:

None

☐ Edge Enhance

Edge Threshold: 8

☐ Median Filter:

Enable

☐ Watermark:

Enable

File Name:

Browse...

Left: 0

Width: 0

Top: 0

Height: 0

Opacity: 100 %

☐ No Chroma

☐ Banner

Interlacing Options

If your video input is interlaced *and* your resolution height is greater than the field height, choose deinterlace to convert interlaced video to progressive. For resolutions less than or equal to the field height (320x240 for example for and NTSC source or 960x540 for HD 1080 sources), no deinterlacing is required and none should be specified.

Choose inverse telecine (IVT) to convert film-based interlaced 30 fps video to progressive 24 fps. If IVT is specified and the encoding mode is interlaced, the encoder will produce IVT flags (top field first, bottom field first, repeat first field) in the bitstream; decoders will then know how to display the 24 fps progressive video on an interlaced display.

Deinterlace Mode

If deinterlacing is chosen, the encoder provides multiple methods for deinterlacing. The default, a Motion Adaptive deinterlace, attempts to preserve spatial information in areas of motion while removing interlace artifacts in areas of motion. The Advanced option provides a higher quality deinterlace than the other options, but also uses significantly more CPU cycles. The Blend option blends two fields, maintaining temporal information through motion blur. The Interpolate mode removes temporal information and interpolates even fields from odd fields, while Interpolate Denoise applies a noise reduction filter after the deinterlace. The Line Double option simply creates even field lines as direct copies of the odd field lines. The CAVT option uses content adaptive vertical temporal filtering.

Noise Reduction Filter

You can choose between various noise reducing filters, or choose None for no filter. The Light filter uses less noticeable filtering that should have less impact on picture quality. The Normal filter provides for moderate noise reduction. The Smooth filter provides for large noise reduction at the expense of softer images.

If you choose one of the above filters, you can also check the **Edge Enhance** check box and enter a threshold to add the edge enhance mode to the filter. This mode detects edges in the image, and the detected edges are not processed by the filtering. The intent of this mode is to preserve edges and sharpness while still applying noise reduction to smoother areas. The edge threshold specifies the sensitivity of the edge detection. Lower values increase sensitivity, while higher values may detect few if any edges. The range is 8 to 128.

Median Filter

Check the **Enable** check box to use the Median filter. This filter is a standard image processing filter best used on noisy images.

Watermark

To add a watermark image, check the **Enable** check box and browse or provide the path to the watermark image file. The File Name field must include the path to the existing image file either on the AnyRes Live encoder (recommended) or a network

share accessible by AnyRes Live. The watermark image file may be of type .gif, .bmp, .jpg, .jpeg, .png, .tif, or .tiff. You can use the same image file for different streams.

Left and Top specify the pixel location where the upper left pixel of the watermark will be placed. Default Left 0, Top 0 is the upper left of the image. Enter Left -1, Top -1 to specify alignment to the bottom right location.

Width and Height cause the watermark image to be resized (enlarged or reduced) to be width pixels wide and height pixels high. Resizing is not supported for bitmaps that have an alpha channel. Both the original and resized watermark must be no larger than the encoded image size. Default Width 0, Height 0 specifies no resizing. You may want to use different resolution watermark images for different streams to preserve observed watermark size, or the same image can be scaled to different sizes.

Optionally specify the opacity of the watermark. The default is 100% opaque.

Check the **No Chroma** check box to remove color information from the watermark image to display it in black and white.

Check the **Banner** check box to cause the watermark image to move two pixels per frame across the output image from right to left.

Additional Compression Settings Area

▼ Additional Compression Settings

Codec:	Default	Adjust quality/density	Scene Change Detection:	Off
Number of B Frames:	None		Interlace Mode:	Progressive
Number of Reference Frames:	2		Output Color Description:	<input type="checkbox"/> Enable
Write Sequence End Code:	True		Transfer Characteristics:	ITU-R BT.709-5
Write AU Delimiters:	True		Color Primaries:	ITU-R BT.709-5
Number of Threads:	Default		Matrix Coefficients:	RGB

Reset to Defaults

The latest AnyRes Live software includes an updated H.264 codec to provide market leading video quality. The updated H.264 codec is the default setting on the encoder and we recommend that you use it for all new preset creations. The legacy H.264 codec is provided as an option for continued use in production environments, until an opportunity is available to test and move to the updated H.264 codec. We highly recommend that you use the default H.264 codec for optimal performance and overall video quality.

Use the **Codec** setting to choose either the Default codec or the Legacy codec. Available settings change according to the codec currently in use. If you choose the Default codec, you can click the **Adjust quality/density** link to go to the Quality vs. Density slider in the H.264 Encoding Parameters area.

The advanced compression settings area gives you control over many of the H.264 encoding tools. This is important because, depending on your decoding solution (whether it be a mobile device, an STB or a PC), you may need to limit or want to expand on the tools used for encoding. Click **Reset to Defaults** to reset all additional compression settings to their default value.

Number of B Frames

This setting specifies the maximum number of B frames to use between other types of frames, up to 3. B frames are not supported for Baseline profile. Adaptive Streaming requires this setting to be the same and recommends a setting of None for all adaptive streams.

Note: Number of B Frames should always be set to None when using Baseline profile or when using RTP/3GPP Output.

Number of Reference Frames

This setting specifies the number of reference frames. The range of valid values is 0 to 4.

Write Sequence End Code

This setting may be set to True or False to write the end of sequence code. Default True.

Weighted Prediction

This setting may be set to True or False to write the end of sequence code. For the Legacy codec, the default is True. For the Default codec, this setting is not yet supported.

Write AU Delimiters

This setting may be set to True or False to write access unit delimiters. The default is True.

Number of Threads

This setting specifies the number of threads to use for encoding, up to the number of processors. This setting is intended to take advantage of multiple processors. The default is set internally by the encoder.

Weighted Prediction

This setting, available only for the Legacy codec, may be set to True or False to set the weighted prediction (WP) for P-frames, which results in more efficient encoding especially in case of scenes containing fades. The default is True.

Max Quant

This setting, available only for the Legacy codec, defines the maximum quantization parameter to use. If multicasting is enabled on the TS Output tab, this setting will be 51 and cannot be modified. Valid values are 30 to 51. The default is 51.

Min Quant

This setting, available only for the Legacy codec, defines the minimum quantization parameter to use. Valid values are 0 to 51. The default is 0.

Scene Change Detection

This setting specifies the mode for inserting key frames on scene detection. Valid values IDR and Off. The default is IDR. Adaptive Streaming requires this setting to be the same and recommends a setting of Off for all adaptive streams.

Interlace Mode

This setting controls whether the encoder is in Progressive mode or one of two Interlaced modes. Progressive uses frame encoding. Interlaced (Fields) uses field encoding. Interlaced (MBAFF) enables Macroblock Adaptive Frame/Field decisions in the encoder, allowing it to choose, on a per-macroblock basis, whether to use frame or field encoding. The interlace mode must be Progressive for RTP/3GPP Output. The default is Progressive.

Output Color Description

This insertion of color description information into the H.264 video stream is optional. If desired or required by the end playback device, you can enable insertion of the three color description settings (**Transfer Characteristics**, **Color Primaries**, and **Matrix Coefficients**) by checking the Output Color Description check box. With this checked, further specify the three color characteristics that describe the color of your source video. If Output Color Description is unchecked, these settings are not output. For details on the color options, see the ISO/IEC 14496-10 specification.

Entropy Encoding Mode

This setting, available only for the Legacy codec, may be set to CABAC (Context-based Adaptive Binary Arithmetic Coding (not allowed for Baseline profile)) or CAVLC (Context-based Adaptive Variable Length Coding). Default varies by preset.

ME Subpel Mode

This setting, available only for the Legacy codec, specifies the subpixel motion search depth. Full—only full pixel position will be examined, Half—half-pixel positions will be added to the search, Quarter—both half and quarter pixel positions will be added. The default is Quarter.

ME Search Range

This setting, available only for the Legacy codec, specifies the motion estimation search range in full pixels. Valid values 63, 127, 255, or 511. The default is 127.

Advanced Codec Settings Area

The screenshot shows the 'Advanced Codec Settings' panel with the following settings:

- Override Preset Settings: ☒ Override
- Adaptive B Frames: Fast
- Deblocking Filter: ☒ Enable Alpha: 0 Beta: 0
- CABAC: ☒ Enable
- Weighted Prediction: Simple
- Motion Estimation Method: Hexagon
- Motion Estimation Range: 16
- Subpel Refinement: QPel SAD 2 iterations
- Mixed References: ☐ Enable
- Trellis: Disabled
- Macroblock Tree Rate Control: ☒ Enable
- Lookahead Frames: 10
- Minimum IDR Interval: 1
- B Frame References: Normal
- Chroma QP Offset: 0

The Advanced Codec Settings area is available only for the Default codec. Most of these setting values are initially determined by the Quality vs. Density slider in the H.264 Encoding Parameters area of the H.264 Video tab. Use the fields in this area to change the slider value of any of the following advanced settings:

Adaptive B Frames

When enabled, this setting chooses the adaptive B frame placement algorithm. Optimal can be significantly slower than Fast.

B Frame References

This setting controls the use of B frames as references. Choose Strict to limit to one reference B frame per minigop (I or P frame plus following B frames). Normal allows multiple reference B frames per minigop. Choose None if B frames should not be used as references.

Minimum IDR Interval

This setting sets the minimum number of frames between IDR frames. Valid values are 1 to 60.

Deblocking Filter

Check the **Enable** check box to use the in-loop deblocking filter, and specify Alpha and Beta offsets. Higher values lead to more smoothing of block edges. Valid values are -6 to 6.

CABAC

Check the **Enable** check box to use CABAC (Context-based Adaptive Binary Arithmetic Coding). Uncheck the check box to fall back to CAVLC (Context-based Adaptive Variable Length Coding).

Chroma QP Offset

Specify an offset to add to the quantizer of chroma planes when encoding. This offset can be negative.

Weighted Prediction

This setting enables weighted prediction for P-frames, which results in more efficient encoding especially in case of scenes containing fades. Choose Simple for fade analysis without reference duplication, or choose Smart for fade analysis with reference duplication.

Motion Estimation Method

Choose the motion estimation method from the drop-down list. Diamond is the simplest and fastest method, Hexagon is more complex and slower, and Uneven Multi-Hex is the most complex and slowest method.

Motion Estimation Range

Specify, in pixels, the maximum range of the motion estimation search. Valid values are 4 to 24.

Subpel Refinement

From the drop-down list, choose the subpel estimation complexity. Full Pel Only is the fastest and simplest option, and subsequent options increase in complexity and slowness. We do not recommend Full Pel Only and QPel SAD 1 iteration unless speed is very important.

Mixed References

Check the **Enable** check box to choose references based on an 8x8 partition, instead of per macroblock. Enabling this additional processing can increase quality when using multiple reference frames.

Trellis

This setting enables the Trellis quantization algorithm. Macroblock is faster but less efficient than All Decisions.

Macroblock Tree Rate Control

Check the **Enable** check box to attempt to improve compression by tracking temporal propagation across frames.

Lookahead Frames

Choose the maximum number of lookahead frames to use in rate control decisions.

Captioning Area

▼ Captioning

Source: Line 21

608: ☐ CC1/CC2
☐ CC3/CC4

708: ☐ Service 1
☐ Service 2
☐ Service 3
☐ Service 4

SAMI Output: ☐ Enabled

Open Captions: ☒ Enabled (Note: Open Captioning applies to all output streams)

Cue Point Output: ☐ Enabled (Note: Cue Point Output applies to all flash output streams)

Teletext Subtitles: ☒ Enabled

Teletext Mode: Word by Word ▼

Note: Closed Captions cannot be enabled with either Open Captions or Teletext Subtitles.

608

Check the appropriate check box(es) to process field 1 (CC1/CC2) and/or field 2 (CC3/CC4) captioning on the input source and to either have CEA 608 data inserted into the video elementary stream user data area or to enable SAMI output. Closed Captions are not available on 25 fps sources.

708

If you have checked the appropriate check boxes under 608 to process field 1 (CC1/CC2) and field 2 (CC3/CC4) captioning on the input source, use the corresponding Service 1 through Service 4 check boxes to have up-converted CEA-708 data inserted into the video elementary stream user data area.

SAMI Output

If 608 Closed Captioning is enabled, check the **Enable** check box to generate a .smi output file in the same directory as the WMV file specified on the Output tab. To output SAMI, at least one of the 608 check boxes must be enabled.

Open Captions

When Open Captions is enabled, Line21 CC1 captions are decoded and rendered on top of the incoming video. Thus, unlike 608, 708 or SAMI captioning, the rendering of open captions is done on top of video in the encoder instead of in the decoder. This processing is independent of 608, 708, and SAMI captioning and none of these options need to be enabled for open captions.

Open captions requires NTSC 720x480 SD input sources.

Note: When open captions is enabled, it is enabled for all output streams, including any VC-1, VP6 Flash, H.265, or H.263 streams.

Cue Point Output

When cue point output is enabled, NTSC caption information is automatically inserted into all Flash streams as a cue point.

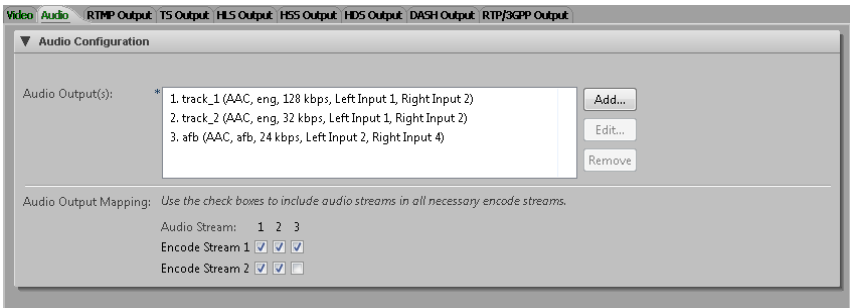
Teletext Subtitles

If teletext subtitle data is present in the input video and the source format is PAL, and teletext has been configured on the Input Video tab, check the **Enable** check box to insert the teletext subtitle data in output streams that support teletext.

Teletext Mode

If Teletext Subtitles is enabled, choose the mode. Line by Line mode tries to collect complete lines before sending the cue point. In this mode, any leading white space is removed from all the subtitle text. Word by Word mode sends out subtitle text as soon as it is decoded from the video frame.

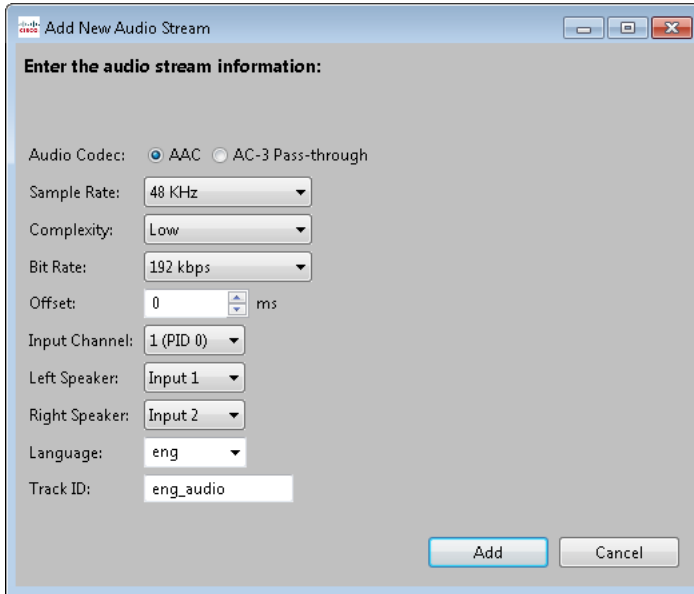
H.264 Audio Tab



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For H.264 audio configuration, a list of currently configured audio outputs is displayed. Click **Add** to display the Add New Audio Stream dialog box to configure additional AAC or AC-3 Pass-through audio outputs:

AAC Audio



To configure an AAC audio stream, choose AAC for the Audio Codec. You can modify the following settings:

Sample Rate

Specify the sample rate of input audio from the list of available rates in hertz, which varies based on the bit rate chosen above. The input audio will be resampled as needed to this target sample rate prior to encoding the audio.

Complexity

The complexity defaults to Low complexity, or choose HE v1 or HE v2.

Bit Rate

Choose from the list of available audio bit rates.

Offset

Enter an offset (in milliseconds) between audio and video. The offset can be a positive or negative value. Typically, this value is 0 unless you know that your

source audio and video are not in sync. If not in sync, use this setting to adjust timing such that they are re-synced prior to encoding.

Input Channel

For IP Input video, choose from the available audio inputs.

Left Speaker

Choose from the available audio inputs to specify the Left stereo output. For Mono audio, populate only this drop-down list, and choose None for Right Speaker.

Right Speaker

Choose from the available audio inputs to specify the Right stereo output. For Mono audio, choose None.

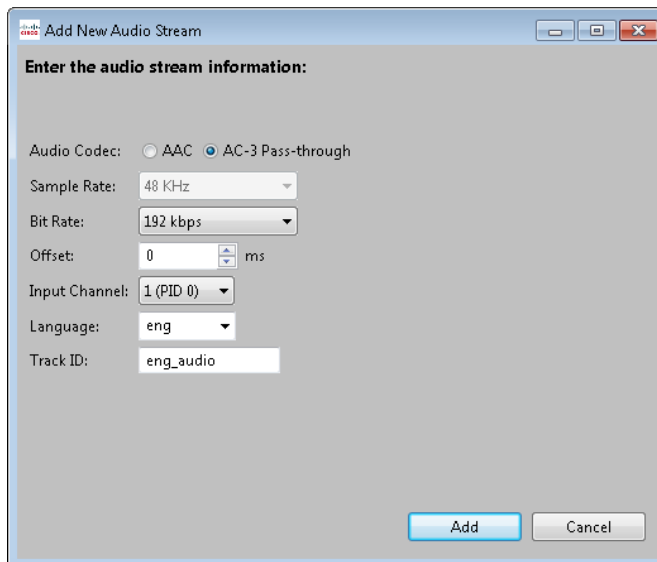
Language

Enter the ISO-639-2/T language label to apply to each stream.

Track ID

Specify the ID for the audio stream. Each HSS audio stream must have a unique ID. The client player should allow the user to choose between various audio IDs.

AC-3 Pass-through Audio



To configure an AC-3 Pass-through audio stream, choose AC-3 Pass-through for the Audio Codec. You can modify the following settings:

Sample Rate

Display only. The only supported AC-3 pass-through sample rate is 48 kHz.

Bit Rate

Choose from the list of available audio bit rates.

Offset

Enter an offset (in milliseconds) between audio and video. The offset can be a positive or negative value. Typically, this value is 0 unless you know that your source audio and video are not in sync. If not in sync, use this setting to adjust timing such that they are re-synced prior to encoding.

Input Channel

Choose from the available audio inputs.

Language

Enter the ISO-639-2/T language label to apply to each stream.

Track ID

Specify the ID for the audio stream. Each HSS audio stream must have a unique ID. The client player should allow the user to choose between various audio IDs.

Audio Output Mapping

For mapping audio outputs, each enabled encode stream has a check box for each enabled audio stream. Check the check box for any audio stream that should be encoded with a given encode stream. Each audio output must be mapped to at least one encode stream. The Media Processor AS5100, AS7100, and AS8100 are only licensed for one audio stream per encode stream. AnyRes Live 5300, 8300, and 9300 are licensed for up to 8 audio streams per encode stream.

H.264 RTMP Output Tab

The screenshot shows the 'H.264 RTMP Output Tab' configuration window. At the top, there are tabs for 'Video', 'Audio', 'RTMP Output', 'TS Output', 'HLS Output', 'H5S Output', 'HDS Output', 'DASH Output', and 'RTP/SGPP Output'. The 'RTMP Output' tab is selected. Below the tabs, there is a section for 'Enable RTMP Output' with a checked checkbox and a 'Copy' button. The 'Streaming to Origin Server' section contains a list of origin servers with one entry: 'rtmp://192.0.2.1/live/examplestream'. To the right of this list are 'Add...', 'Edit...', and 'Remove' buttons. Below the list, there are several settings: 'Delay' (0 ms), 'onFI Timecode Interval' (0 ms), 'Reconnect Retry Count' (5), 'Retry Delay' (20 sec), 'Minimum Pre-Roll' (0 sec), and 'Closed Captions' (Enable checkbox, Send only CC1/CC2 checkbox). The 'MP4 Archive File' section has 'Stream: Main' (Enable checkbox, Generate unique file name checkbox) and 'MP4 File' (text field, Use Full Path checkbox).

Streaming to RTMP Origin Server Area

Check the **Enable Streaming** check box to push content to a RTMP origin server. Click **Copy** to populate the stream settings from the settings of the primary stream.

Origin Server(s)

Up to eight unique origin servers can be configured. To add the information to identify an origin server, click **Add** to display the Add New Origin Server dialog box.

The screenshot shows the 'Add new Origin Server' dialog box. It has a title bar with the Cisco logo and the text 'Add new Origin Server'. The main area is titled 'Enter the Origin Server information:'. It contains the following fields: 'URL:' (rtmp://), 'Stream Name:', 'Authentication:' (None), 'Username:', and 'Password:'. At the bottom, there are 'Add' and 'Cancel' buttons.

First, enter the URL for the origin server with the format of *server-IP/application-name*. By default, AnyRes Live uses port 1935. To override the port, use

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`rtmp://server-IP:port/application-name`. Next, enter the stream name. For Authentication, choose the CDN (content distribution network) from the drop-down list and enter the username and password to enable authentication with an origin server before the stream can be published. The password characters will not be displayed. If you choose the None option, the Username and Password fields are dimmed and ignored by AnyRes Live.

After specifying the settings of the origin server, click **Add** to add it to the list. To edit an origin server after it has been added, select the server address, and then click **Edit** to display the Update Origin Server dialog box. In this dialog box, modify settings, and then click **Update**. To remove an origin server from the list, select the server address, and then click **Remove**. Multiple origin servers can be selected and removed at once.

Delay

Enter the delay in milliseconds for the output streams. Although this delay setting appears on the H.264 RTMP output tab for each encode stream, there is only one global setting and it affects all H.264 Flash streams.

onFI Timecode Interval

The onFI Timecode Interval can be used to insert timecode metadata into the Flash RTMP stream. The interval in milliseconds specifies the interval between sending onFI messages. If set to 0 (default) this is turned off. Timecode must be provided to the encoder, otherwise no timecode will be sent.

Reconnect Retry Count

Set to the number of times the encoder will retry connecting to the configured server. This setting applies to all H.264 Flash streams. The default is 5.

Retry Delay

Set to the delay in seconds between retry attempts. This setting applies to all H.264 Flash streams. The default is 20 seconds.

Minimum Pre-Roll

Enter the minimum pre-roll, which is the minimum amount of time a splice cue point is to be sent before the splice that it is signaling. This setting applies to all H.264 RTMP output streams. The default is 0 seconds.

Closed Captions

Check the **Enable** check box to enable closed caption data to be streamed to the RTMP origin server. A source for captions must be specified on the H.264 Video tab. Check the **Send only CC1/CC2** check box to restrict the included captions to CC1/CC2 only, instead of the default, which also includes CC3/CC4.

MP4 Archive File Area

The encoder can optionally save the H.264 stream to disk. Check the **Enable** check box and supply a path and file name to save to a user-specified MP4 file. The default MP4 file output format is QuickTime.

When using a network drive, a fully qualified UNC path is required and the AnyRes Live Encoding Service must have permission to create directories and copy files to the destination (for instructions see “Setting Up AnyRes Live to Write to a Network Drive” on page 265).

Note: If no path information is given, the file output is stored at C:\inetpub\wwwroot\encadmin\output on the encoder.

Check the **Generate unique file name** check box to append the current date and time to the file name, which will ensure that previous archive files will not be overwritten.

MP4 archives cannot be written at the same time as TS archives.

H.264 TS Output Tab

The screenshot shows the 'H.264 TS Output' configuration tab. At the top, there are tabs for 'Video', 'Audio', 'RTMP Output', 'TS Output' (which is selected), 'HLS Output', 'H5S Output', 'HDS Output', and 'RTP/3GPP Output'. The 'Adaptive Transport Stream' section is expanded, showing 'Enable Adaptive TS' checked, 'Segment Duration' set to 9 seconds, and 'Fragment Duration' set to 1 second. Below this are sections for 'Transport Stream Markers' and 'Transport Stream Archive', both with expandable arrows. The 'Network Transfer Configuration' section is also expanded, showing two columns for 'Primary' and 'Secondary' settings. Both are checked. The 'Primary' column has 'Destination Address' as 224.0.0.1, 'Network Transfer Mode' as Multicast, 'Preferred Adapter' as Main (LAN1), 'Port' as 55514, 'TTL' as 8, 'Number of Packets' as 7, and 'Segmented ATS' as Enable. The 'Secondary' column has identical settings but with 'Aux (LAN2)' as the preferred adapter. At the bottom, there are expandable sections for 'Transport Stream Configuration', 'Metadata Output', 'Audio Settings', and 'Other Settings'.

The encoder can produce a MPEG-2 single program transport stream (SPTS) containing H.264 video and AAC audio.

Adaptive Transport Stream Area

Check the **Enable Adaptive TS** check box to generate adaptive transport streams. Adaptive TS is required for HLS, and if this option is chosen, HLS Output can also be generated for the same encode stream.

Segment Duration

To specify a segment duration for transport streams to be used by HLS, check the **Enable** check box and enter the duration in seconds. This duration must be a multiple of the key frame interval, which is set on the H.264 Video tab. We recommend using a 10 second duration with a key frame interval of 60 frames for non-PAL video and 50 frames for PAL video. Segment Duration is automatically enabled when Segmented ATS is enabled. If also generating HLS Output on the same stream, the segment duration must be the same as the target duration on the HLS Output tab.

Fragment Duration

To specify a fragment duration for transport streams to be used by HSS, check the **Enable** check box and enter the duration in seconds. This duration must be a multiple of the key frame interval, which is set on the H.264 Video tab. We recommend that you use a 2 second duration with a key frame interval of 60 frames for non-PAL video and 50 frames for PAL video.

Transport Stream Markers Area

▼ Transport Stream Markers	
Adaptive TS RAI Only:	<input type="checkbox"/> Enable
Place RAI at Segment Boundaries:	<input type="checkbox"/> Enable
Place PMT/PAT only at Segment Boundaries:	<input checked="" type="checkbox"/> Enable
Place EBPs at Segment Boundaries:	<input checked="" type="checkbox"/> Enable
Audio EBPs at Fragment Boundaries:	<input type="checkbox"/> Enable

This area includes settings that specify how to mark segments in an adaptive transport stream.

Adaptive TS RAI Only

When creating ATS streams, Check the **Enable** check box to create a single TS stream (not an Apple Segmented stream), which uses only the TS RAI bit to mark segment boundaries and no further processing or segmentation is applied. If unchecked, the ATS stream will be segmented in discrete segments. Audio alignment is not available for this option.

Place RAI at Segment Boundaries

When creating ATS streams, including Apple Segmented data, the RAI (Random Access Indicator) option controls whether the initial IDR frame of a segment has the TS RAI bit set. If this option is not enabled, the RAI bit is never set.

Place PMT/PAT only at Segment Boundaries

When creating ATS streams, including Apple Segmented data, the PMT/PAT option controls whether a segment only has one PMT/PAT at the beginning. If this option is not enabled, PMT/PAT will also occur within the segment.

Place EBPs at Boundaries

When creating ATS streams, Check the **Enable** check box to place Encoder Boundary Points at fragment or segment boundaries, or both, depending on which have been enabled. This setting is automatically enabled if either Segmented ATS or HLS Output are enabled.

Audio EBPs at Fragment Boundaries

When creating ATS streams, Check the **Enable** check box to place fragment Encoder Boundary Points in the audio streams. The audio fragment duration is based upon the fragment duration.

Transport Stream Archive Area

▼ Transport Stream Archive

☒ Enable

In order to schedule archive events at a later time, select manual archive.

File name:

☐ Generate unique file name

☒ Single File Archive

☐ Segmented Archive

☐ Manual Archive

Duration:

0

minutes

Segmented ATS:

☐ Enable

The encoder can optionally save the transport stream to disk. Use this area to save the stream to a user-specified TS file by checking the check box and supplying a path and file name. When using a network drive, a fully qualified UNC path is required and the AnyRes Live Encoding Service must have permission to create directories and copy files to the destination (for instructions see “Setting Up AnyRes Live to Write to a Network Drive” on page 265).

Note:

If no path information is given, the file output is stored at C:\inetpub\wwwroot\encadmin\output on the encoder.

Check the **Generate unique file name** check box to append the current date and time to the file name, which will ensure that previous archive files will not be overwritten. If an output file is not required, then the file name should be left blank.

If Adaptive TS is enabled, archive files will be adaptive TS files.

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You must also choose what type of archive to save: Single File Archive, Segmented Archive, or Manual Archive:

Single File Archive

Choose **Single File Archive** if a single archive file, with the file name specified above, is to be written for the entire duration of the encode. Default archive is single file.

Segmented Archive

Choose **Segmented Archive** to have multiple archives created from a single encode. The length of the archive is specified by the duration in minutes. The filename specified above is the base name for each of the archive files that are created. A counter is incremented and appended to the name for each file that is created.

Manual Archive

Choose **Manual Archive** to set up the encoder to receive commands to start, stop and restart an archive during encoding. The context menu for a scheduled or active event will have a Manage Archives option that launches a dialog box to allow scheduling of manual archive events.

Segmented ATS

Check the **Enable** check box if archive files should be segmented. This option is only available if Adaptive TS is enabled. When this setting is enabled, the Segment Duration setting will also be enabled automatically.

Network Transfer Configuration Area

▼ Network Transfer Configuration	
<input checked="" type="checkbox"/> Enable Primary	<input checked="" type="checkbox"/> Enable Secondary
Destination Address: 224.0.0.1	224.0.0.1
Network Transfer Mode: ⁱ Multicast	Multicast
Preferred Adapter: Main (LAN1)	Aux (LAN2)
Port: 55514	55514
TTL: 8	8
Number of Packets: 7	7
Segmented ATS: <input type="checkbox"/> Enable	<input type="checkbox"/> Enable

Check the **Enable Primary** check box to unicast or multicast out compressed data encapsulated in a MPEG-2 Transport Stream. The Enable Primary check box enables multicast or unicast output. If you enable primary, you can optionally also check the Enable Secondary check box to enable a secondary multicast or unicast output. For the secondary output, settings that must match the primary output are display only.

Destination Address

Enter the destination IP address that the encoder should use to output the TS stream. The default address is 224.0.0.1. The IP address may be a unicast or a multicast address.

Network Transfer Mode

This display only field identifies whether the encoder will use Multicast or Unicast mode to output the TS stream. If the first octet of the destination address above is between 224 and 239, the mode is Multicast. All other addresses are Unicast mode.

Preferred adapter

Choose the preferred adapter to be used for sending output multicast or unicast packets. If this adapter is not available at encode start, the encoder will use another available adapter. For IP source, the encoder will first use Output 1 for primary or Output 2 for secondary, then the other output.

Port

Enter the UDP port number for the multicast or unicast TS stream.

TTL

This setting is the time-to-live value for the multicast or unicast socket.

Number of Packets

This setting indicates how many packets to group together in each multicast or unicast send. No more than 8 packets can be grouped together. Each MPEG-2 TS packet is 188 bytes.

Segmented ATS

Check the **Enable** check box if the multicast or unicast stream should be segmented. This option is only available if Adaptive TS is enabled. When this setting is enabled, the Segment Duration setting will also be enabled automatically.

Transport Stream Configuration Area

▼ Transport Stream Configuration

Program number:

1

PMT PID:

480

Video PID:

481

Audio PID:

482

Program Number

Enter the program number for the transport stream.

PMT PID

Enter the PMT PID for the transport stream.

Video PID

Enter the Video PID for the transport stream.

Audio PID

Enter the PID for the audio stream.

Metadata Output Area

▼ Metadata Output		
General Metadata:	<input type="checkbox"/> Enable	Bit rate: 0 bps PID: 1011
HLSTimed Metadata:	<input type="checkbox"/> Enable	Bit rate: 0 bps PID: 16
HLSTimed Metadata Timecode:	<input type="checkbox"/> Enable	Interval: 3 sec
SCTE-35 Output:	<input type="checkbox"/> Enable	PID: 1190
Teletext:	<input type="checkbox"/> Enable	Bit rate: 0 bps PID: 16
Subtitles:	<input type="checkbox"/> Enable	Bit rate: 0 bps PID: 16

General Metadata

For a general metadata stream, check the **Enable** check box, enter the bit rate in bits per second, and specify the PID.

HLS Timed Metadata

For an HLS timed metadata stream, check the **Enable** check box, enter the bit rate in bits per second, and specify the PID. HLS timed metadata inserts Metadata into the HLS stream and associates it with a particular video frame.

HLS Timed Metadata Timecode

For an HLS timed metadata stream, check the **Enable** check box, and then specify the interval in seconds.

SCTE-35 Output

Check the **Enable** check box to enable a SCTE-35 stream, and then specify the PID.

Teletext (IP only)

For a teletext stream, check the **Enable** check box, enter the bit rate in bits per second, and specify the PID. This setting is only available if teletext is enabled on the Video tab.

Subtitles (IP only)

For a subtitle stream, check the **Enable** check box, enter the bit rate in bits per second, and specify the PID.

Audio Settings Area

▼ Audio Settings

Audio PES Packing Duration (ms):

500

(0 = off, 1 = auto)

Audio PES AU Alignment:

☒ Enable

Audio PES Min AU:

0

(0..255)

Audio Segment Alignment:

☒ Enable

Audio Segment Alignment at Splice Points:

☒ Enable

Audio PES Packing Duration

Typical TS streams have a single audio frame in a PES packet. However, padding associated with this approach can severely impact low bit rate targets. For example, a 32 kb audio stream may have 20 kb of padding.

To significantly reduce padding, use this option to collapse multiple AAC audio frames in a single PES packet. The default is 500 ms of audio data per PES packet. If set to 0, packing is turned off. 1500 ms is the highest level of packing allowed.

This setting impacts any Adaptive TS or HLS stream.

Audio PES AU Alignment

Check the **Enable** check box to align audio AUs (audio access units) to PES packets. If not enabled, an audio AU may span more than one PES packet. The default is enabled.

Audio PES Min AU

Specify the minimum number of AUs per PES. This number of AUs will be included in a PES packet when possible. The default is 0.

Audio Segment Alignment

By default, audio segments are aligned. Uncheck the **Enable** check box to disable audio alignment at segment boundaries.

Audio Segment Alignment at Splice Points

Check the **Enable** check box to align audio only at splice points. We recommend this option if you are using Markers.

Other Settings Area

▼ Other Settings

Nielsen Watermarks: ☒ Enable Source: Left Input 1 (track_1) ▼

PCR Interval: 20 ms

PAT/PMT Interval: 250 ms

Minimum SCTE-35 Pre-roll: 0 ms

Video TS Rate: ☐ Override 1 kbps

Nielsen Watermarks

To include Nielsen audio watermarks as metadata output, check the **Enable** check box and choose an audio channel source that includes the watermarks. Before choosing this source, you must first enable an audio track on the H.264 Audio tab. The Nielsen watermarking data is carried on the Left speaker only. You must also enable HLS Timed Metadata in the Metadata Output area of the TS Output tab. The timed metadata for the Nielsen watermark tags will be inserted as a separate PID in the MPEG-2 transport stream, and will convey Station ID and Broadcast Time Code.

PCR Interval

The PCR interval is the nominal interval that the Program Clock Reference (PCR) is inserted into the transport stream, in milliseconds. The range is 1 to 100. The default is 20.

PAT/PMT Interval

The PAT/PMT interval is the nominal interval that the Program Association Table (PAT) and Program Map Table (PMT) are inserted into the transport stream, in milliseconds. The range is 50 to 1000. The default is 250.

Minimum SCTE-35 Pre-roll

The minimum SCTE-35 pre-roll is the delay, in milliseconds, added to the encoding pipeline to ensure SCTE-35 messages have a consistent minimum pre-roll. The range is 0 to 8000. Use caution when setting this value to anything but zero.

Video TS Rate

Check the **Override** check box and enter the rate in kilobits per second to specify the target video bit rate of the multiplexer. The range of this value is between 100% and 200% of the bit rate specified on the H.264 Video tab. This setting can be used to optimize bandwidth usage in some scenarios, but may cause errors in the multiplexer output if set too low. The default value is computed to use the minimum

bandwidth with a margin of safety. Use this setting to override the default only if you have a means of monitoring the output stream for the desired performance.

H.264 HLS Output Tab

Video Audio RTMP Output TS Output **HLS Output** H265 Output H265 Output RTP/3GP Output

☒ Enable HLS Output Copy Primary HLS Stream Values

▼ Streaming to Origin Server

Master Playlist Name(s): ☐ Include in master playlist

Stream Name: *

Storage URL: * Address:

Command: POST Authentication: Basic

Username: Password:

Publishing URL:

▶ Streaming to Redundant Origin Server

▶ Playlist File Publishing

▶ Audio-Only Stream Settings

▶ Segmentation Settings

▶ Encryption Settings

▶ Other Settings

When HLS Output is enabled, the encoder handles the ingest, encoding, and segmenting of the video stream so that the output is already in the correct format to be played by iOS devices. No external hardware is required to perform any additional segmenting. The output can go directly from the encoder to a web server or CDN for immediate playback.

For H.264 HLS Output, check the Enable HLS Output box. It is required that Adaptive TS be enabled on the TS Output tab. To configure HLS output, provide values for the following settings:

Master Playlist Name

For HLS streams, you can choose to create multiple master playlists. A master playlist is like a master index file that points to the individual HLS streams that are available when a playback device accesses that master playlist.m3u8 file. It may be desirable to have separate master playlists, such as one master playlist for Wi-Fi connections and a separate one for cellular connections. To have a stream included in a master playlist, check the **Include in master playlist** check box and provide that playlist name in the Master Playlist name field. If you would like to have a stream included in more than one master playlist, you can include multiple

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file names, separated by commas. For example: Wifistreams, cellstreams. For each HLS Output stream that is enabled, you can choose to have that stream included in none of the master playlists, one of them, some of them, or all of them.

The iOS player should be pointed to one of these master playlist files, and based on the detected bandwidth, it will play the appropriate stream. Over time, as bandwidth conditions change, the player will automatically be switched up to higher bandwidth streams or down to lower bandwidth streams to always deliver the best possible stream without buffering.

Stream Name

Specify the stream name, which is used as the name of the playlist (.m3u8) file and also the name for the folder where the encoder copies the segment (.ts) files. The stream name must contain valid characters for a file/folder name, and we do not recommend that you include spaces. Each stream on the encoder must use a different stream name. Use the drop-down list to choose each enabled stream to provide a unique name for each.

Note: If the Generate Unique Names setting is enabled, The encoder will generate unique names for segment files by appending the starting date and time for the encode to each segment file, so that stopping and restarting the encoder does not overwrite previous files. This also fixes some issues with file caching at the CDN.

If VOD mode is enabled, a second VOD mode playlist file will be generated with the same name except “_vod” will be appended to the name. When the encoder is stopped, the live playlist file will be replaced by the VOD playlist file.

Storage URL

To specify the URL for where the encoder will copy files, first choose the URL scheme from the drop-down list, and then enter the remainder of the URL address. Based on the address, you may be required to specify additional settings. If you choose an authentication option other than None, you must enter a username and password.

Using HTTP POST or PUT:

First choose http:// or https:// from the drop-down list, and then provide the remainder of the address in the second address field. Examples are:

example.com/apple/

example.com:8080/apple/

Choose POST in the Command drop-down list to specify that HTTP transfer will use a POST command to transfer a file, or choose PUT for the PUT command, which requires the HTTP handler to place the content in the specified location. The

target of the command is the destination file name. The destination is responsible for creating the destination file, including any folders used as part of the destination file name. To do this, the destination may require an HTTP handler. If using authentication, choose Basic or Digest authentication from the drop-down list, and enter the username and password.

Using FTP:

First choose ftp:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

anonymous@w2k8mediasrv/

Enter the FTP username and password in the Username and Password fields.

Using file transfer:

First choose file:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

w2k8mediaServer/AppleTest/

If using file transfer, the AnyRes Live Encoding Service must have permission to create directories and copy files to the destination (for instructions, see “Setting Up AnyRes Live to Write to a Network Drive” on page 265).

Using WebDAV:

First choose webdav:// or webdav-ssl:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

example.com/webdav/

example.com:80/webdav/

If using authentication, choose Basic or Digest authentication from the drop-down list, and enter the username and password.

Publishing URL

Enter the URL that the customer uses to access files. Publishing URL will be used to generate #EXTINF entries in the playlist file. To play back an individual stream, the iOS player should go to Publishing URL/streamname.m3u8. To play an adaptive stream, use Publishing URL/Master playlist file name.m3u8.

If this is left blank, relative URLs are used.

Streaming to Redundant Origin Server Area

Check the **Enable redundant origin server** check box to indicate that secondary URLs will be used for backup servers, and then specify the URLs.

Playlist File Publishing Area

▼ Playlist File Publishing

Ad Network URL: ☒ Enable (Only playlists will be sent to this URL)

Ad Network Storage URL: Address:

Command: Authentication:

Username: Password:

Ad Network URL

Check the **Enable** check box for Ad Network URL to specify a URL that will be used for playlists only. When publishing to the Ad Network URL, the playlist and master playlist files for the stream will be sent to this URL in addition to any primary or secondary URLs. See the Storage URL setting description for instructions.

Audio-Only Stream Settings Area

▼ Audio-Only Stream Settings

Generate Audio-Only Stream: ☒ AAC Enabled

☒ Use Existing JPEG:

☐ Auto-Generate JPEG

Audio JPEG Size: px width x px height

Audio JPEG Quality:

Generate Audio-Only Stream

If AAC Enabled is checked, the encoder will generate an additional audio-only stream. The video will be overlaid with an image file.

To specify the image file, choose Use Existing JPEG and provide the image file name. The image file field must include the UNC path to the existing image file either on the AnyRes Live encoder (recommended) or a network share accessible by AnyRes Live. We recommend that you use a 480x320 size image.

Note: If you do insert an image, the size of the image will be added to the stream bandwidth once every segment.

Alternatively, to have the encoder generate an image of the output video every segment, choose Auto-Generate JPEG. When this is enabled, the encoder will capture a frame from the video, once per segment, and insert that image into the audio-only stream. Note that this can affect the bandwidth of your stream. If you have 10 second segments, you will have 1 image per 10 seconds. With 5 second segments, you will have 2 images per 10 seconds, and so on.

The HLS audio-only stream does not count toward the output stream limit.

The HLS audio-only stream will have a stream name equal to the stream name on the output tab where it is enabled, with “_audio” appended. For example, if the stream name is wifi, and then the audio-only stream will have a stream name of wifi_audio.

The HLS audio-only stream will have its own directory and .m3u8, and will be referenced in the master .m3u8 file.

Audio JPEG Size

The default size for auto-generated image files for audio-only HLS streams is 480 x 320.

Audio JPEG Quality

The default quality setting for auto-generated image files for audio-only HLS streams is 25, range 0 to 100 where 0 specifies not to override the default quality.

Segmentation Settings Area

▼ Segmentation Settings

Target Duration:

9

sec

Duration Type in Playlist:

☒ Floating Point

☐ Integer ?

Segments in Playlist:

3

☐ Enable VOD Mode

Segments per Folder:

0

☐ Enable Sub-Folders

Segment Lag:

0

Segments to Keep:

0

☐ Remove old files

Settings in this area apply to all HLS Output streams.

Target Duration

Specify in seconds the target duration of each segment file. We recommend a value of 10 seconds.

Duration Type in Playlist

Choose **Floating Point** to specify segment durations in the playlist with floating point durations. We recommend using Floating Point to ensure compatibility with players using HLS version 3 or higher. Choose **Integer** to specify that all segments have integer durations as rendered in the playlist. In this case, the playlist may include an occasional variation in a segment duration to account for rounding error accumulation.

Caution: Integer durations may cause output to violate the HLS specification.

Segments in Playlist

Specify the number of segments to be held in the playlist file. After the limit is reached, for every new segment added, the oldest segment will be dropped, creating a sliding window. We recommend a value of 10 entries.

Enable VOD mode:

To create content for VOD instead of live streaming, check the **Enable VOD Mode** check box. In VOD mode, a second playlist file is generated that references all segments (instead of the last x segments), start index count is 0, and the encoder does not delete old segment files. When the encoder is stopped, the live playlist file is replaced by the VOD playlist file.

Segments per Polder

To generate a new subfolder after a certain number of segment files have been generated, check the **Enable Sub-folders** check box, and then specify the number of segment files per folder. This is particularly useful for VOD when saving a large number of segments. We recommend a value of 200.

Segment Lag

Specify the number of segment files to create before generating a playlist file. We recommend a value of 1 (0 for VOD mode). Setting this to 0 can cause playback issues, while setting this to a higher number increases the amount of time before a stream can be played, but only at the initial start of the stream.

Segments to Keep

To remove old segment files after a certain number of files have been generated, check the **Remove old files** check box, and then specify the number of files to keep. This creates a sliding window of segments stored on the server. Once the limit is reached, for every new segment, the oldest segment will be deleted.

This setting does not apply when using VOD mode.

Encryption Settings Area

The screenshot shows a web interface titled "Encryption Settings". It contains several configuration options:

- Encryption Type:** A drop-down menu currently set to "Random AES-128 Key".
- IV Publishing:** A drop-down menu currently set to "Disable".
- Key Rotation:** A checkbox labeled "Generate new key every" followed by a text input field containing "0" and a unit label "segments".
- Key File Name:** A text input field.
- Key Storage URL:** A section with three sub-fields:
 - *Address:** A drop-down menu set to "http://" followed by a text input field.
 - Authentication:** A drop-down menu set to "Basic".
 - Username:** and **Password:** text input fields.
- Key Publishing URL:** A section with a ***Address:** drop-down menu and a text input field.
- Redundant Key Storage URL:** A section with three sub-fields:
 - Address:** A drop-down menu set to "http://" followed by a text input field.
 - Authentication:** A drop-down menu set to "Basic".
 - Username:** and **Password:** text input fields.
- Redundant Key Publishing URL:** A text input field.

For encryption, first choose the Encryption Type from the drop-down list. For each encryption type, the following settings are available:

IV Publishing

The IV publishing setting specifies the available options for including the EXT-X-Key tag IV attribute in a manifest. The IV field is a 128 bit (16 byte) field. This can result in a large amount of data being sent over the network. Use the drop-down list to choose whether to disable inclusion of this field, or to specify when to include it based on your needs.

Include for every segment will include the IV for every TS segment in the playlist file.

Include with new keys will only include the IV field when an encryption key is included in a playlist file

Include for every segment in I-Frame Only will include the IV field in the playlist file only when an encryption key is present, and will include the IV field for every segment in the I-Frame manifest.

Random AES-128 Key or Sample Level Encryption Type

Key Rotation

For key rotation, check the check box and enter the interval of segments to generate a new key.

Key File Name

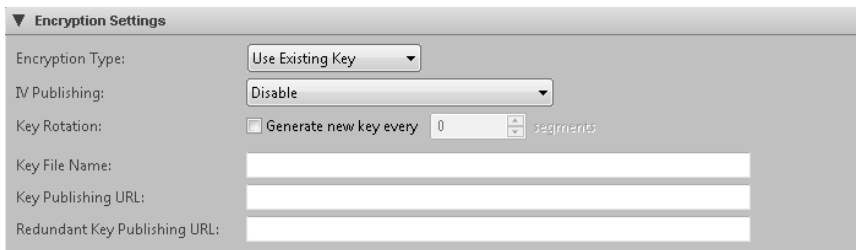
Use this field to enter the file name of the key that will be generated. You should not specify a path here.

Key Storage URL and Redundant Key Storage URL

Enter the URL where the key file will be placed. It is formatted the same as the storage URL for segments and index files. The key storage URL does not have to be same as the storage URL for the other files. You can publish the key to a different location from the files. You can also enter a redundant URL for a backup server if a redundant origin server has been enabled above.

Key Publishing URL and Redundant Key Publishing URL

Enter the key URL that is stored in the index file. This is the URL that the player will use to access the key. The key publishing URL is required. You can also enter a redundant URL for a backup server if a redundant origin server has been enabled above.

Use Existing Key Encryption Type


The screenshot shows the 'Encryption Settings' section of a configuration interface. It includes the following fields and controls:

- Encryption Type:** A dropdown menu currently set to 'Use Existing Key'.
- IV Publishing:** A dropdown menu currently set to 'Disable'.
- Key Rotation:** A checkbox labeled 'Generate new key every' followed by a numeric input field set to '0' and a unit dropdown menu set to 'segments'.
- Key File Name:** A text input field.
- Key Publishing URL:** A text input field.
- Redundant Key Publishing URL:** A text input field.

Key File Name

Enter the full path to the file containing the key. AnyRes Live must have network access to this file, which should have a file extension of .key. In addition, ensure that your Key Storage Servers have .key as a valid mime-type.

Key Publishing URL and Redundant Key Publishing URL

Enter the key URL that is stored in the index file. This is the URL that the player will use to access the key. The key publishing URL is required. You can also enter a secondary URL for a backup server if a redundant origin server has been enabled above.

PlayReady Encryption Type

▼ Encryption Settings

Encryption Type:

PlayReady

IV Publishing:

Disable

Key Rotation:

☐ Generate new key every

0

segments

PlayReady License Provider Information:

Manual Entry

KID:

Seed:

Content Key:

License Acquisition URL:

License Acquisition UI URL:

Domain Service ID:

Custom XML:

Note: PlayReady for HLS output is a Cisco-specific PlayReady implementation. Other PlayReady implementations will not work.

For Manual Entry and Generic license providers, PlayReady for HLS output uses the same settings available with PlayReady for HSS output. For setting descriptions, see “PlayReady Area” for HSS on page 122.

VCAS Encryption Type

▼ Encryption Settings

Encryption Type:

VCAS

IV Publishing:

Disable

Key rotation:

☐ Generate new key every

0

segments

VCAS Server URL:

VCAS Resource ID:

0

Verimatrix VCAS for Internet TV Encryption solution is supported by AnyRes Live. The following parameters are used in conjunction with that service. Before using this service, ensure you have contacted Verimatrix to obtain valid credentials.

Key Rotation

For key rotation, check the check box and enter the interval of segments to generate a new key.

VCAS Server URL

Enter the URL for the server. For a server using SSL and authentication, enter the username/password in the canonical URL form
 https://username:password@domain:port. SSL and authentication are not required for the server URL.

VCAS Resource ID

Enter the resource ID, which is a number that is a unique identifier for the stream. This identifier is for a specific content encryption key, typically a DTV channel number or VOD movie number.

Other Settings Area

Other Settings

Provide Absolute Time in Playlist:	Don't Include
Include Additional Time Data:	Don't Include
Allow Players to Cache Video Data:	Don't Include
Indicate Live or VOD Content:	Disable
Create I-Frame Playlists:	Don't Create
Add Endlist on Stop:	Disable
Cue Point Tag Insertion:	Don't Include
Support AirPlay:	Disable
Use Chunked Transfer for HTTP:	<input type="checkbox"/> Enable
Generate Unique Names:	<input checked="" type="checkbox"/> Enable
<input type="checkbox"/> Enable Authentication for	Akamai
Akamai Token Name:	<input type="text"/>
Akamai Token Key:	<input type="text"/>
Akamai TTL:	0

Settings in this area apply to all HLS Output streams.

Provide Absolute Time in Playlist

To include in the playlist file the time when a TS segment was generated, choose Use wall time to include date/time information generated from the current system time of the encoder. Choose Use LTC time to generate the date/time information from LTC time, if available.

Include Additional Time Data

This setting, unless set to Don't Include, will add a custom field to the EXTINF tag for each TS segment in the playlist. This field provides additional time information for the segment, which can be used to help the client player decide which segment to transition to when changing bit rates and using hardware decryption. Choose from the drop-down list to include either LTC (if available) or PTS for the first video frame of the TS segment.

Allow Players to Cache Video Data

This setting, unless set to Don't Include, will include a EXT-X-ALLOW-CACHE tag in the playlist that indicates whether the player may cache video segments for later playback. Choose Yes to allow the player to cache TS segments, or choose No to explicitly indicate that the client player is not allowed to cache TS segments. Don't Include will not include the tag, which implies that caching is not allowed.

Indicate Live or VOD Content

Choose Enable to include the EXT-X-PLAYLIST-TYPE tag in the playlist file. This tag indicates whether the playlist type is VOD or EVENT, based on the type of playlist being generated. Disable indicates that the tag will not be included.

Create I-Frame Playlists

Choose Create to create I-frame Only playlists in addition to other playlists.

Add Endlist on Stop

To add an end-of-stream indicator to the playlist when the encode is stopped, choose For live and VOD manifest if the indicator should be added for both live and VOD. Choose For VOD manifest only if the indicator should only be added for VOD. Disable indicates that the indicator will not be included. Note that VOD options only apply when VOD mode is enabled.

Cue Point Tag Insertion

To include cue point tags in the playlist, choose Include IN and OUT tags if CUE-IN and CUE-OUT markers should be included. Choose Include IN/OUT/CONT tags for Freewheel if CUE-IN, CUE-OUT, and CUE-OUT-CONT tags should be included.

Support AirPlay

Choose **Enable** to include RESOLUTION and CODEC tags in the playlist. **Disable** indicates that these tags will not be added. Some older HLS players do not play back streams with these tags present in the playlist.

Use Chunked Transfer for HTTP

Contact your CDN for the correct setting. The default is False.

Generate Unique Names

Check the **True** check box to specify that the encoder will generate unique names for segment files and folders. The file and folder names will include the starting date and time for the encode. Generating unique names prevents name collision and accidental removal of old streams when starting a new streams, and also prevents unintentional referencing of old cached files when encoding a new stream. The default is True.

Authentication

Check the **Enable Authentication** check box and choose the CDN (content distribution network) from the drop-down list to enable authentication with a web server. Each file is authenticated as it is sent to the CDN. Authentication is only valid when using HTTP to send files. Custom authentication parameters will display. For Akamai, the token name and key will be provided by Akamai, and TTL is the time-to-live of the encrypted token. A TTL value of 30 seconds is reasonable.

H.264 HSS Output Tab

Video Audio RTMP Output TS Output HLS Output **HSS Output** HD5 Output RTP/SGP Output

☒ Enable HSS Output Copy Primary HSS Stream Values

▼ Streaming to Origin Server

Server: *

Publishing Point: * e.g., smooth/S6000G2.isml/stream(1)

EventID:

Stream Manifest File: Leave blank to have LSM auto-generate manifest

Username:

Password: Password will not be displayed

Publisher Retry Count: 5

Publisher Retry Delay: 20 sec

▶ Streaming to Redundant Origin Server

▶ Additional Settings

▶ PlayReady

▶ HSS Metadata

Streaming to Origin Server/Redundant Origin Server Areas

Check the **Enable HSS Output** check box to enable H.264 HSS output to an Origin server with IIS Media Services installed. You must have previously checked the **Enable Adaptive Streaming** check box on the H.264 Video tab. You can also use an additional redundant origin server. Click **Copy** to populate the stream settings from the settings of the primary stream.

Server

Enter the name of the server. This can be an alias, a fully qualified domain name, or an IP address of the server. The default port is 80.

Publishing Point

Enter the stream ID to identify the video profile for this publishing point. The format for this field is: *publishingPointName.isml/stream(id)*.

Event ID

Optionally specify an Event ID string that will be included in the publishing point URL. This Event ID will allow the publishing point to be reused.

Stream Manifest File

Enter the name and location of the stream manifest file to be used. Leave this field blank to have the LSM auto-generate the manifest file.

Username and Password

If your Server requires authentication, enter the username and password. The password characters will not be displayed.

Publisher Retry Count

Set to the number of times the encoder will retry connecting to the configured server.

Publisher Retry Delay

Set to the delay in seconds between retry attempts.

Additional Settings Area

▼ Additional Settings

Target Fragment Duration:

2000

ms

Include Audio Tracks:

* ☐ track_1

Video/Audio Delay:

0

ms

Custom Video Parameter:

Max Size Override:

☐ Enabled

Width:

0

Height:

0

Low Latency Settings:

☐ Low Latency Enabled

Video Frames/Fragment:

10

Audio Frames/Fragment:

15

Target Fragment Duration

Specify (in milliseconds) the target duration for HSS fragments. This setting should normally be the same as the key frame interval set on the H.264 Video tab. Default key frame interval is 60 frames for non-PAL video and 50 frames for PAL video, which is 2000 ms of video. Although this duration setting appears on the HSS Output tab for each encode stream, there is only one global setting and it affects all H.264 HSS streams.

Include Audio Tracks

Check any combination of available audio stream check boxes to include audio for a given encode stream in the HSS output. One or more check boxes only appear if audio has been enabled for the corresponding encode stream on the H.264 Audio tab. At least one HSS stream must have audio included. A particular audio stream can only be included on one HSS stream.

Video/Audio Delay

Enter the delay in milliseconds for the output streams. Although this delay setting appears on the HSS output tab for each encode stream, there is only one global setting and it affects all H.264 HSS streams.

Custom Video Parameter

The text in this field will be inserted as a custom attribute in the live server manifest. An example use for this field is for multiple camera angles. HSS streams are uniquely identified by a quality level which is essentially the bit rate. Two streams of the same bit rate cannot be distinguished without custom attributes. This field provides the custom attribute to identify this stream. For multiple camera angles, this attribute could identify the specific video input.

An example format for this field is:

```
<param name="cameraAngle" value="coach-cam" valuetype="data" />
```

Thus two different encoders could provide two different camera angles. Each encoder has the same number of streams and bit rates for those streams. However, one encoder has for each stream, for example, a custom video parameter of:

```
<param name="cameraAngle" value="main" valuetype="data" />
```

and the other encoder has for each stream a custom video parameter of:

```
<param name="cameraAngle" value="alternate" valuetype="data" />
```

Max Size Override

Check the **Enabled** check box to override the maximum output video size, and specify the new maximum width and height for this stream.

Use this setting to handle anamorphic/widescreen video. For example, 16x9 content is often anamorphically resized to 720x480 for delivery over SDI. Instead of resizing this 720x480 video back to 848x480 prior to compression, simply compress the content as is and set the max size override to 848x480. You should only need to do this to your top resolution stream. On playback, the Silverlight player will handle reversing the anamorphic processing and will display the content resized back to its original 16x9 848x480 size.

Low Latency Settings

For H.264 HSS output, a Low Latency option is available. This option relies on Low Latency HSS updates to IIS and the client Player.

The typical latency for HSS is upwards of 16 seconds and is attributed to the underlying fragmentation and buffering of fragments from end to end. For low latency HSS, smaller fragments are used and are based on a number of frames versus fragmentation on time/GOP boundaries. Currently, the HSS Low Latency IIS/Player have been optimized for video fragments of 10 frames and audio fragments of 15 frames.

To use low latency, check the **Low Latency Enabled** check box and specify the frames per fragment for video and audio.

PlayReady Area

▼ HSS PlayReady

☐ Enable PlayReady

Encryption Type: Manual PlayReady

KID:

Seed:

Content Key:

License Acquisition URL:

License Acquisition UI URL:

Domain Service ID:

Custom XML:

PlayReady provides encryption support for HSS Output.

Check the **Enable PlayReady** check box to use HSS PlayReady. To complete the rest of the PlayReady setup, you will need a third-party PlayReady License provider. This provider may provide you with setup information which you can manually enter in the boxes provided. Some third-party PlayReady platforms have been integrated to automatically retrieve the setup information. To use the third-party approach, you will also need a business relationship with the third party.

If you choose an Encryption Type for a third-party provider such as BuyDRM, additional settings will be available at the time of event creation.

If you choose Manual Entry, provide values for the following settings:

KID

Enter the Key ID. This value is required. The KID must be supplied as a GUID.

GUID should be entered in the form of: **{cab50618-c7e1-4e84-b683-c086d15170eb}**

Seed

The encryption seed. *Either* a seed or a content key is required.

Content Key

The encryption content key. *Either* a seed or a content key is required.

License acquisition URL

Enter the URL. A License acquisition URL is required.

License acquisition user interface URL

This is an optional field. If supplied, enter the URL.

Domain Service ID

This is an optional field. If supplied, enter a Domain Service ID. The ID may be supplied as either a GUID or Base64 string.

GUID should be entered in the form of: **{cab50618-c7e1-4e84-b683-c086d15170eb}**

An example Base64 string is: **GAa1yuHHhE62g8CG0Vfw6w==**

Custom XML

This is an optional field. The PlayReady license provider may specify custom data for the PlayReady header. If supplied use this custom XML field to include it.

HSS Metadata Area

▼ HSS Metadata	
User Track 1:	<input checked="" type="checkbox"/> Enabled
Name:	Metadata
Sparse track:	<input checked="" type="checkbox"/> Enabled
Subtype:	DATA
FourCC:	plain text
Manifest output:	<input type="checkbox"/> Enabled
Bit rate:	1000
<hr/>	
SCTE-35 Track:	<input checked="" type="checkbox"/> Enabled
Name:	AdMarkers
Sparse track:	<input checked="" type="checkbox"/> Enabled
Subtype:	SCTE35
FourCC:	plain text
Manifest output:	<input checked="" type="checkbox"/> Enabled
Bit rate:	1000
<hr/>	
CC/Subt... Track:	<input checked="" type="checkbox"/> Enabled
Name:	captions
Sparse track:	<input type="checkbox"/> Enabled
Subtype:	CAPT
FourCC:	TTML/DFXP
Manifest output:	<input type="checkbox"/> Enabled
Bit rate:	1000

HSS output offers the ability to have three metadata tracks that may be enabled independently. One track is available for user-supplied data, one track is used for

Markers (SCTE-35) and the final track is used for Closed Captioning/Subtitles. For captions, a source for captions must be specified on the H.264 video tab. For subtitles, teletext subtitles must be enabled on the H.264 Video tab.

All tracks are HSS "Text" tracks.

For each track, accept or alter the following settings:

Name

Each track can be given a name. For CC/Subtitles, there is special processing in the Silverlight Smooth Streaming Media Element (SSME) player for handling text tracks with the name "captions," so we recommend leaving this track name as captions.

Sparse Track

The Sparse track check box specifies that samples into these tracks do not happen at regular intervals. The User and SCTE-35 tracks are Sparse, and the CC/Subtitle track should not be Sparse. Currently, all Sparse tracks have "video" as their parent track.

Subtype

Accept or modify the predefined subtype that will be passed through to the manifest. The CC/Subtitle track should always be CAPT regardless if the text is captions or subtitles.

FourCC

The optional FourCC field is simply passed through to the manifest. For the CC/Subtitle track, FourCC should be set to TTML/DFXP for the encoder to generate DFXP-based metadata. This track may also be set to plain text. The default is TTML/DFXP.

Manifest output

The manifest output check box specifies whether the metadata track samples are to be reflected in the manifest. For the CC/Subtitle track, this option generally should not be enabled unless some debugging is desired. With the Manifest Output checked, the sample data for the track is base64 encoded and included in the manifest. Over time this can create an excessively large manifest and thus in general it is best to not enable this option.

Bit rate

The Bit rate entry specifies an estimate of how much bandwidth in bits per second the metadata stream will use. If unknown, leave this at the default. The range is 1 to 10000.

H.264 HDS Output Tab

Enable HDS Output

Streaming to Origin Server

Stream Name: * Stream1

Streaming to Primetime Origin: ☒ Enable AES Security Token:

Origin Server(s): *

Additional Settings

Master Manifest File Name: *

Target Fragment Duration: * 4 sec

Fragments per Segment: * 1

Segments to Keep: * 450

Closed Captions: ☐ Enable

Splice Point Insertion: ☐ Enable

Prepare Stream for JIT: ☐ Enable

For H.264 HDS Output, check the Enable HDS Output check box.

Streaming to Origin Server Area

To configure HDS output, provide values for the following settings:

Stream Name

Enter the stream name. Stream name must contain valid characters for a file/folder name, and spaces are not recommended. Each stream in the preset must use a different stream name. Use the stream drop-down list to choose each enabled stream to provide a unique name for each.

Streaming to Primetime Origin

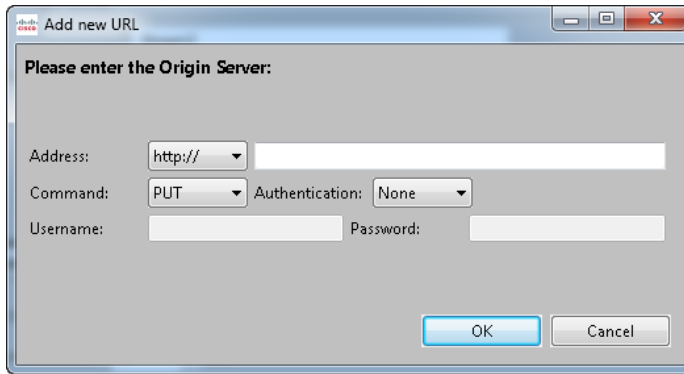
Check the **Enable** check box to publish HDS output to the Adobe Primetime origin server.

AES Security Token

If Adobe has provided you with a 128-bit AES security token, you can enter the token in this field. This token will be appended to the outgoing HTTP request header.

Origin Server(s)

Up to eight origin servers can be configured. To add the information to identify an origin server, click **Add** to display the Add New URL dialog box.

A screenshot of a Windows-style dialog box titled "Add new URL". The dialog box has a light gray background and a blue title bar. Inside, the text "Please enter the Origin Server:" is displayed. Below this, there are four input fields: "Address:" with a dropdown menu showing "http://" and a text box; "Command:" with a dropdown menu showing "PUT"; "Authentication:" with a dropdown menu showing "None"; and "Username:" and "Password:" text boxes. At the bottom right, there are "OK" and "Cancel" buttons.

First, choose the URL scheme from the drop-down list, and then enter the remainder of the URL address. For Authentication, if you choose the None option, the Username and Password fields are dimmed and ignored by the encoder. If you enabled Streaming to Primetime Origin, only HTTP PUT is available.

Using HTTP POST or PUT:

First, choose http:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

example.com/hds/

example.com:8080/hds/

Choose POST in the Command drop-down list to specify that HTTP transfer will use a POST command to transfer a file, or choose PUT for the PUT command, which requires the HTTP handler to place the content in the specified location. The target of the command is the destination file name. The destination is responsible for creating the destination file, including any folders used as part of the destination file name. To do this, the destination may require an HTTP handler.

Using file transfer:

First, choose file:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

w2k8mediaServer/hds/

If using file transfer, the Encoding Service must have permission to create directories and copy files to the destination (for instructions, see "Setting Up AnyRes Live to Write to a Network Drive" on page 265).

CISCO LINEAR STREAM MANAGER

After specifying the settings of the origin server, click **Add** to add it to the list. To edit an origin server after it has been added, select the server address, and then click **Edit** to display the Update Origin Server dialog box, modify settings, and click **Update**. To remove an origin server from the list, select the server address, and then click **Remove**.

Additional Settings Area

Except for Master Manifest File Name, each setting in this area must be set on stream 1, and applies to all enabled HDS streams.

Master Manifest File Name

Enter the base name that will be used for naming index and segmented files on the server. These files will include the master manifest file name, a segment index, and an appropriate file extension. The master manifest file name must contain valid characters for a file/folder name, and spaces are not recommended.

Target Fragment Duration

Specify (in seconds) the target duration for HDS fragments. This setting must be a multiple of the key frame interval set on the H.264 Video tab. For segmented outputs, segment size must be an integral number of seconds. A good size fragment is small enough to send over the Internet, and long enough to prevent multiple calls to the server. We recommend a value of 4 seconds, but the value may be altered depending on the GOP size. The range is 2 to 60 seconds.

Fragments per Segment

Specify the number of fragments per segment. A value of 1 indicates that the entire segment consists of one fragment. The range is 1 to 30, and the default is 1.

Segments to Keep

To remove old segment files after a certain number have been generated, specify the number of segments to keep. This creates a sliding window of segments stored on the server. Once the limit is reached, for every new segment, the oldest segment will be deleted. A segment consists of two files, one .f4f and one .f4x. A value of 0 indicates that no segment files will be deleted from the server. The range is 0 to 1000, and the default is 450.

Closed Captions

Check the **Enable** check box to enable closed caption data to be streamed to the origin server. A source for captions must be specified on the H.264 Video tab.

Splice Point Insertion

Check the **Enable** check box to insert splice cue points. These cue points can be used by the player to show an ad instead of the current video.

Prepare Stream for JIT

Check the **Enable** check box to change HDS configuration settings to those that allow the HDS stream to be processed by Just-In-Time packaging.

H.264 DASH Output Tab

The screenshot shows a software configuration window with a tabbed interface. The 'DASH Output' tab is selected and highlighted in green. At the top, there is a row of tabs: Video, Audio, RTMP Output, TS Output, HLS Output, HDS Output, HDS Output, DASH Output, and RTP/SGPP Output. Below the tabs, there is a section with a checked checkbox labeled 'Enable DASH ISO-BMFF Output'. Underneath this is a section titled 'Streaming to Origin Server' with a downward arrow. This section contains four input fields: 'Stream Name:' with the value 'Stream1', 'MPD File Name:' with the value 'Example.mpd', 'Storage URL:' with the value 'http://example.com', and 'Publishing URL:'. To the right of the 'Storage URL' and 'Publishing URL' fields are three buttons: 'Add...', 'Edit...', and 'Remove'. At the bottom of the window, there are two expandable sections: 'Audio/Subtitles' and 'Additional Settings', both with right-pointing arrows.

For H.264 DASH Output, check the **Enable DASH ISO-BMFF Output** check box. To configure DASH output, provide values for the following settings:

Streaming to Origin Server Area

Stream Name

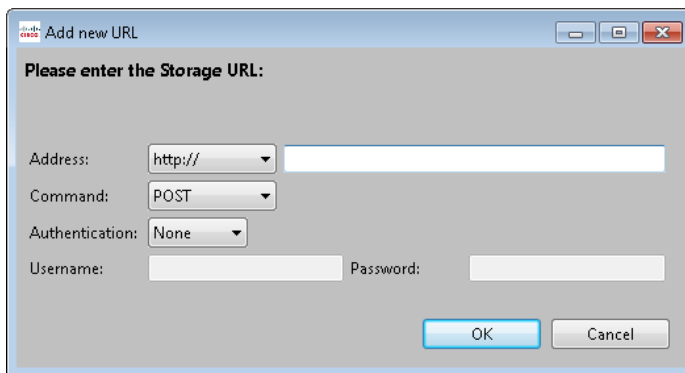
Enter the stream name, which is used as the base for all the output for the stream. Stream name must contain valid characters for a filename, and spaces are not recommended. Each stream on the encoder must use a different stream name. Use the Stream drop-down list to choose each enabled stream to provide a unique name for each.

MPD File Name

Enter the MPD (Media Presentation Description) file name. This file name does not include a path. If you do not include the .mpd extension, it will be appended when you save the preset.

Storage URLs

To specify one of up to four URLs for where the encoder will copy DASH files, click the **Add** button to display the Add new Storage URL dialog box.

The image shows a dialog box titled "Add new URL" with a Cisco logo in the top-left corner. The dialog has a title bar with standard window controls. The main content area is titled "Please enter the Storage URL:". It contains four fields: "Address:" with a dropdown menu showing "http://" and a text input field; "Command:" with a dropdown menu showing "POST"; "Authentication:" with a dropdown menu showing "None"; and "Username:" and "Password:" text input fields. At the bottom right, there are "OK" and "Cancel" buttons.

Choose a URL scheme from the drop-down list, and then enter the remainder of the URL address. Based on the address, you may be required to specify additional settings.

Using HTTP POST or PUT:

First, choose `http://` or `https://` from the drop-down list, and then provide the remainder of the address in the second address field. For example:

`example.com/example/`

`example.com:8080/example/`

Choose POST in the Command drop-down list to specify that HTTP transfer will use a POST command to transfer a file, or choose PUT for the PUT command, which requires the HTTP handler to place the content in the specified location. The target of the command is the destination file name. The destination is responsible for creating the destination file, including any folders used as part of the destination file name. To do this, the destination may require a HTTP handler.

If using HTTP authentication, choose Basic or Digest from the drop-down list, and enter the username and password. Custom authentication parameters will display.

Using file transfer:

First, choose `file://` from the drop-down list, and then provide the remainder of the address in the second address field. For example:

`w2k8mediaServer/example/`

If using file transfer, the Encoding Service must have permission to create directories and copy files to the destination (for instructions, see “Setting Up AnyRes Live to Write to a Network Drive” on page 265).

Using webdav:

First, choose webdav:// or webdav-ssl:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

example.com/example/

example.com:8080/example/

If using HTTP authentication with webdav-ssl, choose Basic or Digest from the drop-down list, and enter the username and password. Custom authentication parameters will display.

Publishing URL

To specify up to two URLs where media files may be retrieved by the end client, first click the **Add** button to display the Add new Publishing URL dialog box. Enter the HTTP URL address that the customer uses to access files. Next, click **OK**. To edit or delete a previously entered URL, select the URL, and then click **Edit** or **Remove**. If the Publishing URL list is left blank, relative URLs are used.

Audio/Subtitles Area

▼ Audio/Subtitles

Selected Audio Streams

<input type="checkbox"/>	Name	Index	Codec	Bit Rate	Sample Rate	Language
<input checked="" type="checkbox"/>	eng_audio	1	AAC	96 kbps	48 KHz	eng

Selected Subtitles

<input type="checkbox"/>	Type	Track ID	Language
<input type="checkbox"/>	CC1	-	eng
<input checked="" type="checkbox"/>	CC2	-	eng

Available Audio Streams

<input type="checkbox"/>	Name	Index	Codec	Bit Rate	Sample Rate	Language
<input type="checkbox"/>	spa_audio	2	AAC	192 kbps	48 KHz	spa
<input checked="" type="checkbox"/>	fra_audio	3	AAC	192 kbps	48 KHz	fra

Available Subtitles

<input type="checkbox"/>	Type	Track ID	Language
<input type="checkbox"/>	CC3	-	eng
<input checked="" type="checkbox"/>	CC4	-	eng

Subtitle Output Format: WebVTT

DASH output can contain multiple audio streams and subtitle tracks.

On the right side of this area are tables of available audio and subtitles that may be included in the DASH output.

Note: DASH subtitle support is an EFT only feature. For further assistance, contact Cisco support.



Prerequisites

Before selecting audio and subtitles to include in the output, it is necessary to enable each of the audio streams and subtitle tracks so that they will be displayed in the tables of available audio and subtitles. Ensure that at least one audio stream is enabled. For subtitle tracks to be available in the Subtitles table, either Teletext or 608 Closed Caption tracks must be enabled in the Captioning area of the H.264 Video tab.

Adding Audio Streams and Subtitle Tracks

The Available Audio Streams table lists all enabled H.264 AAC streams. The name of the audio stream is the track ID entered on the H.264 Audio tab, and the index is the audio stream number from the H.264 Audio tab. Each entry in the table also displays the codec, bit rate, sample rate, and 3-letter language identifier.

The Available Subtitles table lists all subtitle tracks that have been enabled in the Captioning area of the H.264 Video tab. These subtitle tracks can be 608 Closed Captions or Teletext tracks, but not a combination of both. Each teletext entry in the table displays the unique Track ID and 3-letter language identifier entered on the Input tab. Closed Caption entries in the table do not have a Track ID, and the language identifier defaults to “eng” because the language cannot be detected. The subtitle language identifier can be modified by clicking the value and choosing another language identifier from the drop-down list.

To add an audio stream or subtitle track to the Selected table, check the check box next to the stream or track and click the left arrow button , which will remove it from the Available table and place it in the Selected table. To remove a stream or track from the Selected table, check its check box and click the right arrow button . The check box in the table header selects all streams or tracks in that table.

Subtitle Output Format

If you are including subtitle tracks in the DASH output, choose the WebVTT or the SMPTE-TT output format from the drop-down list. Ensure that your client player supports the chosen format.

Additional Settings Area

▼ Additional Settings

Segment Duration: 2 sec

Remove Old Segments: ☒ Enable

Time Shift Window: 600 sec

MPD Update Interval: 10 sec

Use Segment Timeline: ☐ Enable

Initialization Segment Template: \$RepresentationID\$-init.mp4

Media Segment Template: \$RepresentationID\$-\$Number\$.mp4

Segment Duration

Enter the segment duration in seconds. The segment duration must be a multiple of the key frame interval. The range is 0 to 100. The default is 2 seconds.

Remove Old Segments

Check the **Enable** check box to remove old segments from origin servers. The default is Enable.

Time Shift Window

This value is used to ensure that the generated DASH MPD contains references to the correct number of segments. When Remove Old Segments is enabled, this value is also used to determine the interval after which old segments should be removed from the origin server. The range is 0 to 200000 seconds. The default is 600 seconds.

MPD Update Interval

Enter the frequency at which the MPD is updated on the origin server, in seconds. The initial MPD will be written at the end of the first interval. The range is 1 to 86400 seconds. The default is 10 seconds.

Use Segment Timeline

Check the **Enable** check box to include explicit information on each media segment in the DASH MPD. The default is disabled. This setting affects the macros to use with the Media Segment Template string.

Initialization Segment Template

Enter the string that is used to determine the filenames to which the initialization segments for each stream are stored on the origin server. These filenames will be referenced in the DASH MPD. To insert a macro in the template at the cursor location, click **Insert Macro**, choose a macro from the dialog box, and click **OK**. At least one macro is required, or the initialization segment will be overwritten. The default template is \$RepresentationID\$-init.mp4.

Media Segment Template

Enter the string that is used in the DASH MPD to determine the filenames to which the media segments for each stream are stored on the origin server. To insert a macro in the template at the cursor location, click **Insert Macro**, choose a macro from the dialog box, and click **OK**. At least one macro is required, or the media segment will be overwritten, and you must use either the \$Number\$ or the \$Time\$ macro, but not both. The default template is \$RepresentationID\$_\$Number\$.mp4.

If **Use Segment Timeline** is enabled, the \$Time\$ macro must be included, and the \$Number\$ macro is not supported.

H.264 RTP/3GPP Output Tab

The screenshot shows the 'RTP/3GPP Output' tab selected in the Cisco Linear Stream Manager. The interface includes a 'Copy' button and 'Primary RTP/3GPP Stream Values' text. Under the 'Streaming to RTP/3GPP Origin Server' section, the following settings are visible:

- Enable RTP/3GPP Output:** Checked
- RTP Mode:** Multicast/Unicast RTP Streaming (dropdown)
- RTP Output Address:** * (text field)
- Network Transfer Mode:** Unicast (dropdown)
- Preferred Adapter:** Default selected by route (dropdown)
- Push Port:**
 - RTP Audio: 2 (spin box)
 - RTP Video: 2 (spin box)
- Stream Name:** (text field)
- MTU Size:** 0 (spin box)

Below this section is a collapsed tab labeled 'Streaming to Redundant RTP/3GPP Origin Server'.

Cell phones have limited bandwidth and capabilities, and in addition wireless networks may place additional restrictions on network resources. We recommend verifying these capabilities in the network and setting video and audio encoding parameters on the H.264 Video and Audio tabs accordingly.

Recommendations for video streaming are as follows:

Encoding Parameter	Recommended Setting for RTP/3GPP Output
Profile	Baseline
Level	1.1 to 1.3
Bit Rate	64 to 192 kbps
Key Frame Interval	10 to 30 frames
Output Resolution	176x144 to 320x240

Output Frame Rate	5 to 15 fps
Number of B Frames (Advanced Compression Settings)	None

Recommendations for audio streaming are as follows:

Encoding Parameter	Recommended Setting for RTP/3GPP Output
Bit Rate	12 to 48 kbps for Low complexity 16 to 48 kbps for HE v1 complexity 16 to 32 kbps for HE v2 complexity
Sample Rate	Up to 44.1kHz

Choose the stream for RTP/3GPP output and check the **Enable Streaming** check box to push content to a RTP/3GPP server. RTP/3GPP output cannot be enabled at the same time as other output formats.

Streaming to RTP/3GPP Origin Server Area

To use an optional second server, check the **Enable Streaming** check box in the Streaming to Redundant RTP/3GPP Origin Server area. Next, choose the RTP mode, which determines the other available settings.

When you are streaming via RTP unicast or multicast instead of sending to an RTSP server, the parameters that you define are slightly different, and in some cases, may have different meanings. Although both RTP and RTSP streaming tend to go to the same type of clients, and use the same codecs, they do so somewhat differently, thus we document these separately.

RTP Mode

Content can be pushed in one of two ways:

Using RTSP Signaling:

To use RTSP signaling to stream to a Darwin server, choose the Streaming to an RTSP Server option from the RTP Mode drop-down list. In this case, the SDP will be transferred in-band to the server and stored in the filename specified in the Stream Name field. The RTSP session will also negotiate RTP ports for media transfer. See *RTSP Settings* below for setting descriptions.

Using raw RTP traffic:

Choose Multicast/Unicast RTP Streaming from the RTP Mode drop-down list. Raw RTP packets will be transferred over the defined ports. See *RTP Settings* below for setting descriptions.

RTSP Settings

RTSP Server

Enter the name of the RTSP server. This can be an IP address of the server or any type of name that can be resolved to an IP address by AnyRes Live.

Preferred Adapter

Choose to have the default adapter selected by route, or choose the preferred adapter to be used for streaming.

Push Port

If using RTSP signaling, enter the RTSP port number. The default is 554.

Stream Name

The default name of the stream is displayed. If desired, enter a unique stream name. The associated SDP file will be created once the encode has started. By convention, stream names should always be a simple file name (no path components) and a .sdp extension, for example, *live_feed.sdp*. After streaming has started, you can choose Download SDP Files from the encoder context menu of the encoder monitor to download a copy of the SDP file. However, the SDP file has limited usefulness when streaming to an RTSP server.

Check the **Enable Authentication** check box to enable authentication if your RTSP server requires it. Enter the username and password and click **Apply** to add them to your configuration. The password characters will not be displayed.

MTU Size

Set the maximum packet size for RTP. The default is 1500.

RTP Settings

RTP Output Address

Enter the destination IP address. The IP address may be a unicast or a multicast address. For unicast addresses, use any type of name that can be resolved to an IP address by AnyRes Live.

Network Transfer Mode

Display only. This field identifies whether the encoder will use Multicast or Unicast mode to output the stream. If the first octet of the RTP IPV4 output address above is between 224 and 239, the mode is Multicast. All other addresses are Unicast mode.

Preferred Adapter

Choose to have the default adapter selected by route, or choose the preferred adapter to be used for streaming.

Push Port

If streaming raw RTP packets, enter the audio and video port numbers. These port numbers must be even, and the audio and video streams must be on different ports.

Stream Name

The default name of the stream is displayed. If desired, enter a unique stream name. By convention, stream names should always be a simple file name (no path components) and a .sdp extension, for example, *live_feed.sdp*. The associated SDP file will be created once the encode has started. After streaming has started, you can choose Download SDP Files from the encoder context menu of the encoder monitor to download a copy of the SDP file. Choose the location to download the SDP file, which can be used by client software to view the stream. If any encoding parameters are changed, a new SDP file must be created and downloaded to view the modified stream.

MTU Size

Set the maximum packet size for RTP. The default is 1500.

TTL

This setting is the time-to-live value for the multicast socket. For unicast streams, this setting is not available.

H.263 Tab

H.263 Video Tab

The screenshot shows the 'H.263' tab selected in the Cisco Linear Stream Manager. The interface includes tabs for 'Input Video', 'Input Audio', 'VC-1', 'VP6', 'H.264', and 'H.263'. Below the tabs, there is a 'Stream 1' dropdown and a 'Stream count: 4' field with an 'Apply' button. The 'Video' tab is active, showing a checkbox for 'Enable H.263 Encoding' which is checked. Below this is a section for 'H263 Encoding Parameters' with various settings: Profile (Profile 0), Level (10), Bit rate (64 kbps), Key Frame Interval (60 frames), Buffer Size (500 ms), Output Resolution (Custom), Cropping (Left: 0, Right: 1920, Top: 0, Bottom: 1080), Resizing (480 x 480), Resize Mode (Default), Resize Algorithm (Default), Output Frame Rate (1/2x), and Pixel Aspect Ratio (Override to: 1:1). At the bottom, there are expandable sections for 'Pre-Processing', 'Advanced Compression Settings', and 'Captioning'.

Note: H.263 format is not recommended for HD sources.

On the H.263 Video tab, first enable H.263 encoding, and then view or modify the following settings:

Profile

The profile for video encoding specifies a subset of the H.263 syntax required for decoding the stream. The supported profiles are 0 and 3, as defined by the ITU-T Recommendation.

Profile 0—Defined herein to provide a profile designation for the minimal "baseline" capability of H.263.

Profile 3—Also known as Interactive and Wireless Streaming Profile, is defined herein to provide enhanced coding efficiency performance and enhanced error resilience for delivery to wireless devices. In addition to the baseline, Advanced INTRA Coding, Deblocking Filter, Slice Structured Mode and Modified Quantization are supported.

Level

This setting specifies constraints for the encoding parameters.

Level 10—Bit rate up to 64 kbps, output resolutions: QCIF (176x144), Sub-QCIF (128x96), Output frame rate up to 15 fps.

Level 20—Bit rate up to 128 kbps, output resolutions: CIF (352x288), QCIF (176x144), Sub-QCIF (128x96), Output frame rate up to 15 fps for CIF, 30 fps for QCIF and Sub-QCIF.

Level 30—Bit rate up to 384 kbps, output resolutions: CIF (352x288), QCIF (176x144), Sub-QCIF (128x96), Output frame rate up to 30 fps.

Bit Rate

This setting indicates bits per second in units of kbps. For example, a value of 1500 is 1.5 Mbps. Bit rate defaults vary by level.

Key Frame Interval

This value (in frames) sets the maximum distance between key frames. The encoder may output key frames sooner than this interval (if it detects a scene change, for example).

Buffer Size

The buffer size value (in milliseconds) sets the encoder buffer size.

Output Resolution

Choose from the available resolutions or choose Custom, which will make the Cropping and Resizing settings available for modification. The pixel aspect ratio is auto set for each of these presets.

Cropping

The cropping parameters apply a crop to the input image. Note that if an odd number of lines are cropped from the top, the sense of which field is first (top or bottom) will change and you must set the Field Order setting on the Input Video tab accordingly.

Resizing

Specify the output resolution to be applied to the cropped image. If the resolution is different than the original, scaling will be performed. The minimum supported resolution is 64x64.

Resize Mode

Choose Progressive mode, Interlaced mode, or Single Field mode to manually specify how the resizer will process the source video. Single field mode, which should only be used for interlace source material, scales to the destination image using only a single field of the source video instead of performing a deinterlace operation. We highly recommend that you use single field mode for producing progressive frames from a 1080i source. For example, to stream a 1280x720p from a 1920x1080i source, we recommend that you choose the single field mode and resize to 1280x720 instead of performing a deinterlace operation.

Resize Algorithm

Choose Nearest, Linear, Cubic, or Super to choose the algorithm the resize operation uses. By default, Linear is used. Nearest (also referred to as Nearest neighbor) is the worst quality and has the lowest CPU requirements. Cubic is a higher quality scaler than Linear, and Super is a higher quality algorithm than Cubic. However, Super is very processor intensive and is only used for downscaling, and only when the resize is less than approximately half the source size. For example, if scaling single field mode from 1920x1080i, the field size is 1920x540, therefore, use Super if the output size is no more than 270 in height.

Output Frame Rate

Choose 1x to match the input frame rate, or reduce the input frame rate for telecine purposes or frame rate decimation. Available options listed are the input frame rate and 1/2, 1/3, 1/4, 1/5, and 1/6 the input rate. For example, if your input frame rate is 30 fps and you choose 1/2, then your output frame rate is 15 fps.

Pixel Aspect Ratio

To override the automatically set pixel aspect ratio, check the **Override** check box and modify the calculated ratio to the new ratio. If this check box is unchecked, the pixel aspect ratio is calculated based on the output size of the video for the preset sizes. Overriding is useful in cases where source video has a non-standard aspect ratio. For example, a 720x480 input source may have been produced anamorphically, and by indicating the aspect ratio here players can correctly resize video on playback to the non-anamorphic size.

Aspect Ratio Control (IP Input only)

When Dynamic is enabled on the Input Video tab, choose the method that is used to maintain the output aspect ratio when the input resolution or aspect ratio changes during encoding. Manual specifies not to automatically adjust for the aspect ratio. Letter/Pillar will add letterboxing or pillarboxing to achieve the specified output aspect ratio. Crop will modify the crop settings to achieve the specified output aspect ratio. The default is currently set to Crop.

Pre-Processing Area

▼ Pre-Processing

Interlacing Options:

None ▼

Deinterlace Mode:

<default> ▼

Miscellaneous Filters:

None ▼

Interlacing Options

If your video input is interlaced *and* your resolution height is greater than the field height, choose deinterlace to convert interlaced video to progressive. For resolutions less than or equal to the field height (320x240 for example for and NTSC source or 960x540 for HD 1080 sources), no deinterlacing is required and none should be specified.

Choose inverse telecine (IVT) to convert film-based interlaced 30 fps video to progressive 24 fps. If IVT is specified and the encoding mode is interlaced, the encoder will produce IVT flags (top field first, bottom field first, repeat first field) in the bitstream; decoders will then know how to display the 24 fps progressive video on an interlaced display.

Deinterlace Mode

If deinterlacing is chosen, the encoder provides multiple methods for deinterlacing. The default, a Motion Adaptive deinterlace, attempts to preserve spatial information in areas of motion while removing interlace artifacts in areas of motion. The Advanced option provides a higher quality deinterlace than the other options, but also uses significantly more CPU cycles. The Blend option blends two fields, maintaining temporal information through motion blur. The Interpolate mode removes temporal information and interpolates even fields from odd fields, while Interpolate Denoise applies a noise reduction filter after the deinterlace. The Line Double option simply creates even field lines as direct copies of the odd field lines. The CAVT option uses content adaptive vertical temporal filtering.

Miscellaneous Filters

The encoder provides for various noise reducing filters. The normal filter provides for moderate noise reduction. The smooth filter provides for large noise reduction at the expense of softer images.

Advanced Compression Settings Area

▼ Advanced Compression Settings

Number of B Frames:

None ▼

Bit Rate Mode:

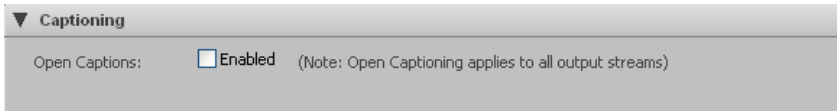
CBR ▼

Number of B Frames

No B frames may be used.

Bit Rate Mode

This setting may be set to CBR or VBR.

Captioning AreaA configuration panel titled 'Captioning' with a dropdown arrow. It contains a label 'Open Captions:' followed by a checkbox labeled 'Enabled' and a note in parentheses: '(Note: Open Captioning applies to all output streams)'.

▼ Captioning

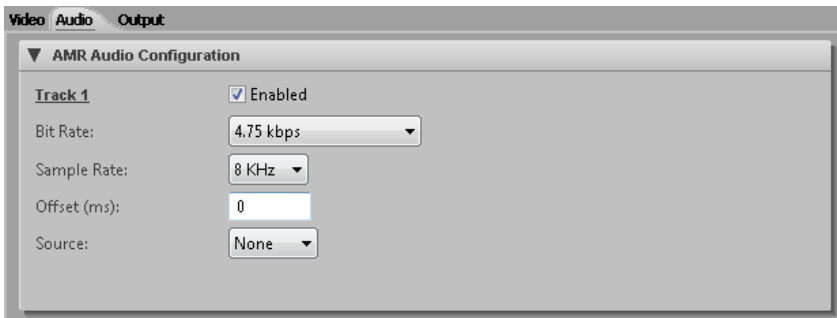
Open Captions: ☐ Enabled (Note: Open Captioning applies to all output streams)

Open Captions

When Open Captions is enabled, Line21 CC1 captions are decoded and rendered on top of the incoming video. Thus, unlike 608, 708 or SAMI captioning, the rendering of closed captions is done on top of video in the encoder instead of in the decoder.

Open captions requires NTSC 720x480 SD input sources.

Note: When open captions is enabled, it is enabled for all output streams, including any VC-1, VP6 Flash, H.265, or H.264 streams.

H.263 Audio TabA configuration panel titled 'H.263 Audio Tab' with tabs for 'Video', 'Audio', and 'Output'. The 'Audio' tab is selected. Inside, there's a section 'AMR Audio Configuration' with a dropdown arrow. It contains settings for 'Track 1': a checked 'Enabled' checkbox, a 'Bit Rate' dropdown set to '4.75 kbps', a 'Sample Rate' dropdown set to '8 KHz', an 'Offset (ms)' text box set to '0', and a 'Source' dropdown set to 'None'.

Video Audio Output

▼ AMR Audio Configuration

Track 1 ☒ Enabled

Bit Rate: 4.75 kbps ▼

Sample Rate: 8 KHz ▼

Offset (ms): 0

Source: None ▼

Check the **Enabled** check box to enable AMR encoding of a mono channel with the following parameters:

Bit Rate

Choose from the list of available audio bit rates in kbps.

Sample Rate

Choose the sample rate of input audio from the list of available rates in hertz.

Offset

Enter an offset (in milliseconds) between audio and video. The offset can be a positive or negative value. Typically, this value is 0 unless you know that your source audio and video are not in sync. If not in sync, use this setting to adjust timing such that they are re-synced prior to encoding.

Source

Choose from the available audio inputs to specify the audio source.

Input Channel (IP Input only)

For presets using IP input, choose from the available audio input channels to specify the audio stream to encode.

H.263 Output Tab

The screenshot shows a configuration window for H.263 output. At the top, there are tabs for 'Video', 'Audio', and 'RTP/3GPP Output', with 'RTP/3GPP Output' being the active tab. Below the tabs, there is a checkbox labeled 'Enable RTP/3GPP Output' which is checked. To the right of this checkbox is a 'Copy' button and the text 'Primary RTP/3GPP Stream Values'. Below this is a section titled 'Streaming to RTP/3GPP Origin Server' with a dropdown arrow. Inside this section, there are several configuration fields: 'RTP Mode' with a dropdown menu set to 'Multicast/Unicast RTP Streaming'; 'RTP Output Address' with a text field containing an asterisk; 'Network Transfer Mode' with a radio button selected for 'Unicast'; 'Preferred Adapter' with a dropdown menu set to 'Default selected by route'; 'Push Port' with two sub-fields, 'RTP Audio' and 'RTP Video', both set to '2'; 'Stream Name' with a text field; and 'MTU Size' with a spinner set to '0'. At the bottom of the window, there is a section titled 'Streaming to Redundant RTP/3GPP Origin Server' with a right-pointing arrow.

Cell phones have limited bandwidth and capabilities, and in addition wireless networks may place additional restrictions on network resources. We recommend verifying these capabilities in the network and setting video and audio encoding parameters on the H.263 Video and Audio tabs accordingly.

Recommendations for video streaming are as follows:

Encoding Parameter	Recommended Setting for RTP/3GPP Output
Profile	3
Level	30

CISCO LINEAR STREAM MANAGER

Bit Rate	64 to 128 kbps
Key Frame Interval	10 to 30 frames
Output Resolution	176x144 (QCIF)
Output Frame Rate	5 to 15 fps

Choose the stream for RTP/3GPP output and check the **Enable Streaming** check box to push content to a RTP/3GPP server. RTP/3GPP output cannot be enabled at the same time as other output formats.

Streaming to RTP/3GPP Origin Server Area

To use an optional second server, check the **Enable Streaming** check box in the Streaming to Redundant RTP/3GPP Origin Server area. Next, choose the RTP mode, which determines the other available settings.

When you are streaming via RTP unicast or multicast instead of sending to an RTSP server, the parameters that you define are slightly different, and in some cases, may have different meanings. Although both RTP and RTSP streaming tend to go to the same type of clients, and use the same codecs, they do so somewhat differently, thus we document these separately.

RTP Mode

Content can be pushed in one of two ways:

Using RTSP Signaling:

To use RTSP signaling to stream to a Darwin server, choose Streaming to an RTSP Server from the RTP Mode drop-down list. In this case, the SDP will be transferred in-band to the server and stored in the filename specified in the Stream Name field. The RTSP session will also negotiate RTP ports for media transfer. See *RTSP Settings* below for setting descriptions.

Using raw RTP traffic:

Choose Multicast/Unicast RTP Streaming from the RTP Mode drop-down list. Raw RTP packets will be transferred over the defined ports. See *RTP Settings* below for setting descriptions.

RTSP Settings

RTSP Server

Enter the name of the RTSP server. This can be an IP address of the server or any type of name that can be resolved to an IP address by AnyRes Live.

Preferred Adapter

Choose to have the default adapter selected by route, or choose the preferred adapter to be used for streaming.

Push Port

If using RTSP signaling, enter the RTSP port number. The default is 554.

Stream Name

The default name of the stream is displayed. If desired, enter a unique stream name. The associated SDP file will be created once the encode has started. By convention, stream names should always be a simple file name (no path components) and a .sdp extension, for example, *live_feed.sdp*. After streaming has started, you can choose Download SDP Files from the encoder context menu of the encoder monitor to download a copy of the SDP file. However, the SDP file has limited usefulness when streaming to an RTSP server.

Check the **Enable Authentication** check box to enable authentication if your RTSP server requires it. Enter the username and password and click **Apply** to add them to your configuration. The password characters will not be displayed.

MTU Size

Set the maximum packet size for RTP. The default is 1500.

RTP Settings

RTP Output Address

Enter the destination IP address. The IP address may be a unicast or a multicast address. For unicast addresses, use any type of name that can be resolved to an IP address by AnyRes Live.

Network Transfer Mode

Display only. This field identifies whether the encoder will use Multicast or Unicast mode to output the stream. If the first octet of the RTP IPV4 output address above is between 224 and 239, the mode is Multicast. All other addresses are Unicast mode.

Preferred Adapter

Choose to have the default adapter selected by route, or choose the preferred adapter to be used for streaming.

Push Port

If streaming raw RTP packets, enter the audio and video port numbers. These port numbers must be even, and the audio and video streams must be on different ports.

Stream Name

The default name of the stream is displayed. If desired, enter a unique stream name. By convention, stream names should always be a simple file name (no path components) and a .sdp extension, for example, *live_feed.sdp*. The associated SDP file will be created once the encode has started. After streaming has started, you can choose Download SDP Files from the encoder context menu of the encoder monitor to download a copy of the SDP file. Choose the location to download the SDP file, which can be used by client software to view the stream. If any encoding parameters are changed, a new SDP file must be created and downloaded to view the modified stream.

MTU Size

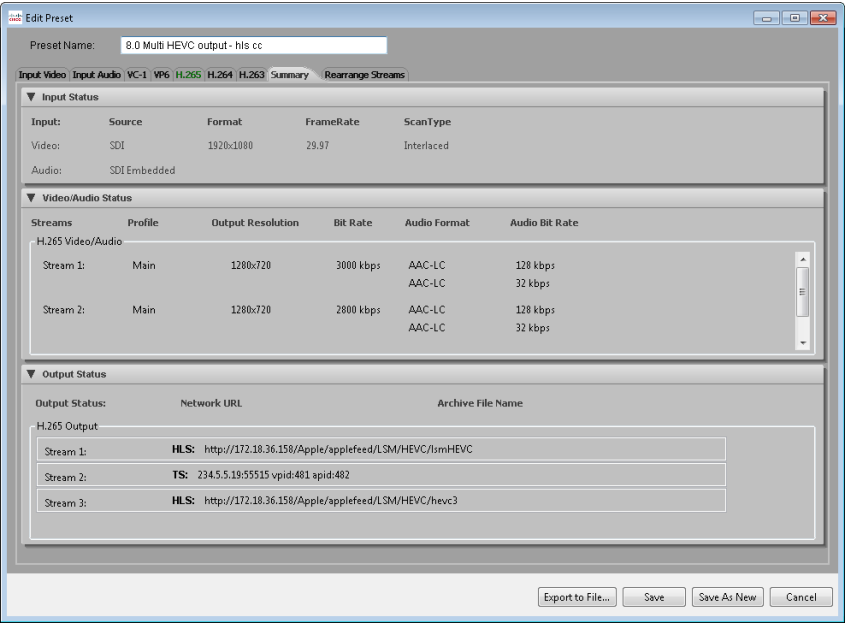
Set the maximum packet size for RTP. The default is 1500.

TTL

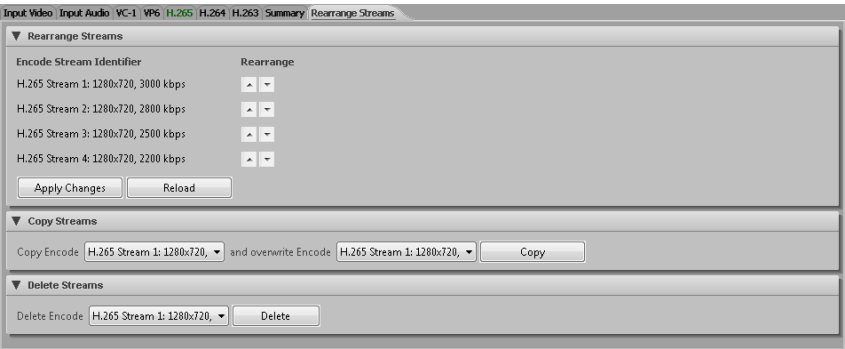
This setting is the time-to-live value for the multicast socket. For unicast streams, this setting is not available.

Summary Tab

Click the Summary tab of the preset editor to display the Summary screen, which includes the summary of Video, Audio and Output values of the encoded streams.



Rearrange Streams Tab



This tab allows you to rearrange and copy stream settings between available streams.

Rearrange Streams Area

This area lists all enabled streams within the preset in stream number order. Enabled streams will display the output format, stream number, resolution, and bit rate.

To rearrange streams, use the arrow buttons to move a stream to the desired position. Repeat the process for any other streams to be rearranged. When stream settings are in the desired order, click **Apply Changes**. After applying, the first stream will be Stream 1, and so forth.

Copy Streams Area

The Copy Encode drop-down list lists all enabled streams in the preset. Enabled streams display the output format, stream number, resolution, and bit rate of the stream.

To copy streams, choose the stream to be copied in the Copy Encode drop-down list. Next, choose the stream to overwrite in the second drop-down list. Finally, click **Copy** to copy the settings from the first stream to the second. The settings from the second stream will be lost.

Delete Streams Area

The Delete Encode drop-down list displays all enabled streams in the preset. Enabled streams display the output format, stream number, resolution, and bit rate of the stream.

To delete a stream, choose the stream to be deleted in the Delete Encode drop-down list and click **Delete**. The deleted stream will be disabled and moved to the end of the stream list. A deleted stream will only have its encoding disabled; all other settings will be left intact.

Saving a New Preset


After you have specified all settings for the encoded file, click **Save**.

Once the preset has been saved, it can be found in the Unassigned preset group until it is added to one or more other preset groups.


Exporting a Preset

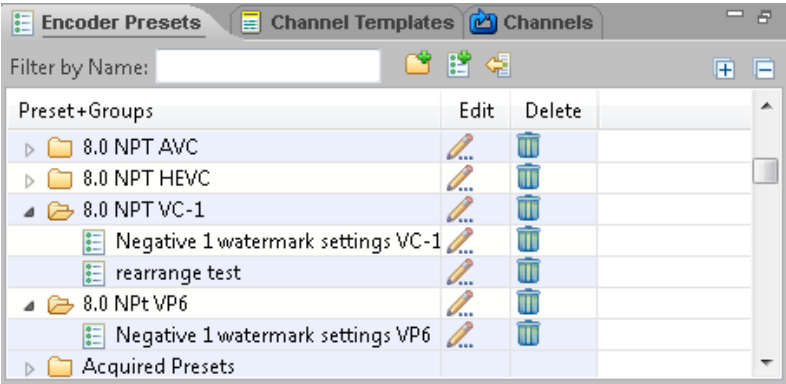
Export the preset to a file that can be imported into an AnyRes Live encoder through the encoder web interface. Click **Export to File** and enter the file name and location. The default file name is the preset name with a .settings extension.

Importing a Preset

From the preset editor, you can import a preset file from an AnyRes Live encoder by clicking **Import** and browsing to the file. You can also click the **Import an existing Preset**  icon on the Encoder Presets tab.

Editing an Existing Preset


To edit an existing preset, click the **Edit**  icon next to the preset.



The preset editor opens and loads the settings of that preset. Modify and save settings in the same way as a new preset, as detailed above. You can use the Preset Group drop-down list to move the preset to a different preset group. Changing the preset name will rename the existing preset, not create a new one. Standard or acquired presets cannot be saved to the original name.

To create a new preset from an existing one, change the preset name to the new name, use the drop-down list to choose the non-standard group where the preset will be placed, and click **Save as New**.

Deleting an Existing Preset

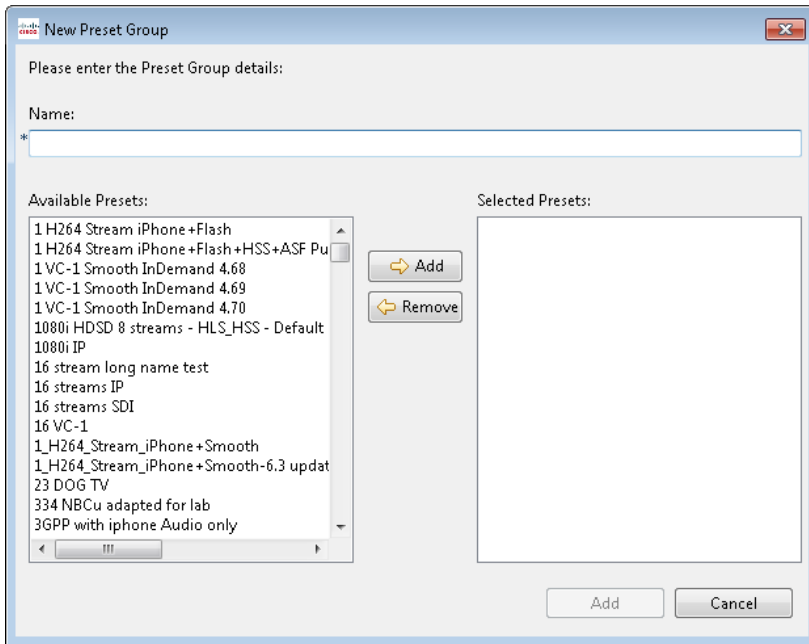
To delete an existing non-standard preset, first remove it from all preset groups as described in “Editing a Preset Group.” Once the preset is no longer a part of any preset groups, it is listed in the Unassigned group. To delete the preset permanently, click the **Delete**  icon next to the preset in the Unassigned group.

Preset Groups

The LSM provides default presets in preset groups. When a new preset is added, it is placed in the Unassigned preset group, which only lists presets that are not associated with any other preset group. Presets may be added to any combination of groups.


Creating a New Preset Group

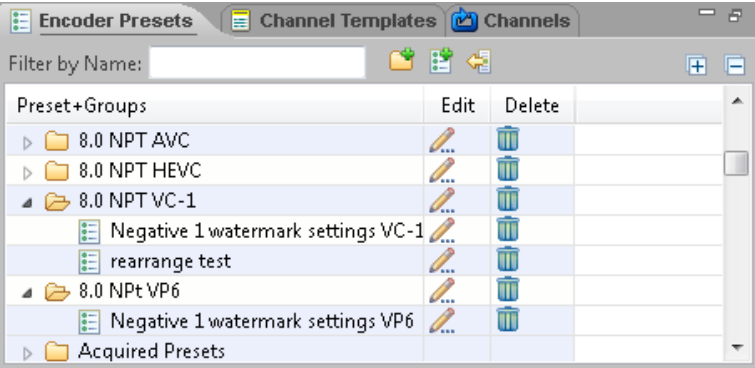
To create a new preset group, click the **New Preset Group**  icon to display the New Preset Group dialog box.



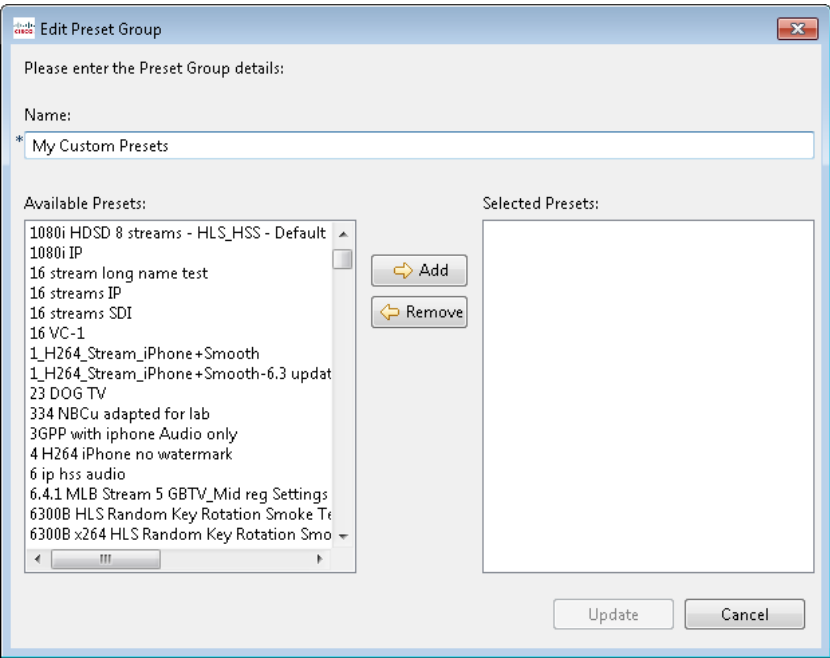
First, enter the preset group name in the **Name** text box. Next, highlight an available preset that will be part of the group, and then click **Add**. Continue to add presets until the group is complete. Finally, click **Add** to create the new preset group. A preset may be listed in multiple groups.

Editing a Preset Group

Standard preset groups shipped with the LSM cannot be edited. To edit an existing non-standard preset group, click the **Edit**  icon next to the preset group name.



The Edit Preset Group dialog box appears.




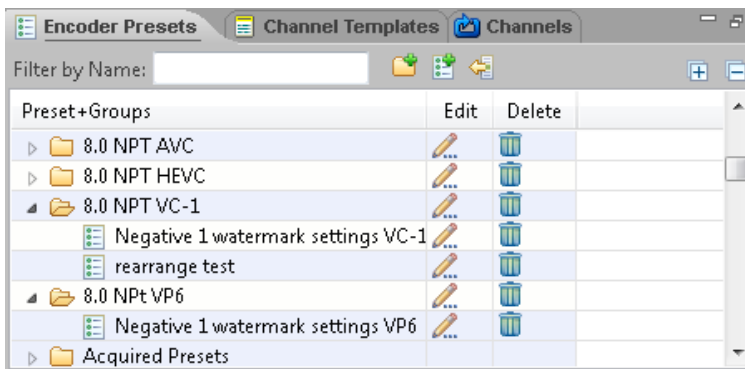
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You can modify the preset group name in the **Name** text box. To add or remove a preset from the group, select the preset name, and then click **Add** or **Remove** as appropriate. Finally, click **Update** to save the modified preset group.

If a preset is removed from all groups, it is placed in the Unassigned group. Presets cannot be deleted until they have been removed from all other groups, which places them in the Unassigned group.

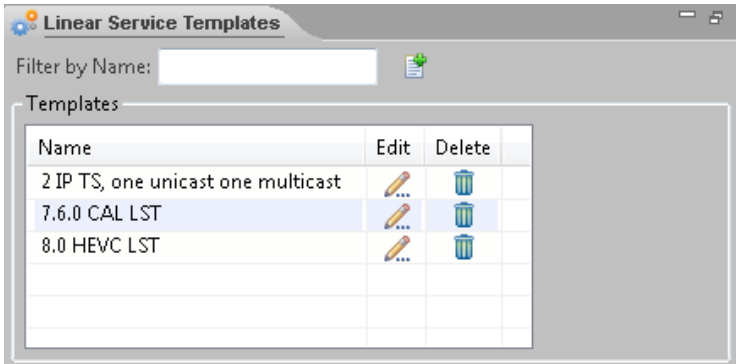
Deleting a Preset Group

To delete an existing non-standard preset group, click the **Delete**  icon next to the preset group name.



Click **Yes** to confirm that the preset group should be deleted. Any preset still in that group that is not associated with another group will be placed in the Unassigned group.

Linear Service Template Creation

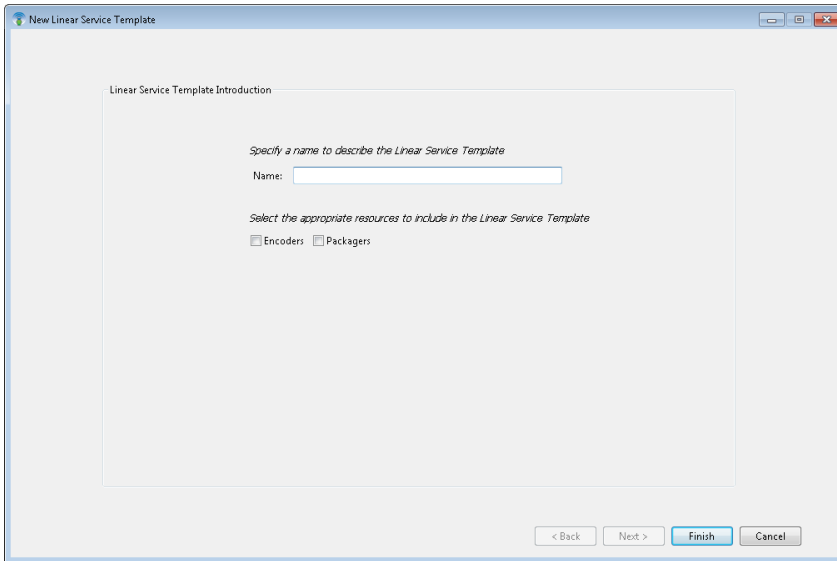


For encoders, a Linear Service Template (LST) specifies the essential makeup of a Live Streaming Event or Channel. The template includes the combination of encoders, whether individually allocated in a 1:N scenario where multiple primary and backup encoders can be provisioned, or allocated in an encoder pool (N:M scenario) that can be provisioned for redundancy purposes over the lifetime of some future Channel or live streaming event. The number of presets required must match the number of individual primary encoders chosen in the 1:N scenario, or must be less than or equal to the number of encoders within the group. Where applicable, the SDI input source configuration can be specified on a preset by preset basis. If N encoders are chosen and all N are associated with an SDI Router, then a default input source must be specified for each encoder/preset. In the encoder group scenario, the group must

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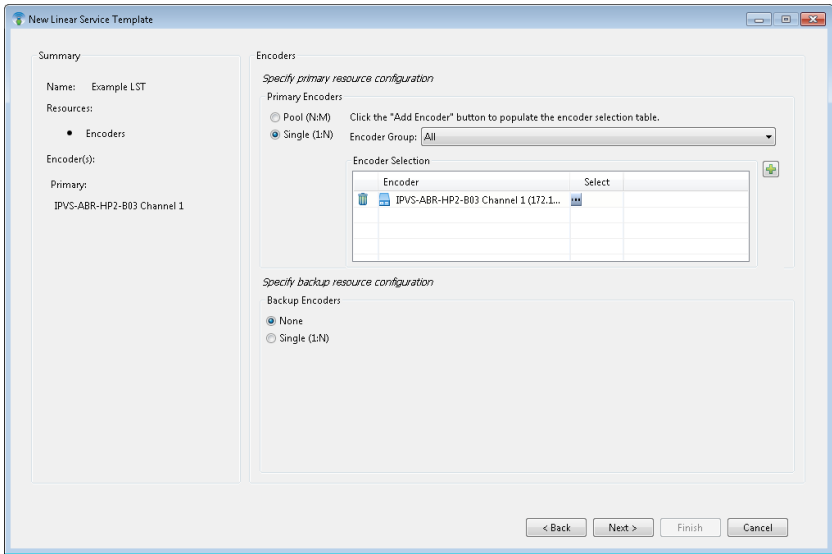
contain encoders attached to the same SDI router for the concept of SDI failover to work successfully.

To create a new LST, click the **New Linear Service Template**  icon to open the creation wizard.




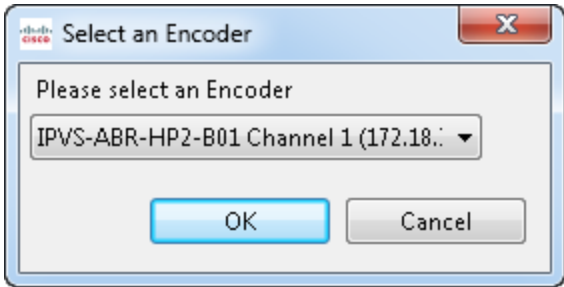
The image shows a screenshot of the 'New Linear Service Template' wizard window. The window has a title bar with the text 'New Linear Service Template' and standard window controls. The main content area is titled 'Linear Service Template Introduction'. It contains two sections: 'Specify a name to describe the Linear Service Template' with a 'Name:' label and a text input field, and 'Select the appropriate resources to include in the Linear Service Template' with two checkboxes labeled 'Encoders' and 'Packagers'. At the bottom right, there are four buttons: '< Back', 'Next >', 'Finish', and 'Cancel'.

The New Linear Service Template wizard has three screens. On the first screen, name the LST. If the LSM has both encoders and packagers, check the **Encoders** check box. Click **Next**.



On the second screen, first choose whether the primary resource configuration is a Pool (N:M) configuration or a Single (1:N) configuration.

If Single, optionally restrict the available encoders to a specific encoder group, then choose one or more encoders for the streaming event eventually spawned from this template. To add an encoder, click the **Add Encoder**  Icon, choose the encoder in the Select an Encoder dialog box, and then click **OK**.



Repeat adding encoders until all primary encoders are specified.

To add one or more backup encoders, click the **Single** radio button in the Backup Encoders area, and then add encoders in the same way as the primary area. A backup encoder is reserved for the same period of time that the primaries are reserved for. If a primary encoder fails and the server cannot talk to it anymore, the backup will be configured to start encoding. The server will take the preset and override information

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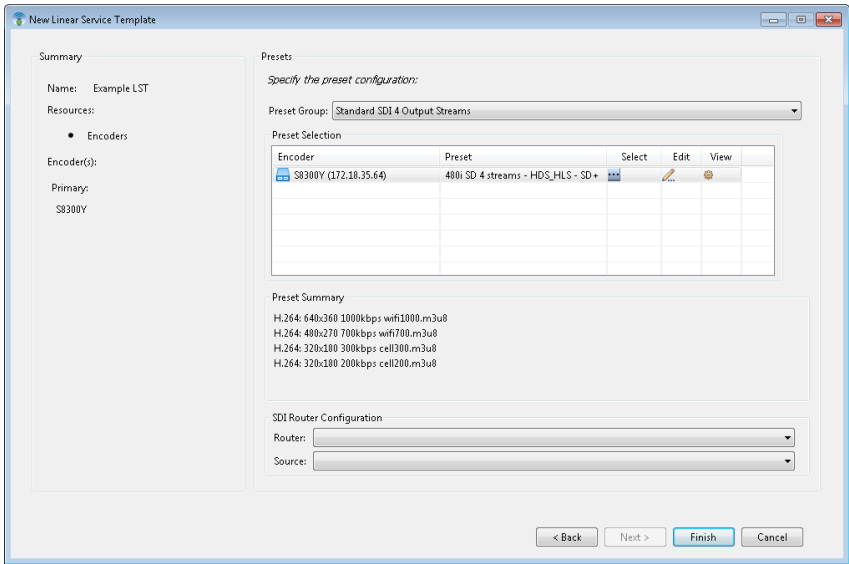
from the encoder that failed and send that information to the backup node and tell the backup to start encoding. If H.264 RTMP streams are enabled and the encoder experiences connection issues to the Flash Media server during a failover, the video outage time may be increased while the Flash streams try to reconnect. Currently, this switch only occurs if the primary fails. If the primary fails and there are no available backup encoders defined in the event, then that event will fail. You can have multiple primary and backup nodes defined in an event. If a primary fails, the server will just choose one of the backups to use.


For a Pool configuration, click the **Pool** radio button in the Primary Encoders area.

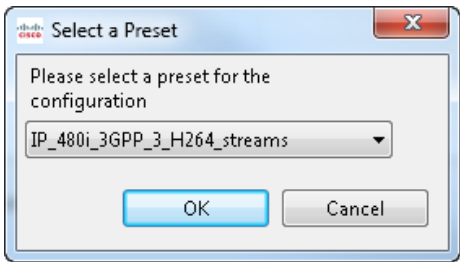
The screenshot shows the 'New Linear Service Template' window. The left pane has a 'Summary' section with the following details: Name: Example LST, Resources: Encoders, Encoder Group(s): Primary: N 8300. The right pane is titled 'Encoders' and contains two sections. The first section, 'Specify primary resource configuration', has a 'Primary Encoders' subsection where the 'Pool (N:M)' radio button is selected and the 'Encoder Group' dropdown is set to 'N 8300'. The second section, 'Specify backup resource configuration', has a 'Backup Encoders' subsection where the 'None' radio button is selected. At the bottom of the window are four buttons: '< Back', 'Next >', 'Finish', and 'Cancel'.

A pooled configuration uses an encoder group containing at least 1 encoder. Backup resources are implied when using a pooled approach because each encoder within the encoder group can serve as a backup to the node chosen to run the event.


Choose the encoder group for the pool, and then click **Next**.



In a Single (1:N) configuration, each of the valid primary nodes must be assigned a preset. Optionally, first use the Preset Group drop-down list to restrict the available presets to a specific preset group. Then, click the  icon in the Select column for each encoder to display the Select a Preset dialog box.



Choose the preset for the encoder, and then click **OK**. Repeat for any additional encoders.

In the case of a pooled (N:M) configuration, you can add any number of presets as long as enough encoders exist in the pool to handle the capacity of presets provisioned. To add presets, click the **Add Preset**  icon and choose the preset in the Select a Preset dialog box.

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Once presets are added, they can be edited. Also, the override settings for that preset, consisting of any IP input or output publishing parameters, can be viewed but not altered from this screen.

If any encoder is connected to an SDI Router, you must choose the input source configurations for the encoder by choosing the router and source from the drop-down lists in the SDI Router Configuration area.

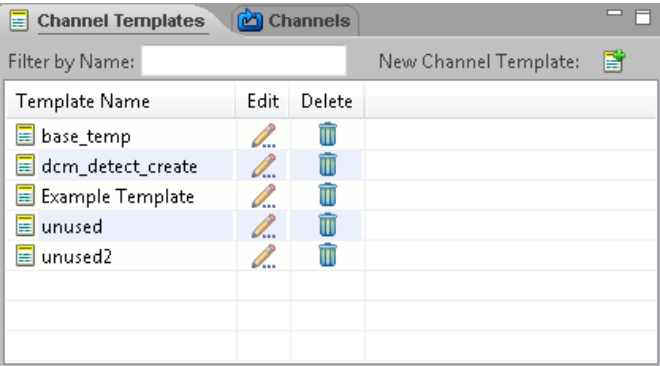
When the LST is complete, click **Finish**.

Channel templates are used to store settings for creating Channels for packagers. The Channel Templates tab is located on the Configure tab.

Channel Templates for Packagers

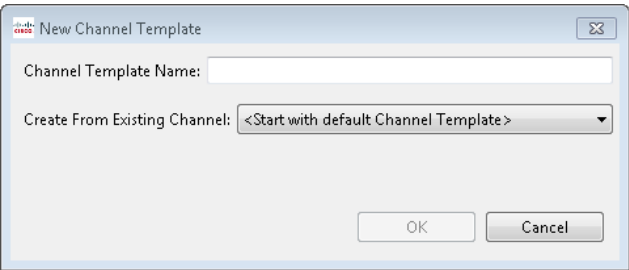
4

Enter text in the Filter by Name field to restrict the display of templates to those that begin with a particular character string.



Channel Template Editor

To create a new Channel template, click the **New Channel Template** icon to display the New Channel Template dialog box.



Enter a name for the Channel template and choose to either start with a default Channel template, or choose an existing Channel to create the new Channel, and then click **OK**. The Channel template editor will appear.


To edit an existing template, click the **Edit** icon next to the template name to display the Channel template editor.

This editor is divided into five areas of configuration, each with its own tab:

- **Input**—Provides input configuration fields.
- **Output**—Provides output configuration fields.
- **Mapping**—Provides mapping of inputs to outputs.
- **Presentations**—Provides configuration of master index files.
- **Global**—Provides ESAM parameters.

The Channel editor, available from within the Channel configuration, includes an additional Channel Redundancy tab that provides configuration for redundant instances of a Channel. Some other settings, such as PlayReady settings, are only available in the Channel editor and will not appear in the Channel template editor. For information about Channel creation, see “Channels for Packagers” on page 210.

The tab labels will change from black to green when that tab is enabled, except for the Channel Redundancy tab in the Channel editor if certain conditions are met. The Channel Redundancy tab is green if redundancy is enabled and all enabled outputs have redundant configurations. However, the Channel Redundancy tab is yellow if redundancy is enabled but an output which does not support redundancy is configured (DASH or DASH-TS). This tab is also yellow if redundancy is enabled and HDS is enabled, but the HDS servers table in the Channel Redundancy tab is blank.

Each area within the tabs can be collapsed or expanded by clicking the down arrow next to the area label. Helpful tips are provided by clicking or hovering your mouse cursor over the  icon in the top right corner of an area.

Editing and Saving Templates

After modifying settings in a Channel template, click **OK** to save. New templates must be saved with a name that is unique in the Cisco Linear Stream Manager (LSM). Existing templates cannot be modified and saved with the same name if they are already in use by one or more Channels. If a template has been used to create a Channel, that template is considered “in use” and cannot be modified or deleted. To save a modified Channel that is in use, you must change the name and save it as a new template.

General Template Settings

At the top of the editor, you can modify settings that apply to the template in general.

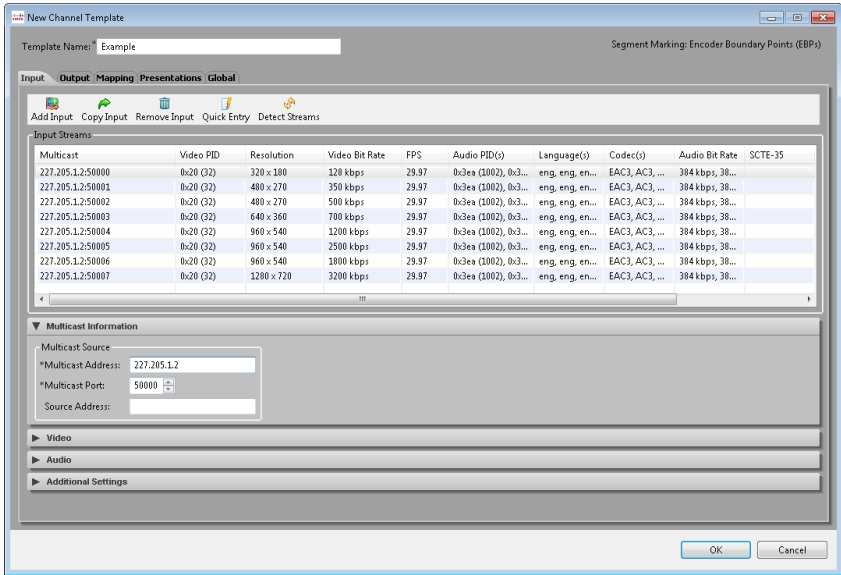
Template Name

Enter the name of the Channel template. This name must be unique in the LSM. If the template has been used to create a Channel, you must change the name to save the modified template.

Segment Marking

Display only. Encoder Boundary Points (EBPs) This mode uses the EBPs (explicit markings in the Transport Stream where ABR segmentation may occur) to determine boundary points for segmentation/fragmentation.

Input Tab



The Input tab defines input streams and allows viewing and modification of input stream settings. The task bar at the top of the tab provides the following tasks:

Add Input

Click **Add Input** to add a definition for another input stream.

Copy Input

In the table, select the line of the input that you want to copy, and then click **Copy Input** to add an input with the same settings as the selected input. Copy Input is not available when multiple streams are selected.

Remove Input

In the table, select the line of the input that you want to remove, and then click **Remove Input** to delete an input stream.

Quick Entry

See the Quick Entry Method description below for instructions on using Quick Entry to add input streams.

Detect Streams


Click **Detect Streams** and choose a packager group to use the multicast source to detect settings for selected streams. The LSM will populate the Video, Audio, and Additional Settings areas with detected stream settings for that port.

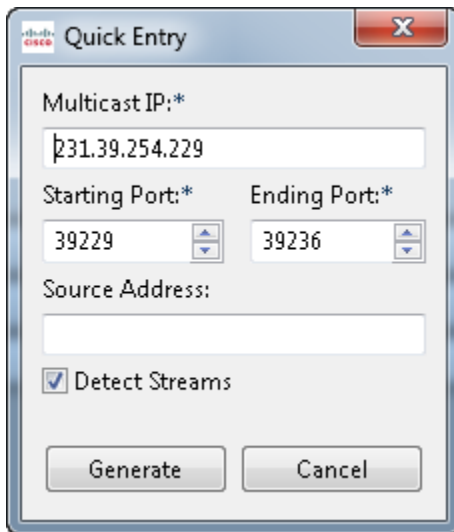
Supported stream types:

- 0x1b—H264 video
- 0x0f—AAC audio
- 0x11—AAC audio
- 0x81—Dolby Digital audio
- 0x87—Dolby Digital Plus audio
- 0x06—EN 300 472/EN 301 775 Teletext (descriptor 0x56 must be present)
- 0x86—DPI (SCTE35)

Two primary methods are available to define the settings for multiple multicast input streams:

Quick Entry Method

To define multiple input streams in one step, click the **Quick Entry**  icon in the task bar to display the Quick Entry dialog box.




The image shows a Windows-style dialog box titled "Quick Entry" with a Cisco logo in the top-left corner and a close button (X) in the top-right corner. The dialog contains the following fields and controls:

- Multicast IP:*** A text input field containing the value "31.39.254.229".
- Starting Port:*** A spin box containing the value "39229".
- Ending Port:*** A spin box containing the value "39236".
- Source Address:** An empty text input field.
- ☒ **Detect Streams**
- At the bottom, there are two buttons: **Generate** and **Cancel**.

Enter the Multicast IP, Starting Port, Ending Port, and optional Source Address for the streams that you are defining, up to 16 input streams. Optionally, check the **Detect Streams** check box to detect the stream settings for each port. Click **Generate**, choose a Packager Group from the drop-down list in the Select a Packager Group dialog box, and click **OK** to define an input stream for each port in the series. After the stream definitions are generated, you can modify the settings for each stream by clicking the stream entry, and then adjusting the settings for that stream in the Video, Audio, and Additional Settings areas.

Individual Stream Method

To specify each input stream individually, enter the Multicast Address and Multicast Port in the Multicast Information area. Optionally, also enter the Source Address, which is the IP address of the equipment that is generating the stream. Optionally click the **Detect Streams**  icon to populate the Video, Audio, and Metadata Stream areas with detected stream settings for that port. For subsequent streams, click **Add Input** to add a definition for a lower bit rate stream, change the port number in the Primary Multicast area, and repeat the preceding steps.

Note: If a template was used to set the initial stream settings and the next input stream is already defined, click the input stream in the list instead of clicking **Add Input** to define the next stream.

Video Area

▼ Video

Video

*PID:101

*Bit Rate:3000 kbps

*Resolution:1280 x 720

B Frames:☒ Enabled

Frame Rate:29.97

View or modify the following stream settings for each video input stream:

PID

Enter or verify the Video PID of the transport stream.

Resolution

Enter or verify the resolution of the transport stream. The minimum resolution is 64x64.

Frame Rate

Enter or verify the Video PID of the transport stream.

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Bit Rate

Enter or verify the bit rate of the transport stream in bits per second. The range is 96 kbps to 10 Mbps.

B Frames

Optionally check the **Enabled** check box to if the transport stream includes B frames.

Audio Area

The screenshot shows the 'Audio' configuration window. On the left, there is a table with columns: Audio PID, Language, Codec, Sample Rate, and Audio Bit Rate. The first row contains the values: 0x66 (102), eng, AAC, 48 Hz, and 96 kbps. To the right of the table is a section titled 'Audio' with several controls: an 'Enable Audio' checkbox (checked), 'Add Audio' and 'Remove Audio' buttons, and input fields for '*PID:' (102), '*Bit Rate:' (96 kbps), 'Codec:' (AAC), 'Language:' (eng), 'Sample Rate:' (48 KHz), and '*Channels:' (2).

Audio PID	Language	Codec	Sample Rate	Audio Bit Rate
0x66 (102)	eng	AAC	48 Hz	96 kbps

Audio configuration controls:

- ☒ Enable Audio
- Add Audio Remove Audio
- *PID: 102
- *Bit Rate: 96 kbps
- Codec: AAC
- Language: eng
- Sample Rate: 48 KHz
- *Channels: 2

Check the **Enable Audio** check box to enable audio for an input stream. At least one audio stream is required for HSS output. A transport stream may contain multiple audio streams. To view the settings for an audio stream, click the stream in the table to display or modify its settings. To manually add an audio stream, click **Add Audio** and modify the audio settings. To remove an audio stream, select the stream in the table, and then click **Remove Audio**.

View or modify the following stream settings for each audio input stream:

PID

Enter or verify the Audio PID of the transport stream.

Bit Rate

Enter or verify the bit rate of the audio in bits per second. The range is 6 kbps to 512 kbps.

Sample Rate

Display only. View the sample rate of the audio in hertz, based on the input audio.

Codec

Choose or verify the codec for the input stream from the drop-down list.

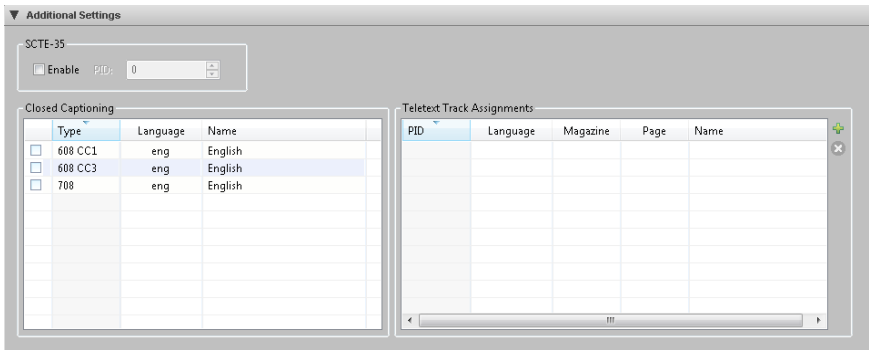
Language

Enter or verify the ISO language identifier for the input stream.

Channels

Enter or verify the number of audio channels in the input stream.

Additional Settings Area



The Additional Settings area is used to include SCTE-35, Closed captioning, or Teletext in the available input streams that can be mapped to output streams. If you have used Detect Streams, the detected inputs will be populated in this area. You should only enable these additional settings if they are present in the multicast input. Closed Captioning can be enabled if the source material contains closed captions embedded in the video, which is most common with NTSC-based content. European content (PAL) is more likely to make use of teletext tracks, which are carried on separate PIDs within the incoming transport stream.

SCTE-35

To enable SCTE-35 metadata for an input stream, check the **Enable** check box and enter or verify the metadata PID of the transport stream.

Closed Captioning

Check the check boxes for 608 CC1, 608 CC3, or 708 to process closed captioning on the input source and have CEA-608 or 708 data inserted into the video elementary stream user data area. Closed captioning can only be enabled on the first input stream of an encode channel.

Teletext Track Assignments

For teletext input, click the **Add** icon to include up to 8 teletext tracks. For each track, enter a PID, language, magazine, page, and name.

The language identifier is a unique 3-letter value. The range for the magazine is 0 to 7, and the range for the page in the specified magazine is 0 to 255. The default name is *track_language*, where language is the language identifier.

Including Closed Captions or Teletext in Streams

After choosing to include closed captions or teletext in the Additional Settings area of the Input tab, you can only include them in an output if the chosen output

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publisher supports that captioning type. Only HLS supports teletext output. The following table shows which publishers support CC 608 or 708.

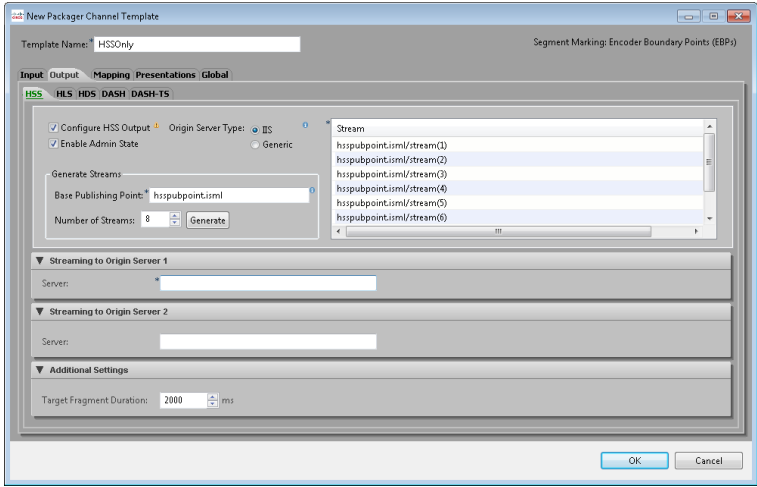
Publisher	CC 608	CC 708
HSS	Yes	Yes
HLS	Yes	No
HDS	No	Yes
DASH	No	No
DASH-TS	No	No

To associate enabled closed captions or teletext input with a given output publisher, the inputs must be mapped to output streams within the specific publisher's mapping area on the Mapping tab.

Publisher rules for mapping differ:

1. For HSS, enable closed captions only for the first stream in the mapping area.
2. For HLS, add additional streams in the output tab, one for each language that is available for CC or teletext. Then, proceed to map the captioning PID to the additional stream. WebVTT is available by default for HLS streams with CC or teletext enabled. HLS does not support both CC and teletext on a video stream.
3. For HDS, add additional streams in the output tab, one for each language that is available for CC. Then, proceed to map the closed captions PID to the additional stream.

Output Tab



The Output tab defines output streams and allows viewing and modification of output stream settings. On each tab, check the **Configure Output** check box to configure that output. Also, each tab provides an **Enable Admin State** check box. An output must have admin state enabled to be published. To configure the settings of an output without publishing it, uncheck the Enable Admin State check box.

Each tab also has a Generate Streams area. Enter the desired number of streams in the Number of Streams field and click **Generate** to populate the table of output streams. For HLS or HDS, to include alternative audio streams, the number of streams to generate should be the number of video streams plus the number of alternative audio streams. For HLS, to include SCTE-35, closed captions, teletext, or alternative audio streams, the number of streams to generate should be the number of video streams plus the number of other types of streams. For any output except HSS, you can click the populated stream values to modify them.

HSS Tab

To output Microsoft HTTP Smooth Streams (HSS), click the HSS tab and check the **Configure HSS Output** check box. Next, click an **Origin Server Type** radio button. If you change the origin server type, click **Generate** again to populate the table of output streams with the stream names for that type. The IIS option is the default, and will publish to a Microsoft IIS origin server. Click the Generic radio button to publish to a generic origin server, such as Apache. Choosing the Generic option will remove the Base Publishing Point field in the Generate Streams area, and will change available settings in the Streaming to Origin Server and Additional Settings areas.

Next, enter values for the following settings:

Base Publishing Point

For IIS origin servers, enter the base name of the publishing point on the origin server through which the encoded streams will be broadcast. This publishing point will be incorporated in the stream names in the table.

Number of Streams

Enter the number of streams to generate.

Click **Generate** to generate streams, and then view or modify the following settings:

Streaming to Origin Server Areas for IIS Origin Servers

Server

Enter the name of the origin server. This can be an alias, a fully qualified domain name, or an IP address of the server. Optionally, also enter the IP address of a redundant server in the Streaming to Origin Server 2 area.

Streaming to Origin Server Areas for Generic Origin Servers

▼ Streaming to Origin Server 1

Storage URL: *Address:

Command: Authentication:

Username: Password:

Publishing URL:

Storage URL

To specify the URL for where the packager will copy files, first choose the URL scheme from the drop-down list, and then enter the remainder of the URL address. If you choose an authentication option other than None, you must enter a username and password.

Using HTTP POST or PUT:

First choose http:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

example.com/example/

example.com:8080/example/

Choose POST in the Command drop-down list to specify that HTTP transfer will use a POST command to transfer a file, or choose PUT for the PUT command, which requires the HTTP handler to place the content in the specified location. The target of the command is the destination file name. The destination is responsible for creating the destination file, including any folders used as part of the destination

file name. To do this, the destination may require an HTTP handler. If using authentication, choose Basic or Digest authentication from the drop-down list, and enter the username and password.

Using file transfer:

First choose file:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

w2k8mediaServer/example/

Using WebDAV:

First choose webdav:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

example.com/webdav/

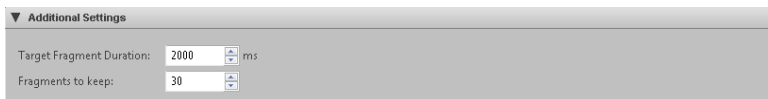
example.com:80/webdav/

If using authentication, choose Basic authentication from the drop-down list, and enter the username and password.

Publishing URL

Enter the URL that the customer uses to access files. If this is left blank, relative URLs are used.

Additional Settings Area



The screenshot shows a section titled "Additional Settings" with a downward arrow icon. Below the title, there are two settings: "Target Fragment Duration" with a value of 2000 and a unit of ms, and "Fragments to keep" with a value of 30. Both values are in input fields with up and down arrows.

Target Fragment Duration

Enter (in milliseconds) the target duration for HSS fragments. The default is 2000 ms.

Fragments To Keep

(Generic origin server only) To remove old fragment files from the origin server after a certain number of files have been generated, enter the number of files to keep. This creates a sliding window of fragments stored on the origin server. Once the limit is reached, for every new fragment, the oldest fragment will be deleted.

HSS PlayReady Area

PlayReady provides encryption support for HSS Output Fragments.

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In the Channel editor only, check the **Enable PlayReady** check box to use HSS PlayReady. To complete the rest of the PlayReady setup, you will need a third-party PlayReady License provider. This provider may supply you with setup information which you can manually enter when the Encryption Type is Manual Entry.

Some third-party PlayReady Platforms have been integrated to automatically retrieve the setup information. To use the third-party approach, you will also need a business relationship with the third party. If you choose an Encryption Type for a third-party provider such as BuyDRM, additional settings will be available at the time of Channel creation after you close the Channel editor.

If you choose Manual PlayReady, provide values for the following settings:

KID

Enter the Key ID. This value is required. The KID must be supplied as a GUID.

GUID should be entered in the form of: **{cab50618-c7e1-4e84-b683-c086d15170eb}**

Content Key

The encryption content key. **Either** a seed or a content key is required.

License Acquisition URL

Enter the URL. A License acquisition URL is required.

License Acquisition user interface URL

This is an optional field. If supplied, enter the URL.

Domain Service ID

This is an optional field. If supplied, enter a Domain Service ID. The ID may be supplied as either a GUID or Base64 string.

GUID should be entered in the form of: **{cab50618-c7e1-4e84-b683-c086d15170eb}**

An example Base64 string is: **GAa1yuHHhE62g8CG0Vfw6w==**

Custom XML

This is an optional field. The PlayReady license provider may specify custom data for the PlayReady header. If supplied, use this custom XML field to include it.

HLS Tab

Configure HLS Output

☒ Enable Admin State

Generate Streams

☐ Create Audio-Only Stream

☒ Independent Audio Streams

Number of Streams: **Generate** **Reset**

Stream	Bit Rate
StreamA	3000000
StreamB	1500000
StreamC	1250000
StreamD	1000000
StreamE	750000
StreamF	500000
StreamG	350000
StreamH	150000

Streaming to Origin Server 1

Storage URL:

Command:

Authentication:

Username: Password:

Publishing URL:

Streaming to Origin Server 2

Storage URL:

Command:

Authentication:

Username: Password:

Publishing URL:

Playlist File Publishing

Segmentation Settings

Other Settings

To output HLS Streams, click the HLS tab, check the **Configure HLS Output** check box, enter the number of streams and click **Generate** to generate streams. To include an audio-only stream, check the **Create Audio-Only Stream** check box before generating streams. The **Independent Audio Streams** check box, enabled by default, indicates that audio streams will not be multiplexed with their relative input video streams; thus, producing alternate audio streams instead. In order to revert to a default multiplexed audio/video stream configuration, click **Reset** to remove all output streams, uncheck the **Independent Audio Streams** check box, and click **Generate** to re-generate streams. To include SCTE-35, closed captions, teletext streams, or additional alternative audio streams, the number of streams to generate should be the number of video streams plus the number of other types of streams. For each output stream, view or modify the following settings:

Stream

Enter the stream name, which is used as the name of the playlist (.m3u8) file. Stream name must contain valid characters for a file/folder name, and we do not recommend that you use spaces. Each output stream must use a different stream name.

Bit Rate

View or modify the generated bit rate in bits per second.

Streaming to Origin Server Areas

Storage URL

To specify the URL to be used by the packager when sending HLS files (segments, playlists) to the origin server, first choose the URL scheme from the drop-down list, and then enter the remainder of the URL address. Based on the address, you may be required to specify additional settings. For all address types except file and WebDAV, you must enter a username and password.

Using HTTP POST or PUT:

First choose http:// or https:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

example.com/apple/

example.com:8080/apple/

Choose POST in the Command drop-down list to specify that HTTP transfer will use a POST command to transfer files to the origin server, or choose PUT for the PUT command, which requires the HTTP handler to place the content in the specified location. The target of the command is the destination file name. The destination is responsible for creating the destination file, including any folders used as part of the destination file name. To do this, the destination may require an HTTP handler. If using HTTPS, optionally choose Basic authentication from the drop-down list, and enter the username and password.

Using WebDAV:

First choose webdav:// or webdav-ssl:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

example.com/webdav/

example.com:80/webdav/

If using authentication, choose Basic or Digest authentication from the drop-down list, and enter the username and password.

Publishing URL

Enter the URL that the customer uses to access files. Publishing URL will be used to generate #EXTINF entries in the playlist file. To play back an individual stream, the iOS player should go to Publishing URL/streamname.m3u8. To play an adaptive stream, use Publishing URL/master playlist file name.m3u8.

If this is left blank, relative URLs are used.

Playlist File Publishing Area

▼ Playlist File Publishing

Ad Network URL:

☒ Enable (Only playlists will be sent to this URL)

Ad Network Storage URL:

Address:

http://

Command:

POST

Authentication:

None

Username:

Password:

Ad Network URL

Check the **Enable** box for Ad Network URL to specify a URL that will be used for playlists only. When publishing to the Ad Network URL, the playlist and master playlist files for the stream will be sent to this URL in addition to any primary or redundant URLs. See the Storage URL setting description for instructions.

Segmentation Settings Area

▼ Segmentation Settings

Target Duration:

10

sec

Duration Type in Playlist:

☒ Floating Point

☐ Integer

Segments in Playlist:

20

☐ Enable VCO Mode

Segments per Folder:

100

☒ Enable Sub-Folders

Segment Lag:

1

Segments to Keep:

22

☒ Remove old files

Dimmed settings are not configurable and display the default value for this release.

Target Duration

Enter the target duration of each segment, in seconds. We recommend a value of 10 seconds.

Duration Type in Playlist

Choose **Floating Point** to specify segment durations in the playlist with floating point durations. We recommend using Floating Point to ensure compatibility with players using HLS version 3 or higher. Choose **Integer** to specify that all segments have integer durations as rendered in the playlist. In this case, the playlist may include an occasional variation in a segment duration to account for rounding error accumulation.

Caution: Integer durations may cause output to violate the HLS specification.

Segments in Playlist

Enter the number of segments to be held in the playlist file. After the limit is reached, for every new segment added, the oldest segment is removed, creating a sliding window. We recommend a value of 10 entries.

Segments per Folder

To generate a new subfolder after a certain number of segment files have been generated, check the **Enable Sub-folders** check box, and then enter the number of segment files per folder. We recommend a value of 200. Unchecking the

Enable Sub-Folders check box will disable subfolder generation; thus, HLS segments and playlists will be published to a single output directory on the origin server.

Segment Lag

Enter the number of segment files to create before generating the associated playlist file. We recommend a value of 1. Setting this to 0 can cause playback issues, while setting this to a higher number increases the amount of time before a stream can be played, but only at the initial start of the stream.

Segments To Keep

To remove old segment files from the origin server after a certain number of files have been generated, check the **Remove Old Files** check box, and then enter the number of files to keep. This creates a sliding window of segments stored on the origin server. Once the limit is reached, for every new segment, the oldest segment will be deleted.

HLS Encryption Settings Area

In the Channel editor only, check the **Enable Encryption** check box to use encryption. If using Manual PlayReady, enter the encryption settings.

If not using Manual Entry, choose BuyDRM, CKM, Generic AES-128, Manual Adobe Access, VGC, Cisco Keystore (Verimatrix –VGC implementation), or Verimatrix for encryption. The encryption settings will be available at the time of Channel creation.

Other Settings Area

▼ Other Settings

Provide Absolute Time in Playlist:	Don't Include ▼
Include Additional Time Data:	Don't Include ▼
Allow Players to Cache Video Data:	Don't Include ▼
Indicate Live or VOD Content:	Disable ▼
Create I-Frame Playlists:	Don't Create ▼
Add Endlist on Stop:	Disable ▼
Cue Point Tag Insertion:	Don't Include ▼
Support AirPlay:	Disable ▼
Use Chunked Transfer for HTTP:	<input type="checkbox"/> Enable
Generate Unique Names:	<input checked="" type="checkbox"/> Enable
<input type="checkbox"/> Enable Authentication for	Akamai ▼
Akamai Token Name:	<input type="text"/>
Akamai Token Key:	<input type="text"/>
Akamai TTL:	0 <input type="button" value="▲"/> <input type="button" value="▼"/>

Settings in this area apply to all HLS Output streams.

Note: Dimmed settings are not configurable and display the default value for this release.

Provide absolute time in playlist

To include in the playlist file the time when a TS segment was generated, choose Use NTP time to insert EXT-X-PROGRAM-DATE-TIME tags in the playlist file. This tag marks each segment in the playlist with the NTP timestamp corresponding to the segment.

Include additional time data

This setting is not included in the playlist.

Allow players to cache video data

This setting is not included in the playlist.

Indicate live or VOD content

Choose Enable to include the EXT-X-PLAYLIST-TYPE tag in the playlist file. This tag indicates whether the playlist type is VOD or EVENT, based on the type of playlist being generated. Disable indicates that the tag will not be included.

Create I-frame Playlists

I-frame Playlists allow players to perform operations similar to fast-forward and re-wind. Players which support this feature require that a special Playlist and set of HLS Segments (which contain only I-frames) be generated. Choose Create to create I-frame Only playlists in addition to other playlists.

Add Endlist on Stop

To add an end-of-stream indicator to the playlist when the encode is stopped, choose For live and VOD manifest if the indicator should be added for both live and VOD. Choose For VOD manifest only if the indicator should only be added for VOD. Disable indicates that the indicator will not be included.

Cue Point Tag Insertion

Cue Point Tag Insertion is used to support advertisements interleaved with HLS content. CUE-OUT markers are included in Playlists to indicate the HLS Segment after which an advertisement might begin; CUE-IN markers are included to indicate the HLS Segment after which an advertisement might end. These will only appear if the content arriving at the packager contains markers (for example, SCTE-35) which indicate these same boundary points.

To include cue point tags in the playlist, choose Include IN and OUT tags if CUE-IN and CUE-OUT markers should be included. Choose Include IN/OUT/CONT tags for Freewheel if CUE-IN, CUE-OUT, and CUE-OUT-CONT tags should be included.

Support AirPlay

Choose **Enable** to include RESOLUTION and CODEC tags in the playlist. **Disable** indicates that these tags will not be added. Some older HLS players do not play back streams with these tags present in the playlist.

Use Chunked Transfer for HTTP

Contact your CDN for the correct setting. The default is False.

Generate Unique Names

Check the **True** check box to generate unique names for segment files and folders. The file and folder names will include the starting date and time for the encode. Generating unique names prevents name collision and accidental removal of old streams when starting new streams, and also prevents unintentional referencing of old cached files when encoding a new stream. The default is True.

Authentication

Check the **Enable Authentication** check box and choose the CDN (content distribution network) from the drop-down list to enable authentication with an origin server. Each file is authenticated as it is sent to the CDN. Authentication is only valid when using HTTP to send files. Custom authentication parameters will display. For Akamai, the token name and key will be provided by Akamai, and TTL is the time-to-live of the encrypted token. A TTL value of 30 seconds is reasonable.

HDS Tab

The screenshot shows the 'HDS' tab selected in the Cisco Linear Stream Manager. The interface includes the following sections:

- Configuration:**
 - ☒ Configure HDS Output
 - ☒ Enable Admin State
- Generate Streams:**
 - Number of Streams: 8
 - Generate** button
 - Stream list: Stream, StreamA, StreamB, StreamC, StreamD
- Streaming to Origin Server:**
 - Origin Server(s): [Empty field]
 - Add...**, **Edit...**, and **Remove** buttons
- Additional Settings:**
 - Publish Segment Files: ☐ Enable
 - Target Fragment Duration: 4 sec
 - Fragments per Segment: 1
 - Files to Keep: 450
 - Flat Manifest File: ☐ Enable
 - Publish Keyframe-Only Video Stream: ☐ Enable

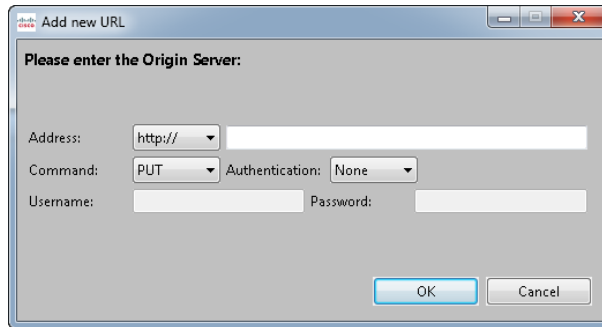
To output HDS Streams, click the HDS tab, check the Configure HDS Output check box, enter the number of streams and click **Generate** to generate streams. For multi-audio, to include alternative audio streams, the number of streams to generate should

be the number of video streams plus the number of alternative audio streams, up to 16 total streams and up to 15 alternative audio streams. For each output stream, view or modify the following settings:

Streaming to Origin Server Area

Origin Server(s)

Up to two origin servers can be configured. To add the information to identify an origin server, click **Add** to display the Add New URL dialog box.



First, choose the URL scheme from the drop-down list, and then enter the remainder of the URL address. For Authentication, if you choose the None option, the Username and Password fields are dimmed.

Using HTTP POST or PUT:

First choose http:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

example.com/hds/

example.com:8080/hds/

Choose POST in the Command drop-down list to specify that HTTP transfer will use a POST command to transfer a file, or choose PUT for the PUT command, which requires the HTTP handler to place the content in the specified location. The target of the command is the destination file name. The destination is responsible for creating the destination file, including any folders used as part of the destination file name. To do this, the destination may require an HTTP handler.

Using file transfer:

First, choose file:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

w2k8mediaServer/hds/

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If using file transfer, the AnyRes Live Encoding Service must have permission to create directories and copy files to the destination (for instructions, see “Setting Up AnyRes Live to Write to a Network Drive” on page 265).

After specifying the settings of the origin server, click **Add** to add it to the list. To edit an origin server after it has been added, select the server address, and then click **Edit** to display the Update Origin Server dialog box. In the dialog box, modify settings, and then click **Update**. To remove an origin server from the list, select the server address, and then click **Remove**.

Using WebDAV:

First, choose `webdav://` from the drop-down list, and then provide the remainder of the address in the second address field. For example:

`example.com/webdav/`

`example.com:80/webdav/`

If using WebDAV, optionally choose Basic authentication from the drop-down list, and enter the username and password.

Additional Settings Area

Publish Segment Files

Check the **Enable** check box to publish segment files instead of only fragment files.

Target Fragment Duration

Enter (in seconds) the target duration for HDS fragments. This setting must be a multiple of the key frame interval. A good size fragment is small enough to send over the Internet, and long enough to prevent multiple calls to the server. We recommend a value of 4 seconds, but the value may be altered depending on the GOP size. The range is 2 to 60 seconds.

Fragments per Segment

If Publish Segment Files is enabled, enter the number of fragments per segment. A value of 1 indicates that the entire segment consists of one fragment. The range is 1 to 30, and the default is 1.

Files To Keep

To remove old segment files after a certain number have been generated, enter the number of segment files to keep. This creates a sliding window of files stored on the server. Once the limit is reached, for every new pair of segment files, the oldest will be deleted. A segment consists of two files, one `.f4f` and one `.f4x`. A value of 0 indicates that no segment files will be deleted from the server. The range is 0 to 1000, and the default is 450.

Flat Manifest File

Check the **Enable** check box to specify a flat manifest file that includes bootstrap information in addition to stream information.

Publish Keyframe-Only Video Stream

Check the **Enable** check box to publish a keyframe-only video stream. This stream is referenced from the content manifest file and allows an HDS player to fast forward or rewind the live content.

HDS Encryption Settings Area

In the Channel editor only, check the **Enable Encryption** check box to use encryption.

Choose CKM or Manual Adobe Access for encryption. The encryption settings will be available at the time of Channel creation.

DASH Tab

The screenshot shows the 'DASH' configuration tab within a software interface. At the top, there are tabs for 'HDS', 'HLS', 'DASH', and 'DASH-TS', with 'DASH' being the active tab. Below the tabs, there are two main sections. The first section contains checkboxes for 'Configure DASH ISO-BMFF Output' and 'Enable Admin State', both of which are checked. Below these is a 'Generate Streams' section with a 'Number of Streams' input set to '8' and a 'Generate' button. To the right of this is a table with a 'Stream' header and four rows labeled '/stream(1)', '/stream(2)', '/stream(3)', and '/stream(4)'. The second section is titled 'Streaming to Origin Server' and contains several input fields: 'Base Stream Name' (set to 'basename'), 'MPD File Name' (set to 'mpdname'), 'Command' (set to 'POST'), 'Authentication' (set to 'None'), 'MPD Storage URL' (with 'Address' set to 'http://' and 'mpdstorage'), 'Initialization Storage URL' (with 'Address' set to 'http://' and 'initstorage'), and 'Media Storage URL(s)' (set to 'http://mediastorage'). There are also 'Username' and 'Password' fields for each URL. At the bottom of this section are 'Add...', 'Edit...', and 'Remove' buttons. The final section is 'Additional Settings' with 'Initialization Publishing URL' (set to 'http://') and 'Media Publishing URL(s)' (empty), also with 'Add...', 'Edit...', and 'Remove' buttons.

To output DASH Streams, click the DASH tab, check the Configure DASH ISO-BMFF Output check box, enter the number of streams and click **Generate** to generate streams, and view or modify the following settings:

Streaming to Origin Server Area

Base Stream Name

Enter the base stream name. If you generate streams after entering this value, this name will be incorporated in the stream names in the table.

MPD File Name

Enter the MPD (Media Presentation Description) file name. This file name does not include a path.

MPD Storage URL

To specify the URL to copy MPD files, first choose the URL scheme from the drop-down list, and then enter the remainder of the URL address. If Authentication is Basic, you must enter a username and password.

Using HTTP POST or PUT:

First choose http:// or https:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

example.com/apple/

example.com:8080/apple/

Choose POST in the Command drop-down list to specify that HTTP transfer will use a POST command to transfer a file, or choose PUT for the PUT command, which requires the HTTP handler to place the content in the specified location. The target of the command is the destination file name. The destination is responsible for creating the destination file, including any folders used as part of the destination file name. To do this, the destination may require an HTTP handler.

Using WebDAV:

First choose webdav:// or webdav-ssl:// from the drop-down list, and then provide the remainder of the address in the second address field. For example:

example.com/webdav/

example.com:80/webdav/

If using authentication, choose Basic or Digest authentication from the drop-down list, and enter the username and password.

Initialization Storage URL

To specify the URL to copy initialization files, first choose the URL scheme from the drop-down list, and then enter the remainder of the URL address. If Authentication is Basic, you must enter a username and password.

Media Storage URL

To specify one or more URLs to copy media files, first click **Add** to display the Add new URL dialog box. Choose the URL scheme from the drop-down list, and then enter the remainder of the URL address. If Authentication is Basic, you must enter a username and password. Next, click **OK**. Repeat the procedure to add additional URLs, up to a maximum of 4 URLs. To edit or delete a previously entered URL, select the URL, and then click **Edit** or **Remove**.

Initialization Publishing URL

Enter the URL that the end client uses to access files. If this is left blank, a relative URL is used.

Media Publishing URL

To specify one or more URLs where media files may be retrieved by the end client, first click **Add** to display the Add new URL dialog box. Enter the URL that the customer uses to access files. Next, click **OK**. Currently, the number of Media Publishing URLs is limited to 1. To edit or delete a previously entered URL, select the URL, and then click **Edit** or **Remove**. If this is left blank, relative URLs are used.

Additional Settings Area

▼ Additional Settings

Publish Media Segments:

☒ Enable

Segment Duration:

2

sec

Remove Old Segments:

☒ Enable

Time Shift Window:

3600

sec

MPD Update Period:

60

sec

Use Segment Timelines:

☒ Enable

Initialization Segment Template:

\$RepresentationID\$-init.mp4

Insert Macro...

Fragment Segment Template:

\$RepresentationID\$_\$Number\$.mp4

Insert Macro...

Availability Start Time:

11/ 8/2012

2:30:57 PM

EST

Publish Media Segments

Check the **Enable** check box to publish all data, including the MPD, initialization segments, and media segments. Disable to publish only MPD and initialization segments. The default is Enable.

Segment Duration

Enter the segment duration in seconds. The range is 0 to 100. The default is 2 seconds.

Remove Old Segments

Check the **Enable** check box to remove old segments from the origin server. The default is Enable.

Time Shift Window

When Remove Old Segments is disabled, this value is used to ensure that the generated DASH MPD contains references to the correct number of segments. When Remove Old Segments is enabled, this value is used to determine the interval after which old segments should be removed from the origin server. The range is 0 to 200000 seconds. The default is 3600 seconds.

MPD Update Period

Enter the frequency at which the MPD is updated on the origin server, in seconds. The range is 1 to 86400 seconds. The default is 2 seconds.

Use Segment Timeline

Check the **Enable** check box to include explicit information on each media segment in the DASH MPD. The default is Enable.

Initialization Segment Template

Enter the string that is used to determine the filenames to which the initialization segments for each stream are stored on the origin server. These filenames will be referenced in the DASH MPD. Click **Insert Macro** to display a dialog box of segment template macro options. The default is \$RepresentationID\$-init.mp4.

Fragment Segment Template

Enter the string that is used in the DASH MPD to determine the filenames to which the media segments for each stream are stored on the origin server. Click **Insert Macro** to display a dialog box of segment template macro options. The default is \$RepresentationID\$_\$Time\$.mp4.

Availability Start Time

Enter the time at which the stream is expected to be available on the origin server.

DASH-TS Tab

HSS HLS HDS DASH DASH-TS

☒ Configure DASH-TS Output
☒ Enable Admin State

Generate Streams
 Number of Streams: 8

Stream	Bit Rate
stream(1)	3096000
stream(2)	1596000
stream(3)	1346000
stream(4)	1000000

▼ Streaming to Origin Server

Origin Servers:

Server Address	Base Publishing Point

Port: 80

▼ Additional Settings

I-Frame Indexing: ☒ Enable

The DASH-TS Publisher indexes the streams for later re-encapsulation using On-Demand Encapsulation (ODE). The indexed information is used by ODE to transform the streams from DASH-TS into client-specific ABR target formats (HLS, HSS). The origin server must be an Apache server.

To output DASH-TS Streams, click the DASH-TS tab, check the Configure DASH-TS Output check box, enter the number of streams and click **Generate** to generate streams, and view or modify the following settings:

Streaming to Origin Server Area

Origin Servers Address

To specify one or more origin servers, first click **Add** to display the Add New Server dialog box. For the Server Address, enter the name of the server. This can be an alias, a fully qualified domain name, or an IP address of the server. For the Base Publishing Point, enter the base name of the publishing point on the server through which the encoded streams will be broadcast. Next, click **OK**. Repeat the procedure to add additional origin servers. To edit or delete a previously entered server, select the server address, and then click **Edit** or **Remove**.

Port

Enter the TCP port number on the Origin Server.

Additional Settings Area

I-Frame Indexing

Check the **Enable** check box to enable extended indexing of the I-frames present in the stream. Extended indexing is required for ODE (On-Demand Encapsulation) to perform trick mode playback, such as fast forward and rewind. The default is Enable.

Mapping Tab

Multicast	Video PID	Resolution	Video Bit Rate	FPS	Audio PID(s)	Language(s)	Codec(s)	Audio Bit Rate	SCTE-35
227.205.1.2:50000	0x20 (32)	320 x 180	120 kbps	29.97	0x3ea (1002), 0x3...	eng, eng, en...	EAC3, AC3, ...	384 kbps, 38...	
227.205.1.2:50001	0x20 (32)	480 x 270	350 kbps	29.97	0x3ea (1002), 0x3...	eng, eng, en...	EAC3, AC3, ...	384 kbps, 38...	
227.205.1.2:50002	0x20 (32)	480 x 270	500 kbps	29.97	0x3ea (1002), 0x3...	eng, eng, en...	EAC3, AC3, ...	384 kbps, 38...	
227.205.1.2:50003	0x20 (32)	640 x 360	700 kbps	29.97	0x3ea (1002), 0x3...	eng, eng, en...	EAC3, AC3, ...	384 kbps, 38...	
227.205.1.2:50004	0x20 (32)	960 x 540	1200 kbps	29.97	0x3ea (1002), 0x3...	eng, eng, en...	EAC3, AC3, ...	384 kbps, 38...	
227.205.1.2:50005	0x20 (32)	960 x 540	2500 kbps	29.97	0x3ea (1002), 0x3...	eng, eng, en...	EAC3, AC3, ...	384 kbps, 38...	
227.205.1.2:50006	0x20 (32)	960 x 540	1800 kbps	29.97	0x3ea (1002), 0x3...	eng, eng, en...	EAC3, AC3, ...	384 kbps, 38...	
227.205.1.2:50007	0x20 (32)	1280 x 720	3200 kbps	29.97	0x3ea (1002), 0x3...	eng, eng, en...	EAC3, AC3, ...	384 kbps, 38...	

Output Stream	Video PID	Bit Rate	Resolution	Audio PID	Name	Language(s)	Codec	Role	SCTE-35	CC (lang)
StreamA	0x20 (32)	120 kbps	320 x 180	0x3ea (1002)	English	eng	EAC3	Muxed		
StreamB	0x20 (32)	350 kbps	480 x 270	0x3ea (1002)	English	eng	EAC3	Muxed		
StreamC	0x20 (32)	500 kbps	480 x 270	0x3eb (1003)	English	eng	AC3	Muxed		
StreamD	0x20 (32)	700 kbps	640 x 360	0x3eb (1003)	English	eng	AC3	Muxed		
StreamE	0x20 (32)	1200 kbps	960 x 540	0x3ec (1004)	English	eng	AAC	Muxed		
StreamF	0x20 (32)	2500 kbps	960 x 540	0x3ec (1004)	English	eng	AAC	Muxed		
StreamG	0x20 (32)	1800 kbps	960 x 540	0x3ed (1005)	English	eng	HE_AAC_V1	Muxed		
StreamH	0x20 (32)	3200 kbps	1280 x 720	0x3ee (1006)	English	eng	HE_AAC_V2	Muxed		
StreamI				0x3ea (1002)	English	eng	EAC3	Audio-Only		

The Mapping tab maps input streams to output streams and allows viewing, but not modification, of their stream settings. The Input Streams area displays all available input streams. Output areas will only appear if that output has been configured.

If you are mapping Closed Captions or teletext inputs to output streams, see “Including Closed Captions or Teletext in Streams” on page 164 for information about the mapping guidelines for each publisher.

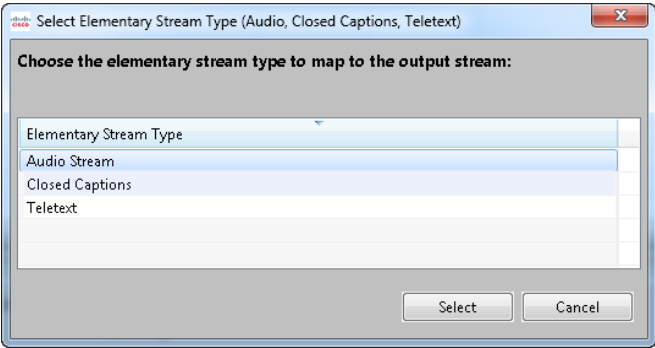
Automatic Mapping

Initially, for the number of defined output streams, the same number of input streams will be automatically mapped to those output streams, beginning with the first input stream. For example, if ten input streams are defined but only eight output streams, the first eight input streams will be mapped to the output streams.

The Input Streams area displays all input streams that have been generated for this Channel. These input streams will be automatically mapped to the same number of output streams, unless a different number of output streams were defined. To manually map one of these input streams to one of the output streams, drag the input line to the desired output stream. You can also select multiple input rows and drag the group of inputs to the output streams. The LSM will automatically populate the stream settings.

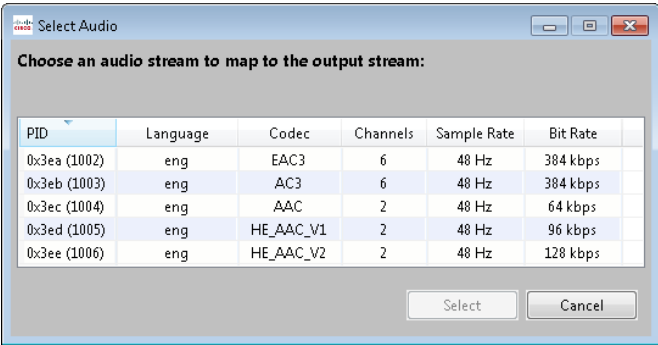
Note: Input streams cannot be mapped to more than one output stream of the same type.

If multiple audio streams were present on an input stream, the first audio PID will be automatically mapped to the corresponding output stream and will be muxed with the output video. To manually map multiple audio, closed captions, or teletext streams from the same input stream, drag the input stream to additional outputs. Because input video streams can only be mapped to one output of the same type, the Channel template editor will interpret the drag to a second output as a mapping of an audio, closed caption, or teletext elementary stream. If there are multiple types of elementary streams, the editor will display the Select Elementary Stream Type dialog box.



Choose the type of elementary stream to map.

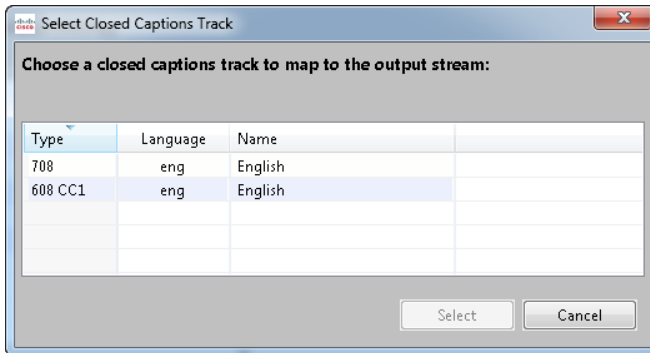
If mapping an audio stream, and more than one alternate audio is included in the input, the editor will then display the Select Audio dialog box.



Choose the audio to be mapped to this output stream, and then click **Select**. The output stream will display the settings for the mapped audio.

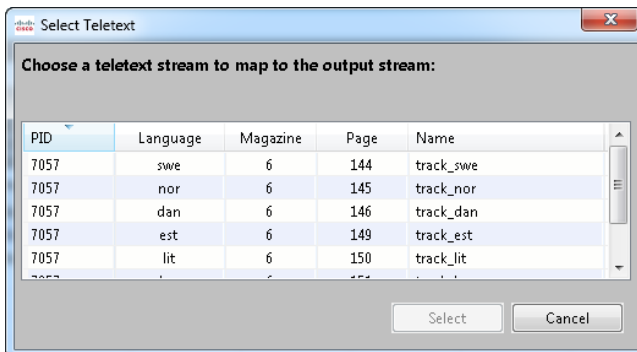
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If mapping closed captions, and more than one alternate captioning type is included in the input, the editor will then display the Select Closed Captions Track dialog box.



Choose the closed captions track to be mapped to this output stream, and then click **Select**. The output stream will display the settings for the mapped captions.

If mapping teletext, and more than one alternate teletext stream is included in the input, the editor will then display the Select Teletext dialog box.



Choose the teletext stream to be mapped to this output stream, and then click **Select**. The output stream will display the settings for the mapped teletext stream.

Output Format Areas

The editor displays an area for each configured output format. Each output format area displays base Channel settings above the table of output streams.

To remove the settings for an input stream that has been mapped to a particular output stream, right-click the entry for that output stream to bring up the following context menu.

HLS

Base Channel

Origin Server 1: Storage: http://hlsstorage

Publishing: hlspublishing

Origin Server 2: Storage:

Publishing:

Output Stream	Video PID	Bit Rate	Resolution	Audio PID	Audio	Name	Language
StreamA	0x65 (101)	3000 kbps	1280 x 720	0x66 (102)	<input checked="" type="checkbox"/>	English	eng
StreamB	<div><div></div>Remove Mapping</div>	1500 kbps	960 x 540	0xca (202)	<input checked="" type="checkbox"/>	English	eng
StreamC	0x12d (301)	1250 kbps	864 x 486	0x12e (302)	<input checked="" type="checkbox"/>	English	eng
StreamD	0x191 (401)	1000 kbps	640 x 360				
StreamE	0x1f5 (501)	750 kbps	640 x 360				

Choose **Remove Mapping** to remove the mapped input stream from this output stream. The output stream itself is not removed. The mapping tab will always display the number of output streams that were defined in the Output tab.

You can also select one or more rows in the table of output streams and delete the mapped input streams before dragging new input streams to map to outputs.

HSS

For HSS output, you must enable between one and three audio tracks. To enable an audio track, check the **Enabled** check box for that stream in the Track column. To change the track ID that uniquely identifies the stream, click the generated ID and enter the modified ID. All enabled audio tracks must have the same bit rate and a different track ID. To change which stream has audio enabled, first uncheck the Enabled check box for the enabled audio, and then check the desired Enabled check box. Before finishing, make sure that all audio output streams also have video mapped to that stream. For HSS or HDS output, if SCTE-35 metadata is enabled on the input stream, check the **Metadata** check box to include metadata in the output of one stream.

HLS and HDS

For HLS and HDS outputs, if alternative audio or audio-only streams have been generated, these streams will be displayed after the video streams. To change the name of the audio of an output stream, click the audio name in the Name column and enter the modified name.

Channel Redundancy Tab

The screenshot shows the 'Channel Redundancy' tab in the Cisco Linear Stream Manager. At the top, there are tabs for 'Input', 'Output', 'Mapping', 'Channel Redundancy' (which is highlighted in green), and 'Presentations'. Below the tabs, there is a checkbox labeled 'Enable Redundant Channel' which is checked. Under the 'Input' section, there is a 'Multicast source' section with a text field for '*Multicast Address' containing '231.39.254.229' and an empty 'Source Address' field. Below this is a 'Time Synchronized' section. Under the 'HLS' section, there are two 'Origin Server' configurations. 'Origin Server 1' has a 'Storage URL' section with a dropdown for 'Address' set to 'http://', a text field for 'hls:storage', a dropdown for 'Command' set to 'POST', a dropdown for 'Authentication' set to 'None', and text fields for 'Username' and 'Password'. Below this is a 'Publishing URL' field containing 'hls:publishing'. 'Origin Server 2' has a similar structure but with empty fields for 'Address', 'Command', 'Authentication', 'Username', and 'Password', and an empty 'Publishing URL' field.

Redundancy and Resiliency

The Channel Redundancy tab is only visible in the Channel editor, available during Channel creation or editing. For information about Channels, see “Channels for Packagers” on page 210.

In the LSM, redundancy refers to the configuration of multiple inputs or outputs that run concurrently and can therefore continue to provide uninterrupted downstream service if any single resource fails. Resiliency refers to the ability of the LSM Server to identify problems with a Channel that warrant relocation of the Channel to an alternate packager, with a minor interruption of service for that Channel.

In this release, redundancy is only available for HSS, HLS, and HDS output formats. If redundancy is configured for some outputs but not others, the tab color is yellow. If redundancy is configured for all enabled outputs, the tab color is green.

The Channel Redundancy tab allows configuration of input and output redundant resources. Check the **Enable Redundant Channel** check box to enable configuration of these resources.

Input Area

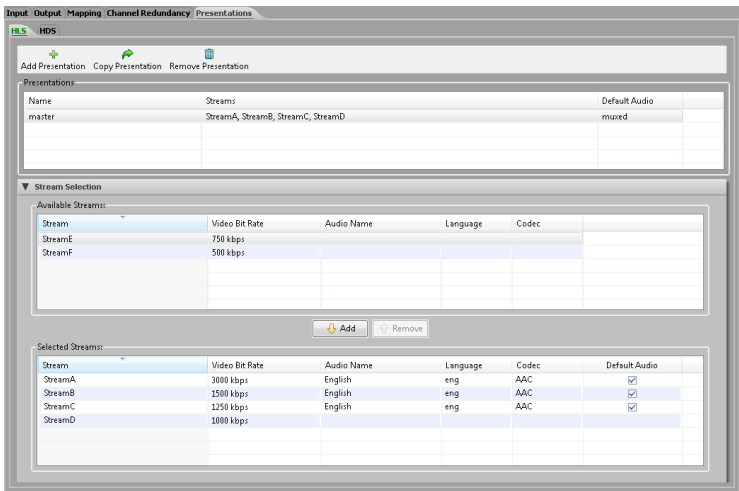
In the Input area, you can enter a redundant Multicast address and optional Source address for the redundant Channel.

HSS, HLS, and HDS Output Format Areas

For HSS, HLS, and HDS outputs, you can configure redundant outputs that will be time synchronized to the primary outputs. Configuring redundant outputs will result in a second Channel that has identical settings except for settings from the Redundancy tab. Redundant Channels should run on different packagers in a group so that if one packager has a problem or failure, the redundant Channel can be used.

For setting descriptions, see the Output tab for the output format.

Presentations Tab



The Presentations tab defines presentations, which are groups of HLS or HDS streams that also specify the default audio for the group. A presentation is the information that is used by the packager to generate HLS master playlist files or HDS master manifest files.

If a previously configured output includes a presentation, click the corresponding tab to view or modify which streams are included in a given presentation. One default presentation named “master” is automatically created when streams are generated. When created, the master presentation includes all configured streams. Individual streams may be added to multiple presentations.

The task bar at the top of the tab provides the following tasks:

Add Presentation

Click **Add Presentation** to add another presentation, which is created with a default name and no streams selected.

Copy Presentation

In the Presentations table, select the line of the presentation that you want to copy, and then click **Copy Presentation** to add a presentation with the same settings as the selected presentation.

Remove Presentation

In the Presentations table, select the line of the presentation that you want to remove, and then click **Remove Presentation** to delete it.

Presentations Table

Click the Name field of a presentation and enter a new name to rename the presentation. To modify the streams included in the presentation, use the Stream Selection area. The Default Audio field is display only, and indicates whether the default audio for the presentation is the audio muxed with a video stream, a designated audio stream, or 'none' if no stream with audio has been selected.

Stream Selection Area

To modify the stream selections for a presentation, select the row of the presentation in the Presentations table. To add an available stream, select the stream from the table and click **Add** to move the stream to the selected streams table. To remove a selected audio stream, make sure that it is selected in the table, and then click **Remove** to move it to the available streams table. Also, streams can be moved individually or in groups by selecting the streams to move, and then dragging them to the other list.

For Audio-only streams, optionally check the Default Audio check box if the stream should be the default audio for the presentation instead of muxed audio.

Master Playlist or Master Manifest Files

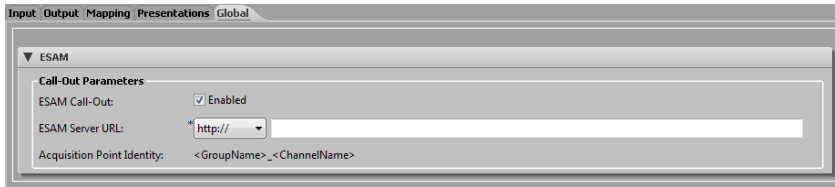
For HLS streams, multiple presentations will create multiple master playlists. A master index file points to the individual HLS streams that are available when a playback device accesses that master playlist.m3u8 file. Having multiple presentations will create separate master playlists, such as one master playlist for Wi-Fi connections and a separate one for cellular connections. The iOS player should be pointed to one of these master playlist files, and based on the detected bandwidth, it will play the appropriate stream. Over time, as bandwidth conditions change, the player will automatically be switched up to higher bandwidth streams or down to lower bandwidth streams to always deliver the best possible stream without buffering.

For HDS streams, multiple presentations will create multiple master manifest files in the same way. Enter the presentation name, which is the base name that is used for naming index and segmented files on the server. The master manifest file name must

contain valid characters for a file/folder name, and we do not recommend that you use spaces.

The default audio specified in the presentation will be marked as the default language in the master playlist or manifest files.

Global Tab



The Global tab provides settings to configure Event Signaling and Management (ESAM). ESAM is used to prepare the outgoing ABR package (HLS, HSS, or HDS) for Alternate Content Substitution (ACS). ACS allows downstream entities such as Manifest Manipulators to provide alternate content, such as targeted or zone-based advertising, blackouts, and the possibility of replacing national content with regionalized content.

ESAM Area

To configure ESAM, check the **Enabled** check box for ESAM call-out and enter the ESAM Server URL.

ESAM Server URL

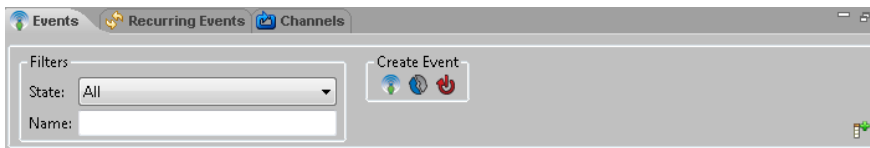
Enter the URL for the endpoint that the packager uses to communicate with the Position Opportunity Information Server (POIS). The packager will send ESAM requests (HTTP POSTs) to this URL.

Acquisition Point Identity


Display only. This field is the identifier string which identifies the entity that acquires the stream and the Channel being acquired, in the format *PackagerGroupName_ChannelName*.

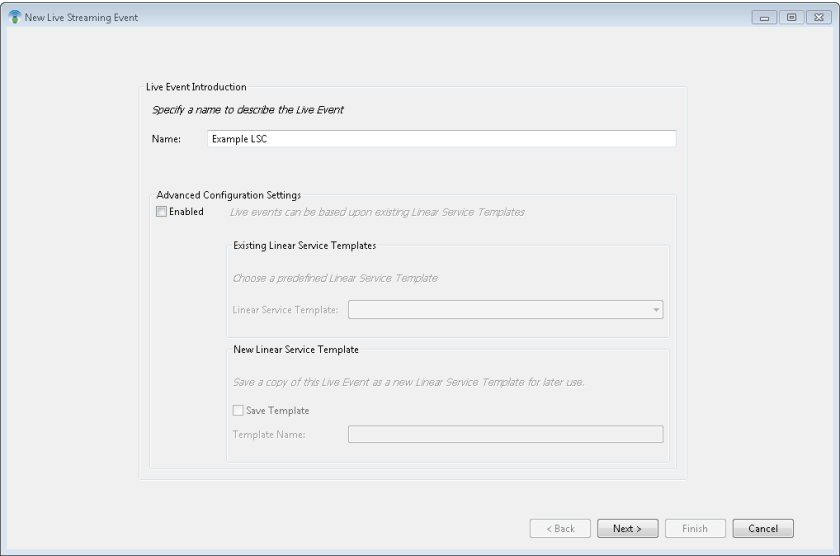
Event Creation and Channels for 5 Encoders

Three types of scheduled events can be created: Streaming, Update, and Reboot events. Streaming events can also be configured as recurring events, which creates multiple streaming events according to the schedule. For encoders, Channels spawn immediate streaming events that continue until stopped.



New Streaming Event

Click the **New Streaming Event**  icon on the Events tab, and the following New Live Streaming Event wizard appears.



The image shows a screenshot of the 'New Live Streaming Event' wizard window. The window has a title bar with the text 'New Live Streaming Event' and standard window controls. The main content area is divided into two sections. The first section, 'Live Event Introduction', contains the instruction 'Specify a name to describe the Live Event' and a text input field labeled 'Name:' with the value 'Example LSC'. The second section, 'Advanced Configuration Settings', contains a checkbox labeled 'Enabled' which is checked. Below this is a note: 'Live events can be based upon existing Linear Service Templates'. There are two sub-sections: 'Existing Linear Service Templates' with the instruction 'Choose a predefined Linear Service Template' and a dropdown menu labeled 'Linear Service Template:'; and 'New Linear Service Template' with the instruction 'Save a copy of this Live Event as a new Linear Service Template for later use.' and a checkbox labeled 'Save Template' which is unchecked. Below this is a text input field labeled 'Template Name:'. At the bottom of the window are four buttons: '< Back', 'Next >', 'Finish', and 'Cancel'.

The New Live Streaming Event wizard has a variable number of screens based on how you choose to specify the configuration of the event.

Live Streaming Event Introduction Screen

Name the event, and then click **Next**. Optionally, check the **Enabled** check box in the Advanced Configuration Settings area and choose to use an existing or new Linear

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Service Template (LST) as the template for the event. For more information on LSTs, see “Linear Service Template Creation” on page 152.

If you choose to use an LST, you will skip the following screen and proceed directly to the Presets screen of the wizard.

Live Streaming Event Encoders Screen

Summary

Name: Example LSC

Resources:

- Encoders

Encoder(s):

Primary:

IPVS-ABR-HP2-B03 Channel 1

Encoders

Specify primary resource configuration

Primary Encoders

☐ Pool (N:M) Click the "Add Encoder" button to populate the encoder selection table.

☒ Single (1:N)

Encoder Group: All

Encoder Selection

Encoder	Select
IPVS-ABR-HP2-B03 Channel 1 (172.1...)	

Specify backup resource configuration


Backup Encoders

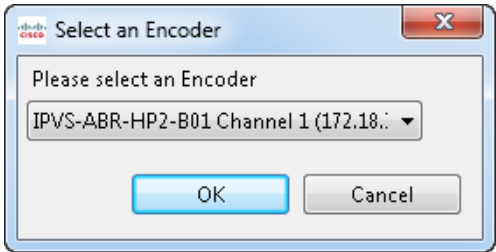
☒ None

☐ Single (1:N)

< Back Next > Finish Cancel

On the Encoders screen, first choose whether the primary resource configuration is a Single (1:N) configuration or a Pool (N:M) configuration.

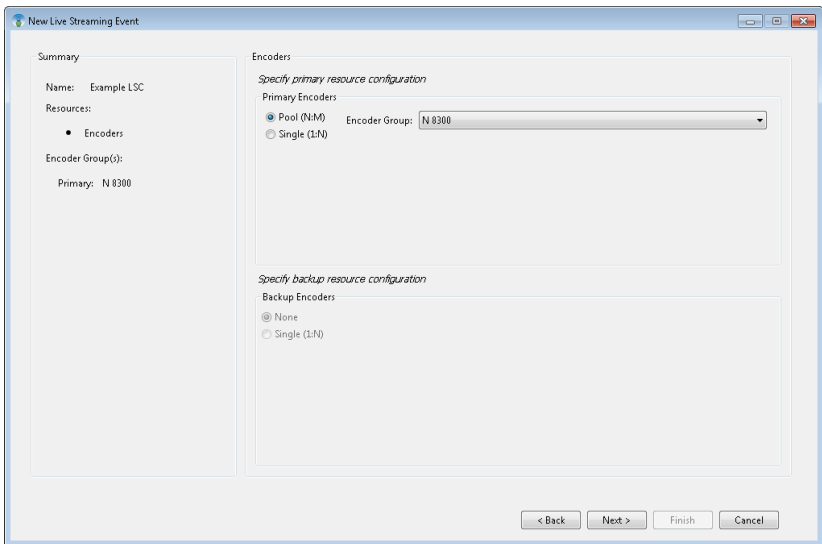
If Single, optionally restrict the available encoders to a specific encoder group, then choose one or more encoders for the streaming event. To add the encoder, click the **Add Encoder**  icon to display the Select an Encoder dialog box. Choose the encoder from the drop-down list in the dialog box, and then click **OK**.



Continue adding encoders until all primary encoders are specified.

To add one or more backup encoders, click the **Single** radio button in the Backup Encoders area, and then add encoders in the same way as the primary area. A backup encoder is reserved for the same period of time that the primaries are reserved for. If a primary encoder fails and the server can no longer communicate with it, then the backup will be configured to start encoding. The server will take the preset and override information from the encoder that failed, send that information to the backup node, and tell the backup to start encoding. Currently, this switch only occurs if the primary fails. If the primary fails and there are no available backup encoders defined in the event, then that event will fail. You can have multiple primary and backup nodes defined in an event. If a primary fails, the server will choose one of the backups to use.

For a Pool configuration, click the **Pool** radio button in the Primary Encoders area.



A pooled configuration is based upon an encoder group containing at least one encoder. Backup resources are implied when using a pooled approach, as each

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encoder within the encoder group can serve as a backup to the node chosen to run the event.

Choose the encoder group for the pool, and then click **Next**.

Live Streaming Event Presets Screen

The screenshot shows the 'New Live Streaming Event' window. It is divided into two main sections: 'Summary' on the left and 'Presets' on the right.

Summary Section:

- Name: Example Event
- Resources:
 - Encoders
- Encoder Group(s):
 - Primary: N 8300
- Preset(s):
 - TS 7.6.0 SDI
 - 7.6.0 RTMP settings test

Presets Section:

Specify the preset configuration:

Preset Group: 7.6.0 NPT H.264

Preset Selection Table:

Preset	Select	Edit	Over...
TS 7.6.0 SDI
7.6.0 RTMP settings test


Preset Summary:

H.264: 1280x720 3500kbps
H.264: 1280x720 3000kbps

SDI Router Configuration:

Router:
Source:

Buttons: < Back, Next >, Finish, Cancel

In a Single (1:N) configuration, each of the valid primary nodes must be assigned a preset. Optionally, use the Preset Group drop-down list to restrict the available presets to a specific preset group. Then, click the  icon in the Select column for each encoder to display the Select a Preset dialog box.


The screenshot shows the 'Select a Preset' dialog box. It has a title bar with the Cisco logo and a close button (X).

Please select a preset for the configuration


IP_480i_3GPP_3_H264_streams

Buttons: OK, Cancel

Choose the preset for the encoder, and then click **OK**. Repeat for any additional encoders.

In the case of a pooled (N:M) configuration, you can add any number of presets as long as enough encoders exist in the pool to handle the capacity of presets provisioned. To add presets, click the **Add Preset**  icon and choose the preset in the Select a Preset dialog box. Once presets are added, they can be edited.

Overrides

Once you choose a preset, the override settings for that preset, such as IP input stream settings or output publishing parameters, can be edited. Overrides are specific to the stream type. To display the Overrides dialog box, click the **Edit Overrides**  icon in the Overrides column of an encoder entry in the Preset Selection table. The Overrides dialog box provides the opportunity to change some output values for encoding from those in the preset. The preset values are filled in for reference. Choose the stream to override, change any settings for that stream, and then proceed to any other streams to override. For H.264, when multiple outputs are included in the stream, you can choose one of the outputs to override by choosing that output type from the drop-down list and clicking **Confirm**. When finished with overrides, click **OK**. If a setting is dimmed, it was not enabled in the preset and therefore is not available to be overridden.

If the preset is for IP source on an IP input AnyRes Live, the Overrides dialog box will include tabs for Input and Output overrides.

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Input/Output Overrides

Input Overrides | **Output Overrides**

Video Input

Primary source

Multicast Address: 239.255.001.076

Multicast Port: 5500

Source address(optional):

Secondary source

Multicast address: 0.0.0.0

Multicast port: 0

Source address(optional): 0.0.0.0

Selected Input Streams

Video

Video PID: 481

Format: 1920x1080

Frame Rate: 29.97

Scan Type: Interlaced

Field Order: Top Field First

▶ Audio Channel Assignments

▶ Digital Program Insertion

▶ Metadata

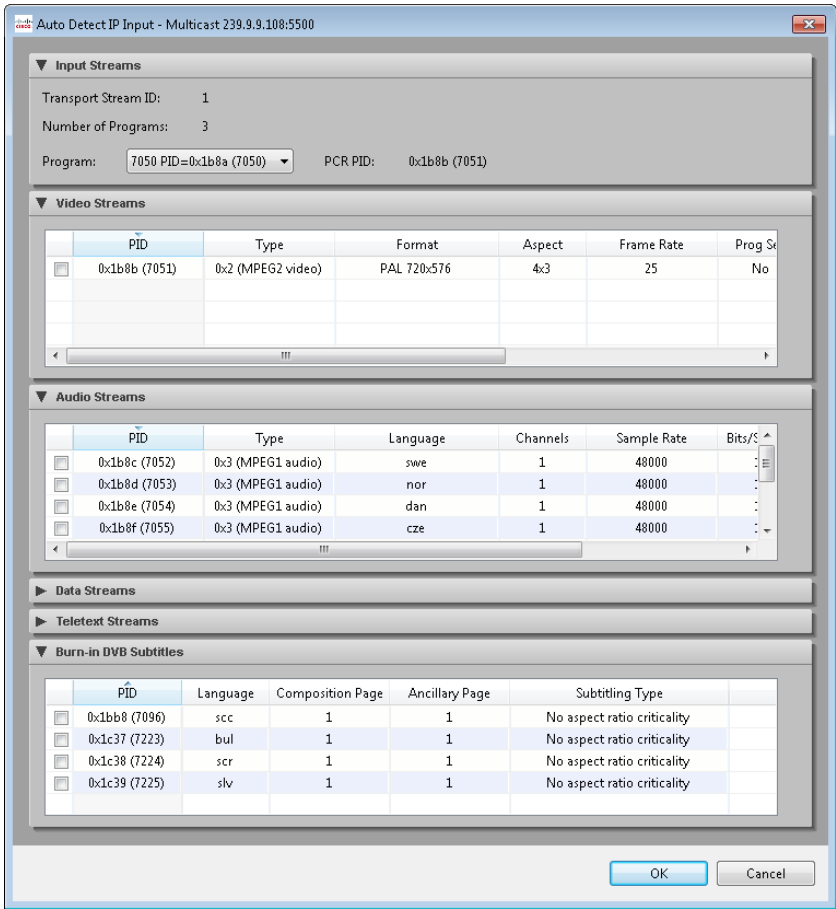
▶ Teletext Track Assignments

▶ Burn-in Bitmap Subtitles

Auto Detect Streams

OK Cancel

Enter a Primary source, a Secondary source, or both. After entering the multicast address and port for the source, you can either manually provide the input stream settings, or click **Auto Detect Streams** to detect any streams. The primary source will be scanned first, and if nothing is returned, the secondary source will be scanned. The secondary source must have the same video PID, resolution, frame rate, and audio PIDs as the primary source. The information for any detected streams is displayed in a new dialog box.



Only elementary streams with supported stream types are presented in the table. Any other elementary streams in the stream are ignored.

Supported stream types:

- 0x02—MPEG2 video
- 0x1b—H264 video
- 0x03—MPEG1 audio (only MPEG1 layer II payload is supported)
- 0x04—MPEG2 audio (only MPEG1 layer II payload is supported)
- 0x0f—AAC audio

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- 0x11—AAC audio
- 0x81—AC-3 audio
- 0x06—AC-3 audio (AC-3 registration descriptor must be present)
- 0x06—EN 300 472/EN 301 775 Teletext (descriptor 0x56 must be present)
- 0x06—EN 300 743 Subtitling (descriptor 0x59 must be present)
- 0x80—MPEG2 video
- 0x86—DPI (SCTE35)

If the input is a MPTS (Multiple Program Transport Stream) then the detected streams table shows the elementary streams of one program at a time. When the specified program is changed, the displayed elementary stream information is updated.

The displayed video bit rate is based on an optional parameter in the stream. If it is not available, the bit rate table entry displays Not specified.

Ingest of mono, 2.0 and 5.1 audio is supported. If the audio is 5.1, it will be automatically downmixed to 2.0 with a normalized downmix matrix. If the audio is mono, it will be automatically copied to 2.0. Then, the normal stereo-to-mono conversions that the encoder supports can be applied if mono output is desired.

If the source contains multiple audio streams, you can map each stream to an audio input channel. Check the check boxes for any combination of streams to map them to input channels, and their PIDs will be populated in the Audio Channel Assignments area.

If the source contains one or more teletext streams, check the check box for any stream that should be available as an output stream.

If the source contains DVB Bitmap Subtitles, you can choose a DVB subtitle track to overlay (burn-in) onto the video by checking the check box for a DVB Subtitle source. The source includes a PID, Language, Composition Page, Ancillary Page, and Subtitling Type. Subtitling Type provides information about appropriate aspect ratio. Leave all check boxes unchecked if no DVB Subtitle sources will be used.

Note: Only bitmapped DVB subtitles will be burned in. Text-based DVB Subtitles are not currently supported.

If the preset includes PlayReady settings, click **Next** to view another screen for those settings. Settings are populated with values from the preset. Use this screen to change

these values to event-specific PlayReady settings. When a generic third-party license provider is being used, only the Provider URL, User ID and Content ID fields may be modified for the event. Content ID is a string that uniquely identifies the event. If BuyDRM or Vualto are being used, other provider-specific fields may be modified for the event.

SDI Router Configuration Area

If any encoder is connected to a SDI Router, you must choose the input source configurations for the encoder by choosing the router and source from the drop-down lists in the SDI Router Configuration area.

After the presets (and routers, if applicable) are configured, click **Next**.

Live Streaming Event Schedule Screen

Summary

Name: Example Event

Resources:

- Encoders

Encoder Group(s):

Primary: N 8300

Preset(s):

TS 7.6.0 SDI

7.6.0 RTMP settings test

Schedule

Specify the live event schedule:

Date/Time

Start: ☐ Now

End: ☐ Indefinitely

Date: 03/26/2014

Time: 9:00:00 AM EDT

Date: 03/26/2014

Time: 10:00:00 AM EDT

Recurring Event: ☐ Enabled

Frequency

☐ Daily ☐ Sun ☐ Mon ☐ Tue ☐ Wed ☐ Thu ☐ Fri ☐ Sat

☒ Weekly Every 1 Week(s)

☐ Monthly Every 1 Month(s)

Date Range

Start Range: 3/26/2014

End Range: 4/10/2014

< Back Next > Finish Cancel

Use the calendars to choose the Start Date and End Date for the event. Use the spin boxes to set the Start Time and End Time for the event. Start time must be in the future for the event to start and End Time must be later than Start Time.

Alternatively, check the **Start Now** check box to have the event start as soon as it is created.

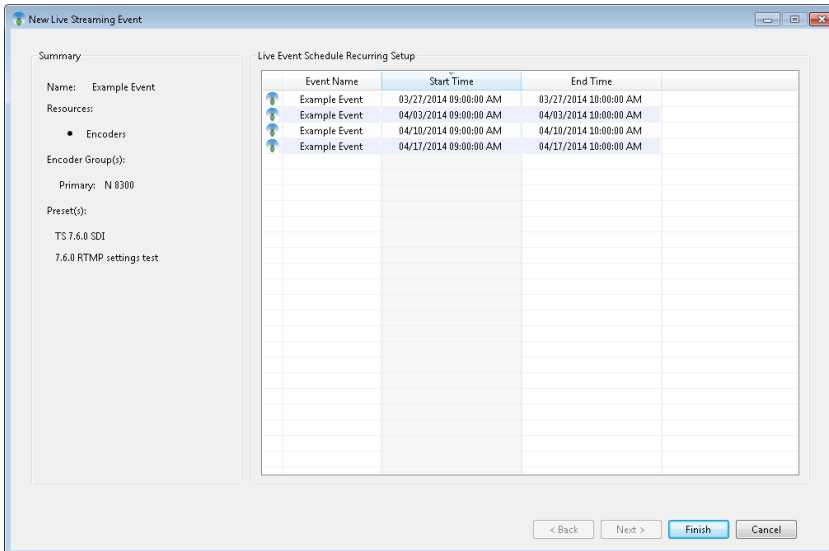
Check the **End Indefinitely** check box if the event has no end time. This event will continue encoding until stopped manually.

Recurring Event Area

A recurring event is simply a way to create multiple events using the same parameters over a range of dates. Once the events are created, each event can be modified individually.

Optionally check the **Enabled** check box in the Recurring Event area to define the event to be a recurring event on a daily, weekly, or monthly basis. You must also specify a range of dates for the recurring event. Start Range must be in the future and End Range must be later than Start Range.

If the streaming event is not a recurring event, click **Finish**. If it is a recurring event, click **Next** to display all instances of the event. This additional screen provides the opportunity to adjust the schedule of future occurrences of the event.



Summary

Name: Example Event

Resources:

- Encoders

Encoder Group(s):

Primary: N 8300

Preset(s):

TS 7.6.0 SDI

7.6.0 RTMP settings: test


Live Event Schedule Recurring Setup

Event Name	Start Time	End Time
Example Event	03/27/2014 09:00:00 AM	03/27/2014 10:00:00 AM
Example Event	04/03/2014 09:00:00 AM	04/03/2014 10:00:00 AM
Example Event	04/10/2014 09:00:00 AM	04/10/2014 10:00:00 AM
Example Event	04/17/2014 09:00:00 AM	04/17/2014 10:00:00 AM

< Back Next > **Finish** Cancel

Click the streaming event icon for any individual event to display the context menu to edit or remove that event. The individual event names and times may also be edited from this dialog box. When this dialog box is displayed, the spawned events have already been created. Therefore, if you click **Cancel**, they will not be deleted but will remain as scheduled events. Click **Finish** when any editing or removing of events is completed.

New Update Event

Click the **New Update Event**  icon in the Events monitor to open the New Update Event dialog box.

New Update Event

View Encoders by Group

Group: All

Encoders

Available Encoders:

Encoder	Version
<input type="checkbox"/> IPVS-ABR-HP2-B01 Cha...	8.0.0.461...
<input type="checkbox"/> IPVS-ABR-HP2-B01 Cha...	8.0.0.461...
<input type="checkbox"/> IPVS-ABR-HP2-B01 Cha...	8.0.0.461...
<input type="checkbox"/> IPVS-ABR-HP2-B01 Cha...	8.0.0.461...
<input type="checkbox"/> IPVS-ABR-HP2-B01 Cha...	8.0.0.461...
<input type="checkbox"/> IPVS-ABR-HP2-B01 Cha...	8.0.0.461...

Add -> <- Remove

Selected Encoders:

Encoder	Version
<input type="checkbox"/> ABR-CMP-UCS3-ESX6 C...	8.0.0.461...
<input type="checkbox"/> ABR-CMP-UCS3-ESX6 C...	8.0.0.461...
<input type="checkbox"/> ABR-CMP-UCS3-ESX6 C...	8.0.0.461...
<input type="checkbox"/> ABR-CMP-UCS3-ESX6 C...	8.0.0.461...
<input type="checkbox"/> ABR-CMP-UCS3-ESX6 C...	8.0.0.461...
<input type="checkbox"/> ABR-CMP-UCS3-ESX6 C...	8.0.0.461...

Update

Current Encoder update file is "CiscoAnyResLive-99.99.99.46444.exe"

Start Date:

June, 2014

Sun	Mon	Tue	Wed	Thu	Fri	Sat
25	26	27	28	29	30	31
1	2	3	4	5	6	7
8	9	10	11	12	13	14
15	16	17	18	19	20	21
22	23	24	25	26	27	28
29	30	1	2	3	4	5

Start Time:

6:08:37 PM

Event Name:

Add


Cancel

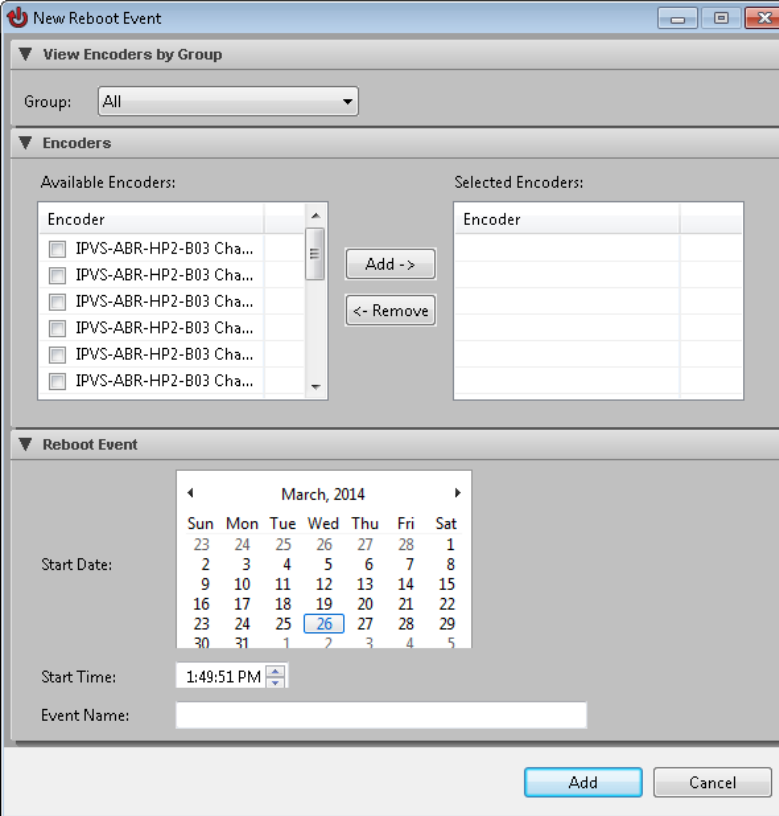
Use this dialog box to choose one or more encoder nodes from the list to have its software updated at a specified date and time. Before creating the event, you must have imported an update file to the Cisco Linear Stream Manager (LSM) Server Node. Choose **File > Import > Encoder Update File** to import an update file.

To create the update event, you can first narrow the list of encoders by choosing an encoder group from the drop-down list. Check the check boxes next to any encoders to update, and then click **Add** to the right of the available encoders list to place them in the list of selected encoders. Finally, specify the date and time of the update event, enter a name for the event, and then click **Add**.

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New Reboot Event

Click the **New Reboot Event**  icon on the Event Creation tab to open the New Reboot Event dialog box.



The dialog box is titled "New Reboot Event" and contains the following sections:

- View Encoders by Group:** A dropdown menu labeled "Group:" with "All" selected.
- Encoders:**
 - Available Encoders:** A list of encoders, each with a checkbox. The list shows six entries: "IPVS-ABR-HP2-B03 Cha...".
 - Selected Encoders:** An empty list.
 - Buttons:** "Add ->" and "<- Remove".
- Reboot Event:**
 - Start Date:** A calendar for March 2014. The date 26 is selected.
 - Start Time:** A time picker set to 1:49:51 PM.
 - Event Name:** A text input field.

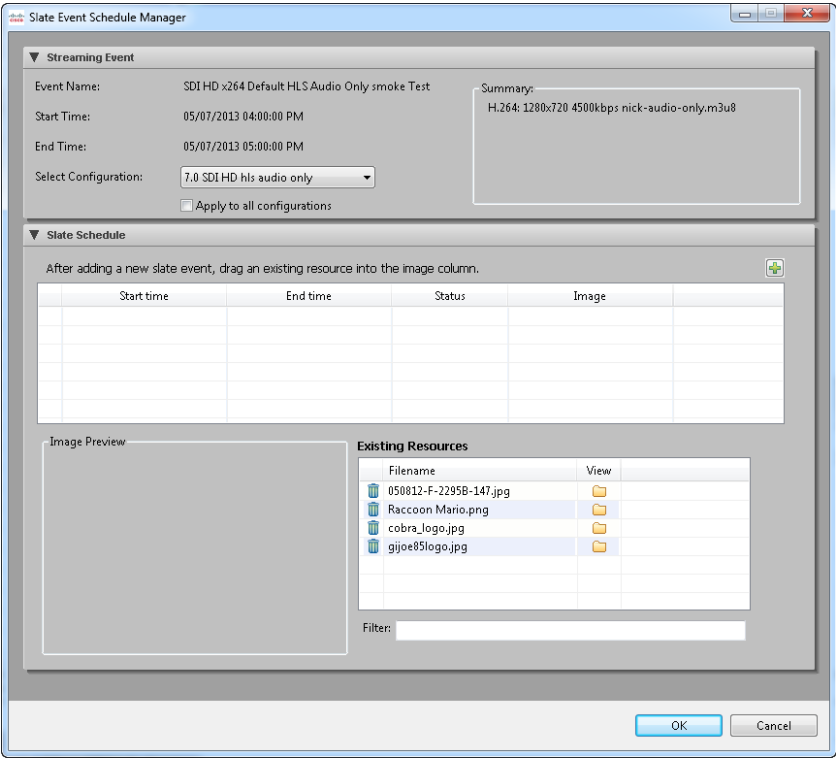
At the bottom right are "Add" and "Cancel" buttons.

Use the New Reboot Event dialog box to choose one or more encoder nodes from the list to be rebooted at a specified date and time. Optionally, narrow the list by choosing an encoder group from the drop-down list. Check the check box next to the encoders that should be rebooted and then click **Add** to the right of the available encoders list to place them in the list of selected encoders. Finally, specify the date and time of the reboot event, designate a name for the event, and then click **Add**.

New Slate Mid Event

Mid events are events that take place during an active streaming event. A Slate Event is a type of mid event where the marker command is sent down to the encoder at a specified time to process an Insert Slate request. Slate events are dependent upon the Slate assets uploaded to the LSM Server through the Manage Slates dialog box in the Content menu. After the slate event is saved, the server is responsible for uploading the slate file to the default slate image directory on AnyRes Live, and for maintaining the slate event schedule throughout the course of the event, processing the Slate On and Slate Off commands at the appropriate intervals.

Use the Manage Slate Events option from the context menu of a scheduled or active event to display the Slate Event Schedule Manager dialog box.



To schedule a slate event during a streaming event, first choose a configuration from the event to replace with the slate, or check the **Apply to all configurations** check box to replace all configurations of the event with the slate for the duration of the slate event.

Next, click the **Add +** icon to add the event, and then click the start and end times and modify them to set the schedule for the slate event. Finally, drag an existing slate file into the empty Image column of the slate events table to choose the slate for the event, and click **OK**.

New SDI Input Source Switch Mid Event

Mid events are events that take place during an active streaming event. An SDI Input Source Switch Event is a type of mid event where the input source is switched from one router input to another. This switch is accomplished by changing the SDI router levels in the attached SDI router at the given scheduled time. This works well in a 24x7 environment where it may be desired to have various input feeds of an SDI router encoded to the same publishing point during one event. Each input feed should be genlocked (precisely synchronized).

Use the Manage SDI Sources option from the context menu of a scheduled or active event to display the SDI Input Source Switch Schedule Manager dialog box.

Streaming Event

Event Name: SDI HD x264 Default HLS Audio Only smoke Test Summary: H.264: 1280x720 4500kbps nick-audio-only.m3u8

Start Time: 05/06/2013 04:00:00 PM

End Time: 05/06/2013 05:00:00 PM

Select Configuration: 7.0 SDI HD hls audio only

☐ Apply to all configurations


SDI Input Source Schedule

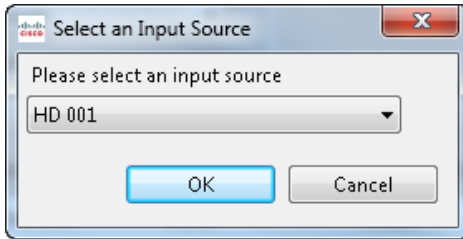
After adding a new SDI Switch event, click on the "Select Input" column to specify the new Input Source Address.

	Start time	End time	Status	Router	Select Input
	05/06/2013 04:00:00 PM	N/A	Draft	SMC Harris Panacea	

OK Cancel

To schedule a SDI source switch event during a streaming event, first choose a configuration from the event to be switched, or check the **Apply to all configurations** check box to replace all configurations of the event with the new input.


Next, click the **Add**  icon to add the event, and click the start time and modify it to set the schedule for the mid event. Click the field in the Select Input column for the event to choose the new input in the Select an Input Source dialog box.



Choose the input source, and then click **OK**.

New Channel

For encoders, a Channel is a template for spawning a live streaming event. A Channel is always based on either an existing Linear Service Template (LST) or one that is created within the New Channel dialog box. Because the encoder and preset configuration have already been provided at this point, the available configuration options for a Channel include an override of the IP Input source (in the case of an IP Preset), or specification of a differing server or publishing point from that stored in the preset. The individual Channels can be started and stopped from the Channels tab on the Monitor tab. The action of starting a Channel with encoders will spawn an individual live streaming event matching its configuration, scheduled to start immediately and continue until the Channel is stopped.

Click the **New Channel**  icon to open the New Channel wizard.

The New Channel wizard has a variable number of screens based on how you choose to specify the configuration of the event.

Channel Introduction Screen

New Linear Service Channel

Linear Service Channel Introduction

Specify a name to describe the Linear Service Channel

Name:

Select a Linear Service Template as a basis for the Linear Service Channel, or create one without a template.

Linear Service Template:

☒ New Linear Service Channel without template

Select the appropriate resources to include in the Linear Service Channel

☐ Encoders ☐ Packagers

< Back Next > Finish Cancel

Name the Channel. Next, choose a Linear Service Template to define the Channel settings. Optionally, click **New Channel without template** and create the Channel without an LST. If your LSM includes both encoders and packagers, check the **Encoders** check box to configure the Channel for encoders. If you are creating this Channel without a template, you will choose encoders and presets in the same way as

an LST. For instructions on creating LSTs, see “Linear Service Template Creation” on page 152.

Click **Next** to proceed to the following screen of the wizard.

Channel Encoders Screen

Summary

Name: Example LSC

Resources:

- Encoders

Encoder(s):

Primary:

IPVS-ABR-HP2-B03 Channel 1

Encoders

Specify primary resource configuration

Primary Encoders:

☐ Pool (N:M) Click the "Add Encoder" button to populate the encoder selection table.

☒ Single (1:N) Encoder Group: All

Encoder Selection

Encoder	Select
IPVS-ABR-HP2-B03 Channel 1 (L72.L...)	<input checked="" type="checkbox"/>
	<input type="checkbox"/>
	<input type="checkbox"/>
	<input type="checkbox"/>
	<input type="checkbox"/>

Specify backup resource configuration

Backup Encoders:

☒ None

☐ Single (1:N)

< Back Next > Finish Cancel

If you created the Channel from an LST, the primary encoders and any backup encoders are already chosen and can be modified. Click **Next** to proceed to the next wizard screen.

Channel Presets Screen

New Linear Service Channel

Summary

Name: Example LSC

Resources:

- Encoders

Encoder(s):

Primary: S6300B Channel 2

S6300B Channel 2

Presets

Specify the preset configuration:

Preset Group: All

Preset Selection

Encoder	Preset	Select	Edit	Override
S6300B Channel 2 (172.18...	1080i H264 8 streams - TS - Defa...			

Preset Summary

H.264: 1280x720 2500kbps
H.264: 960x540 1800kbps
H.264: 960x540 1500kbps
H.264: 640x360 1000kbps
H.264: 480x270 800kbps
H.264: 416x234 400kbps

SDI Router Configuration

Router:

Source:

< Back Next > **Finish** Cancel

If you created the Channel from an LST, the preset chosen in the LST is displayed. You can change the preset, view the preset settings, and modify the available overrides. For information on available overrides, see “Overrides” on page 196. When Channel configuration is complete, click **Finish** to save the Channel.


For information about starting a Channel and its possible states, see “Channels” on page 218.

Channels for Packagers

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For packagers, a Channel defines the Channel configuration and its primary packager resources. Creating a Channel uses the Cisco Linear Stream Manager (LSM) to store this combination of the Channel configuration and the group of packagers chosen to run the Channel when started. This Channel can be configured to have an associated redundant Channel. The Channel can be started and stopped on the Channels tab on the Monitor tab.

New Channel

A Channel configuration for packagers can be based on an existing Channel template or can be created without a template in the New Channel wizard. On the Channels tab on the Configure tab, click the **New Channel**  icon to open the New Channel wizard.

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New Channel

Channel Introduction

Specify a name to describe the Channel

Name:

Select a Channel Template as a basis for the Channel, or create one without a template.

Channel Template: <New Channel without using a template>

< Back Next > Finish Cancel

The New Channel wizard has a variable number of screens based on how you choose to specify the configuration of the event.

Channel Introduction Screen

Enter a name for the Channel. Next, choose a Channel Template to define the Channel settings. Optionally, choose the option to create a new channel without using a template.

Click **Next** to proceed to the following screen of the wizard.

Channel Configuration Screen

Summary

Name: Example

Resources:

- Packagers

Packager Group(s):

Alka

Packagers

Specify resources

Packagers

Pool (NIM)

Packager Group: Alka

Specify channel configuration

Configuration

Channel	HSS	HLS	HDS	DASH	DAS...	Redundant	Edit	Delete
<div>Example</div>		✓	✓					

< Back

Next >

Finish

Cancel

The purpose of this screen is to choose the resource configuration settings and to provide the opportunity to create or modify the Channel settings.

In the Packagers area, click the **Pool** radio button and use the drop-down list to choose the existing Packager group that will be used as the primary resource for the Channel.

Click **Configure** on the right side of the Configuration area. This button will open the Channel editor. Name the Channel and specify settings. See “Channel Template Editor” on page 158 for all Channel setting descriptions. To use encryption, open the PlayReady area of the Output HSS tab and check the **Enable PlayReady** check box, or open the Encryption Settings area of the Output HLS or HDS tab in the Channel editor and check the **Enable Encryption** check box, and choose the encryption type from the drop-down list. Other encryption settings will become available on the next screen of the Channel creation wizard. Click **OK** when you are finished using the Channel editor. After returning to the dialog box, the Channel will be listed in the Configuration table, with check marks indicating which publishers are enabled, as well as whether redundancy is enabled. After the configuration is entered, the Configure button will be disabled. To edit the configuration, click the **Edit** icon to open the Channel editor.

If you are configuring encryption settings, click **Next** to use the Encryption screen. Otherwise, click **Finish** to complete the Channel configuration.

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Channel Encryption Screen

To see the HSS, HLS or HDS encryption settings for a Channel, you must first have enabled an encryption type in the PlayReady area of the Output HSS tab or Encryption Settings area of the Output HLS or HDS tab of the Channel editor. The encryption wizard screen provides encryption settings for the chosen encryption type. For example, the following screen will be displayed for HLS Adobe Access encryption.

Enter the encryption settings for the chosen encryption type. For Adobe Access, the LSM requires that the AdobeAxsConfig.json file be installed in the LSM Server/certificates folder, and the appropriate certificate, policy, and content key files must also be installed on the LSM Server. These files are used to set the global configuration for all Channels configured to use Adobe Access. The AdobeAxsConfig.json file must include the file paths to locate the customer installed Adobe Access configuration files, relative to the LSM Server/certificates folder.

AdobeAxsConfig.json must contain the following required parameters.

Parameter	Description	Example Value
licenseServerURL	The URL associated with the customer's Adobe Access License Server	http://w2k8qa.cisco.com:8080/flashaccessserver/sampletenant
licenseServerCertificate	The Adobe Access License Server issued certificate	Adobe_Flash_Access/license.der

Parameter	Description	Example Value
transportCertificate	The Transport certificate issued by the Adobe Access	Adobe_Flash_Access/transport.der
packagerCertificate	The certificate used by the packager (the packager entity)	Adobe_Flash_Access/packager.pfx
packagerPassword	The Password required to open the Packager Certificate	password
policy	The Policy file issued from the customer's Adobe Access License server	Adobe_Flash_Access/ad-policy.pol
contentKey	The content key used to encrypt the content of each Channel	Adobe_Flash_Access/common-128bitkey.bin

The following is an example AdobeAxsConfig.json file:

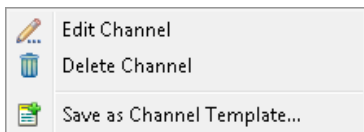
```
{
  "_comment": "ascii string", "licenseServerURL": "http://w2k8qa.cisco.com:8080/flashaccessserver/sampletenant",
  "_comment": "base64 string", "licenseServerCertificate": "Adobe_Flash_Access/license.der",
  "_comment": "base64 string", "transportCertificate": "Adobe_Flash_Access/transport.der",
  "_comment": "base64 string", "packagerCertificate": "Adobe_Flash_Access/packager.pfx",
  "_comment": "ascii password", "packagerPassword": "CiscoPHD8Pkgr",
  "_comment": "base64 string", "policy": "Adobe_Flash_Access/ad-policy.pol",
  "_comment": "base64 string", "contentKey": "Adobe_Flash_Access/common-128bitkey.bin"
}
```

When the configuration is complete, click **Finish** to save the Channel.

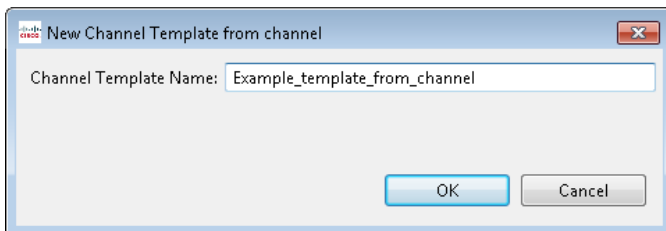
For information about starting a Channel and monitoring its possible states, see “Channels” on page 218.

Saving a Channel as a Template

Once a Channel has been created, its configuration can be saved as a Channel template. Right-click the Channel name on the Channel tab on the Configure tab to display the following context menu.



From this context menu, choose Save as Channel Template. The New Channel Template from *Channel Name* dialog box appears.



Click **OK** to save the configuration of this Channel as a Channel template.

Bulk Import of a Channel Lineup

You can also use a spreadsheet to import an entire lineup of Channels for packagers. To perform this import, use the following procedure:

Note: After the bulk import of the new Channel lineup, the Channels in the spreadsheet file will be the only Channels in the LSM.

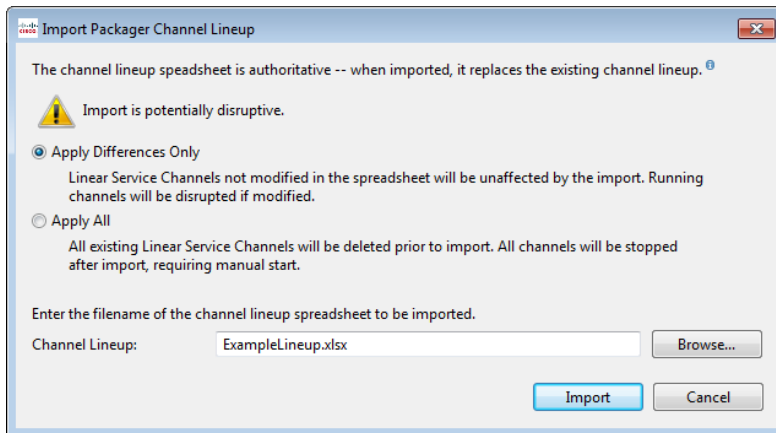
Pre-requisites: Each Channel in the Channel lineup must use a Channel template. Therefore, these templates must be created prior to the bulk import. For example, create one template for SD Channels and another template for HD Channels. The number of input and output streams configured in the template will determine the number of streams created after import. For more information on Channel Templates, see “Channel Template Editor” on page 158. Also, each Channel in the Channel lineup must use a packager group. All packager groups that are used by the Channel lineup

must be created prior to the bulk import. For more information on packager groups, see “Packager” on page 12. Publishing points must also be created prior to import.

1. Choose **Generate Sample** from the File menu to generate a sample .xlsx spreadsheet as a guide for importing a Channel lineup.
2. Save this file in a location with access to spreadsheet software that can edit an .xlsx file.
3. Edit the sample Channel lineup file with the spreadsheet software to define your particular Channel lineup. Values specified in the spreadsheet file will override any corresponding values in the Channel template.

Note: Ensure that all Channel Templates and packager groups have been previously created in the LSM, and that their names in the spreadsheet match exactly with their names in the LSM. These names are case sensitive.

4. Choose **File > Import > Channel Lineup**, which will display the Import Packager Channel Lineup dialog box.



5. If Channels currently exist in the LSM, click the **Apply Differences Only** radio button if you want unmodified Channels to be unaffected by the import and modified Channels to be stopped, updated, and restarted. Make sure that these Channels are included in the spreadsheet exactly as they exist in the LSM. To delete all existing Channels before importing the Channel lineup in the


spreadsheet, click the **Apply All** radio button. If you choose Apply All, all Channels will be stopped after import.

6. Browse to your edited Channel lineup .xlsx file.
7. Click **Import** to create the Channel lineup.
8. The LSM will validate the Channels in the spreadsheet and create all valid Channels. If any Channels are not valid, the LSM will report errors. To correct these errors, edit the spreadsheet file and import again. To start the created Channels, use the Channels tab on the Monitor tab. For information on starting, stopping, and monitoring Channels, see “Channels” on page 218.

Channel Dual Management of Encoder and Packagers

You can also use a Channel to manage both an encoder configured to multicast and one or more packagers configured to use the multicast output as input.

For dual management, first define an encoder preset that configures the encoder to broadcast an H.264 adaptive transport stream on a valid multicast address/ port combination. For detailed information on encoder presets, see “Preset Editor” on page 33.

Once the preset has been stored in the LSM, create a Channel without template. The Channel must include both encoder and packager resources. The encoder chosen in the Channel must use the preset configured above. Next, click **Add Channel Service**  on the right side of the Channel Resources area of Channel resource configuration. This button opens the Channel editor. Input settings will be populated with the TS configuration in the encoder preset. Name the Channel, specify output settings, and click **OK** when finished. See “Channel Template Editor” on page 158 for configuration setting descriptions.

Starting a Channel with both encoders and packagers configured will send the preset and start command to the encoder first, followed by sending the Channel to the packager a default 10 seconds later. If the encoder fails while encoding, and no backup nodes are configured, then the underlying event control logic will stop all parts of the Channel, including the Channel running on the packager.

Overview

Monitoring 7

The Monitor tab displays tabs with information about Channels, Events, Encoders, Packagers, and System Logs. If you choose the AnyRes Live or AnyRes Packaging only views at login or from the View menu, only the tabs that apply to that view are displayed.

Channels

The Channels tab contains a table of configured Channels. See “Channels for Packagers” on page 210, or “Event Creation and Channels for Encoders” on page 191 for more information on creating Channels.

EventsRecurring EventsChannels

Filter by Name:

Page 3 of 4

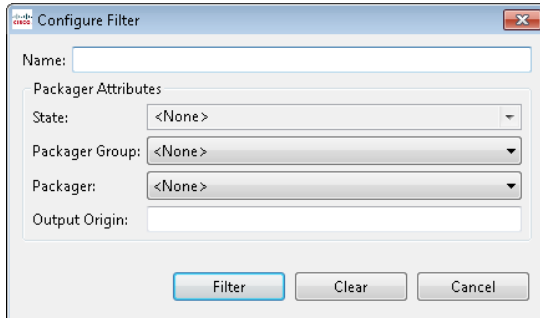
Name	Start/Stop	Resource/Group	State	Input State
all-publishers				
ats-esam				
ats-esam		Aeon-52		Running Ok
att_hd				
channel				
channel		ESAM		Not Running
hds-226-250-1-1				
hds_cc				
hds_cc		Group-1-80		Not Running
hls-226-250-1-1				
hls_alt_dcm				
hls_alti_audio_cmp				
hls_nfs				

Click the Expand all in tree and Collapse all in tree icons to expand or collapse all entries in this tab. Click the Start all , Stop all , and Delete all Channels icons and confirm to start, stop, or delete all Channels across all pages. If you start or stop all Channels, any Channel that is currently already in that state will be unaffected. If you

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delete all Channels, they must all be stopped. Click the Previous Page and Next Page arrow ⇐⇒ icons to view any additional pages of Channels.

Enter a string in the Filter by Name field to filter the view of the Channels tab to Channels that begin with the string, or click the **Advanced Filter** 🧐 icon to open the Configure Filter dialog box.



To filter the current view, choose any combination of the name field and packager attributes as filtering criteria. Any specified criteria will be logically ANDed to determine filtering matches. You can filter by name string (matching the start of the name), state, packager group, packager, or output origin (matching the start of the IP or server address of the origin server).

Click **Filter** to filter the view based on the criteria specified. Click **Clear** to remove all filtering on the current view. Click **Cancel** to close the dialog box without changing filtering.

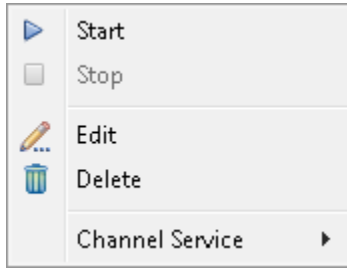
To start a configured Channel, click the **Start** ▶ icon in the Channel row.

When a Channel is started, the Cisco Linear Stream Manager (LSM) creates one additional instance of this Channel if Redundancy is configured. A Channel is then sent to the available AnyRes Packager with the most available capacity within the assigned Packager group. Redundant Channel instances are distributed to different packagers, if available. The LSM automatically appends either “_1” or “_2” to each instance of a redundant Channel to create unique names for each. The Channel will continue running until manually stopped.

When an encoder Channel is started, the LSM creates an encoder node that runs a streaming event on a configured encoder.

The following information is shown for each Channel:

- Right-click the Channel row to display a context menu of available actions.

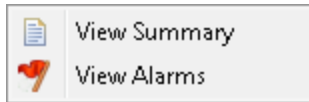


Start/Stop—Based on the current state of the Channel, starts or stops the Channel. To start or stop multiple Channels simultaneously, select multiple Channel rows, right-click one of the rows, and choose Start or Stop from the context menu.

Edit—Opens the Edit Channel dialog box to edit the Channel configuration in the Channel editor. See “Channel Template Editor” on page 158 for configuration setting descriptions.

Delete—Removes the Channel from the Channels tab. This option is not available while the Channel is running.

Channel Service—Displays another context menu of actions that can be performed on any Channel services associated with the Channel.








View Summary—Displays a summary of Channel settings.

View Alarms—Displays active alarms, an alarm summary, and historical alarms for the Channel. For more information about these alarms, see “Packager Alarms” on page 279.

- **Name**—Displays the name of the Channel.
- **Start/Stop**—Displays an icon that can be clicked to start or stop the Channel.
- **State**—Displays the aggregate state of a Channel. Aggregate states represent the overall health of a Channel based on its inputs and outputs. When a Channel has an associated redundant Channel, the Channel state is a combination of the two Channel states. When a Channel and its associated redundant Channel share a packager and each are “Operating as configured”, the Channel state will be “Service degraded” because redundancy is compromised. Possible states are:

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State Icon	Description	Input/Output State
	Operating as configured. Publishing or Receiving	Input and all enabled outputs (publishers) Running OK
	Service degraded	At least one (Input or Output) Running With Alarms, or Unknown
	In transition	At least one (Input or Output) in transition state (Creating, Deleting, Restarting, Relocating, Pending Create)
	Not operational, but should be	At least one (Input or Output) in failed state (Create Failed, Restart Failed, Relocate Failed) OR Homeless state
	Administratively disabled	Input and all enabled outputs (publishers) not running

If global packager alarm triggers have been enabled, as the LSM monitors the health of a Channel, the LSM will relocate a Channel that has met the threshold conditions of the enabled alarms. For more information about these alarm triggers, see “Properties” on page 29.

The following information is shown for each Channel:

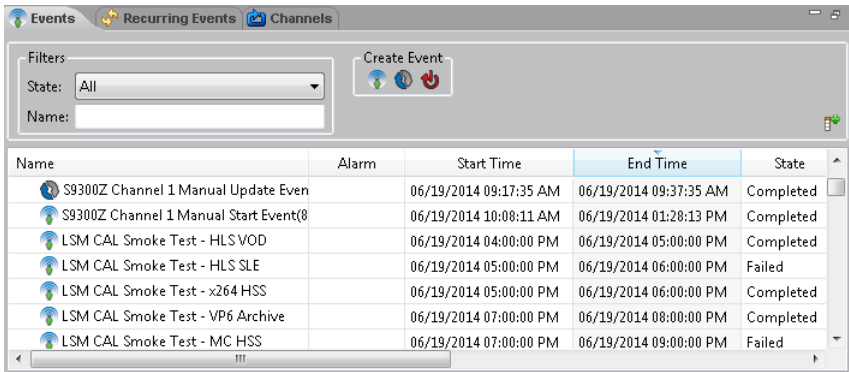
- **Name**—Displays the name of the Channel. Redundant Channels have the same name with _1 or _2 appended.
- **Resource/Group**—For stopped Channels, displays the name of the resource or group/pool that has been configured for the Channel. For started Channels, displays the name of the packager that is running the Channel.
- **Input State**—Displays the state of the Channel input.
- **Output States (HSS, HLS, HDS, DASH, DASH-TS)**—Displays the state of each respective output.

The following information is shown for each encoder node:

- **Name**—Displays the name of the encoder node.
- **Resource/Group**—For stopped Channels, displays the name of the resource or group that has been configured for the Channel. For started Channels, displays the encoder that is running the event.

Events

The Events tab contains tables of information, as well as buttons to create new scheduled events. The columns of the tables are movable and sortable. To move a column, drag the column heading to the desired location. To sort the information in the table by a particular column, click the column heading.



Events can be filtered by Scheduled, Active, Completed, Draft, or Failed states.

Event Recovery

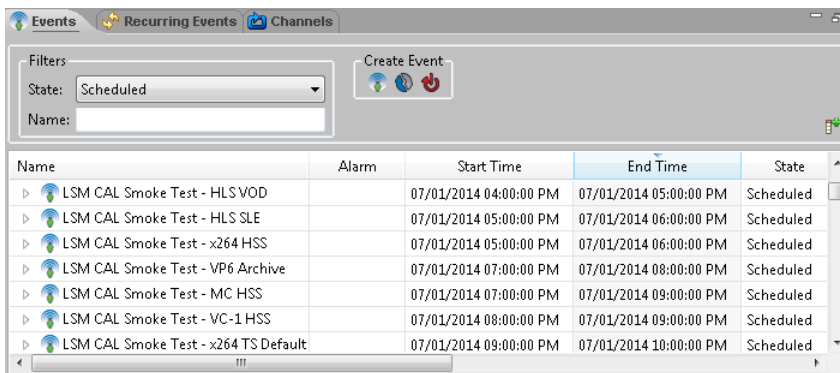
If communication is interrupted between LSM and the encoder during an active streaming event, the encoder is listed in Failed state in the Encoders tab, and the active event is listed as Failed in the Events tab. If communication between LSM and the encoder is restored, the LSM lists the encoder in Pending state because it is still running the event that was started before the interruption. When an encoder transitions to Pending state, the LSM will attempt to match a recently failed event to the event running on the encoder. If successful, the LSM will restore the event as an Active event, will display any associated Channel as Running, and will display the encoder state as Encoding. This automatic event recovery process may take some time, depending on the number of encoders being monitored and the length of loss of communication between the LSM and the encoder. If an encoder is currently in Pending state and you believe it is running an event that was started by the LSM, right-click the encoder in the Encoders tab to display its context menu, and choose the Acquire Running Event option to manually attempt to recover the event.

Scheduled or Draft Events

The Scheduled Events monitor displays all events that are currently waiting to be started. Draft events must be updated before they are ready to be run. If an event is in

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draft state, the LSM has detected a scheduling conflict which must be resolved by changing the start and/or end time of the event. Once the scheduling conflict is resolved, the event will move to the scheduled or active state.



Name	Alarm	Start Time	End Time	State
▶ LSM CAL Smoke Test - HLS VOD		07/01/2014 04:00:00 PM	07/01/2014 05:00:00 PM	Scheduled
▶ LSM CAL Smoke Test - HLS SLE		07/01/2014 05:00:00 PM	07/01/2014 06:00:00 PM	Scheduled
▶ LSM CAL Smoke Test - x264 HSS		07/01/2014 05:00:00 PM	07/01/2014 06:00:00 PM	Scheduled
▶ LSM CAL Smoke Test - VP6 Archive		07/01/2014 07:00:00 PM	07/01/2014 08:00:00 PM	Scheduled
▶ LSM CAL Smoke Test - MC HSS		07/01/2014 07:00:00 PM	07/01/2014 09:00:00 PM	Scheduled
▶ LSM CAL Smoke Test - VC-1 HSS		07/01/2014 08:00:00 PM	07/01/2014 09:00:00 PM	Scheduled
▶ LSM CAL Smoke Test - x264 TS Default		07/01/2014 09:00:00 PM	07/01/2014 10:00:00 PM	Scheduled

Click the ▶ icon for the event to expand the tree to display information for each encoder associated with the event. The following information is shown for each scheduled event:

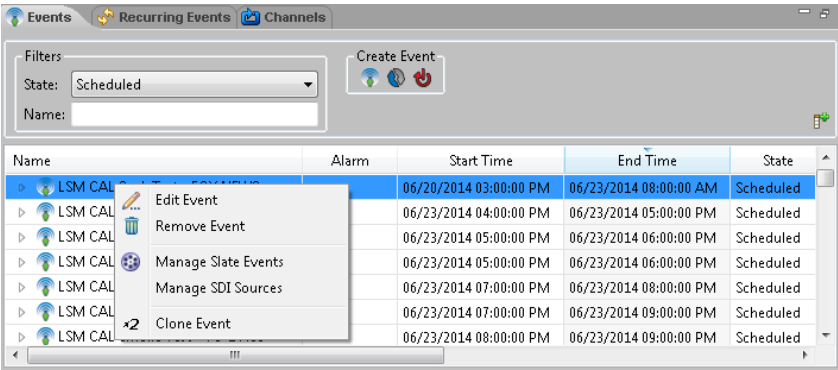
- Event Type icon—Indicates the type of event that is scheduled:
 - ▶ Streaming—The encoder will encode one or more streams, according to the preset.
 - ▶ Update—A software update command will be sent to the encoder at the specified time.
 - ▶ Reboot—The encoder will reboot at the specified time.
- Event Name—Displays the name of the event.
- Alarm—If an alarm is active, a red flag icon is displayed in this column. When the alarm is cleared, no icon is displayed.
- Start time—The time the event is scheduled to start.
- End time—The time the event is scheduled to complete.
- State—The state of the event. The event can be Scheduled or Draft. A Draft state indicates an event that will not run as it is because it has a scheduling conflict, it does not have an encoder node associated with it, or it was cloned and has not yet been changed from the original. Events that are in the draft state must be updated so that they become scheduled. Only events in the scheduled state are ready to be run.
- Encoder/Encoder Group—Name of the encoder or group chosen for the event.

- Redundancy—Primary or Backup.
- Preset—Name of the preset being used for the event.

Click the event type icon to display a context menu of available actions.

Streaming Event Context Menu

Right-click the event to display a context menu of available actions. The following context menus apply to each event type.



Edit Event—Displays the Edit a Streaming Event dialog box.

Remove Event—Deletes the event from the schedule.

Manage Archives—For H.264 TS Output events that have been previously enabled for Manual Archive, use this option to display the Archive Schedule Manager dialog box.

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Archive Schedule Manager

▼ Streaming Event

Event Name: SDI mid event archive test Summary: H.264: 320x240 500kbps

Start Time: 03/24/2014 08:34:54 AM

End Time: 03/24/2014 09:00:00 AM

Select Configuration: Start and Stop Archive Test - x264 ▼


Preset: Start and Stop Ar... - x264 lsm trunk

Select a stream: H.264 Stream 1 Output ▼

▼ Archive Schedule

	Start time	End time	Status	Filename
✖	03/24/2014 08:36:00 AM	03/24/2014 08:40:00 AM	Active	C:\PerfLogs\StartStopArchi...
🗑	03/24/2014 08:41:00 AM	03/24/2014 08:43:00 AM	Scheduled	C:\PerfLogs\StartStopArchi...
🗑	03/24/2014 08:50:00 AM	03/24/2014 09:00:00 AM	Scheduled	C:\PerfLogs\StartStopArchi...

OK Cancel

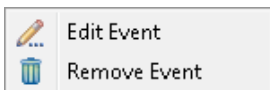
For the scheduled event, specify the configuration and stream to be archived, click the **Add Archive Event**  icon, and enter the start time, end time, and filename for any archives to be saved for this event. You can also edit or delete an existing archive event.

Manage Slate Events—Set up a slate event to take place during this event. See “New Slate Mid Event” on page 204 for details.

Manage SDI Sources—Set up a SDI Input Source Switch event to switch the input source from one router input to another. See “New SDI Input Source Switch Mid Event” on page 205 for details.

Clone Event—Displays the Clone a Streaming Event dialog box to create one or more copies of the event. Creating cloned events is similar to the way that recurring events work. To clone an event, enter the number of exact copies of the event to create. Each of these events is created in a draft state because the times and nodes are the same as the event used to create them. Click **Next** to display all cloned events. Click the event values to edit them, and then click **Finish** to schedule the cloned events.

Update/Reboot Event Context Menu

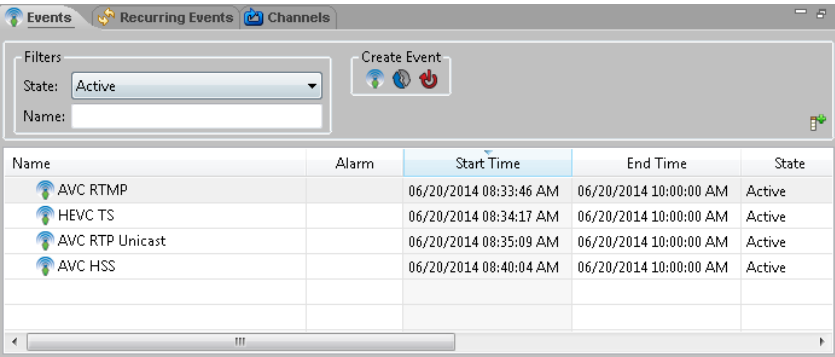



Edit Event—Displays the Edit an Update Event or Edit a Reboot Event dialog box




Remove Event—Removes the event from the schedule

Active Events

The Active Events monitor displays all events that are currently taking place.



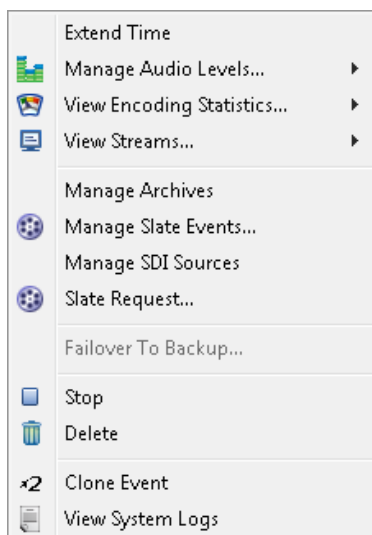
Click the  icon for the event to expand the tree to display information for each encoder associated with the event. The following information is shown for each active event:

- Event Type icon—Indicates the type of event that is scheduled.
 -  Streaming—The encoder will encode one to four streams, according to the preset.
 -  Update—A software update command will be sent to the encoder at the specified time.
 -  Reboot—The encoder will reboot at the specified time.
- Event Name—Displays the name of the event.
- Alarm—Displays an event alarm. There is currently only one event alarm and that alarm indicates that a node has failed within an event. If an event has an alarm, it will be highlighted as red in this monitor and in the Guide.
- Start time—The time the event is scheduled to start.
- End time—The time the event is scheduled to complete.
- State—Active.
- Encoder/Encoder Group—Name of the encoder or group that is running the event.

- Redundancy—Primary or Backup.
- Preset—Name of the preset being used for the event.
- Stream 1 through 4—Link to the output stream. If multiple streams are being produced, there will be a link for each.
- Archive location—File path and name for any defined archive file for the stream.

Streaming Event Context Menu—Active Events

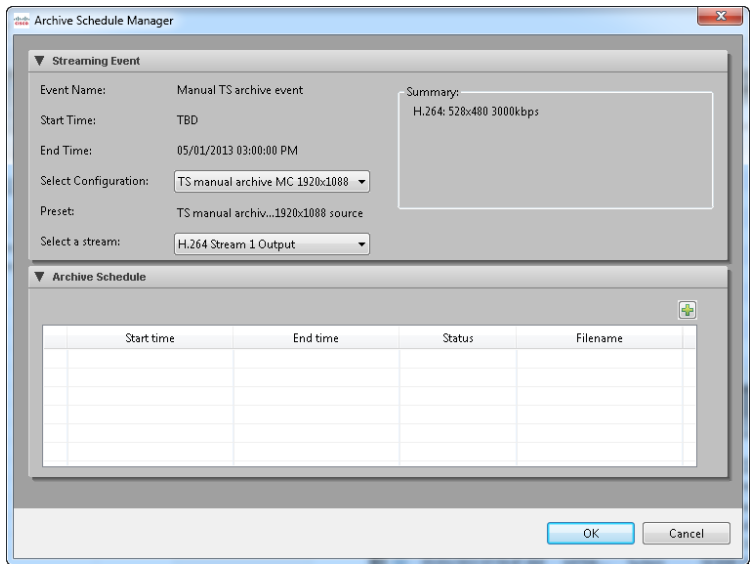
Right-click the event to display a context menu of available actions.




Extend Time—Displays a dialog box to extend the time of the active event. Extending an event can fail if there is another event scheduled on one of the nodes in the event and the extended time overlaps the start of those events. This option is not available for events with an indefinite end time.

Manage Audio Levels—Choose an encoder node to display a dialog box displaying audio channels. At the top of the dialog box, the current decibels of each channel are displayed in real time. Use the slider controls beneath each channel decibel display to add/subtract the relative amount of decibels from what is currently being output. For example, if channel 1 is displaying 50 dB, and channel 1 is set to 10, the overall dB level should increase to 60 dB. Manage Audio Levels will not be available for events streaming on IP Input AnyRes Live models.

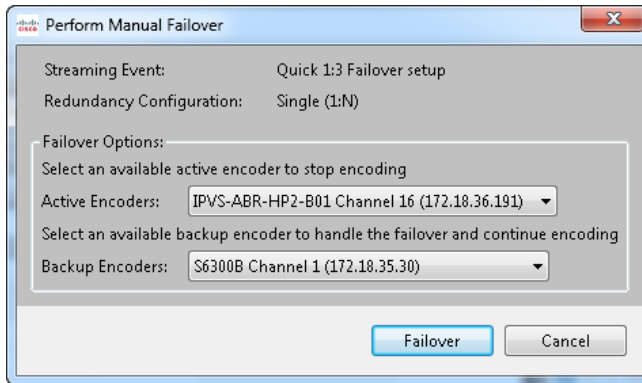
- View Encoding Statistics—Choose a specific node to view its encoding statistics.
- View Streams—Choose an encoder node, and then choose from one of the current streams that are being pushed or pulled from the specified node to view the output.
- Manage Archives—For H.264 TS Output events that have been previously enabled for Manual Archive, use this option to display the Archive Schedule Manager dialog box.



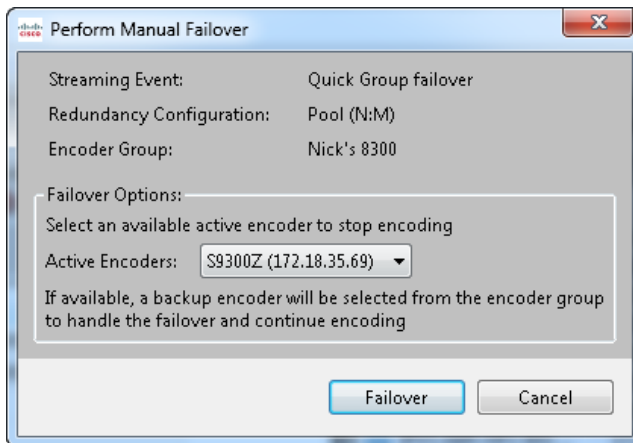
- For the active event, choose the configuration and stream to be archived, and then click the **Add Archive Event**  icon to populate and modify the start time, end time, and filename for any archives to be saved for this event. You can also edit the end time of an archive event that is in progress, or edit or delete an existing archive event that has not started yet.
- Manage Slate Events—Set up a slate event to take place during this event. See “New Slate Mid Event” on page 204 for details.
 - Manage SDI Sources—Set up a SDI Input Source Switch event to switch the input source from one router input to another. See “New SDI Input Source Switch Mid Event” on page 205 for details.
 - Slate Request—See “Slate Request” on page 240 for details on sending a slate request to replace output video with an image file at a specified time.

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Failover to Backup—Choose an active encoder for manual failover to a backup encoder.



For a Single (1:N) redundancy configuration, choose an available backup encoder to handle the failover and continue encoding.



For a Pool (N:M) redundancy configuration, the LSM will choose a backup encoder from the encoder group.

Click **Failover** to begin the manual failover.

If H.264 RTMP streams are enabled and the encoder experiences connection issues to the Flash Media server during a failover, the video outage time may be increased while the Flash streams try to reconnect.

Stop—Stops the active event. The event will be listed in the completed events monitor and details of the event will be retained. If this event was started by a Channel, this option will display Stop Channel name.

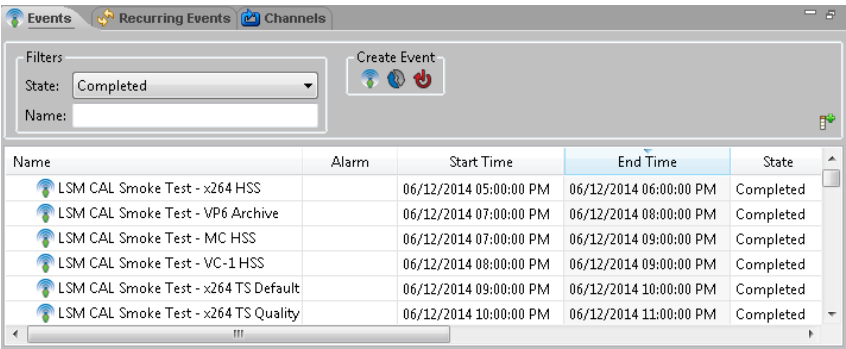
Delete—Deletes the active event. Deleting the event will stop the event and remove all data that was associated with that event.

Clone Event—Displays the Clone a Streaming Event dialog box to create one or more copies of the event. Creating cloned events is similar to the way that recurring events work. To clone an event, enter the number of exact copies of the event to create. Each of these events is created in a draft state because the times and nodes are the same as the event used to create them. Click **Next** to display all cloned events. Click the event values to edit them, and then click **Finish** to schedule the cloned events.

View System Logs—Displays the list of system logs specific to this event.


Completed or Failed Events




The Completed Events monitor displays all events that have completed. The Failed events monitor displays all events that have failed.



The screenshot shows a software window titled 'Events' with tabs for 'Events', 'Recurring Events', and 'Channels'. The 'Events' tab is active. Below the tabs is a 'Filters' section with a 'State' dropdown menu set to 'Completed' and a 'Name' text box. To the right of the filters is a 'Create Event' button with a plus icon. Below the filters is a table with the following columns: Name, Alarm, Start Time, End Time, and State. The table contains six rows of event data, all with a state of 'Completed'.

Name	Alarm	Start Time	End Time	State
LSM CAL Smoke Test - x264 HSS		06/12/2014 05:00:00 PM	06/12/2014 06:00:00 PM	Completed
LSM CAL Smoke Test - VP6 Archive		06/12/2014 07:00:00 PM	06/12/2014 08:00:00 PM	Completed
LSM CAL Smoke Test - MC HSS		06/12/2014 07:00:00 PM	06/12/2014 09:00:00 PM	Completed
LSM CAL Smoke Test - VC-1 HSS		06/12/2014 08:00:00 PM	06/12/2014 09:00:00 PM	Completed
LSM CAL Smoke Test - x264 TS Default		06/12/2014 09:00:00 PM	06/12/2014 10:00:00 PM	Completed
LSM CAL Smoke Test - x264 TS Quality		06/12/2014 10:00:00 PM	06/12/2014 11:00:00 PM	Completed

Click the  icon for the event to expand the tree to display information for each encoder associated with the event. The following information is shown for each completed event:

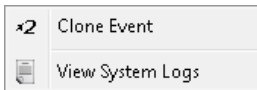
- **Event Type icon**—Indicates the type of event that is scheduled.
 -  **Streaming**—The encoder will encode one to four streams, according to the preset.
 -  **Update**—A software update command will be sent to the encoder at the specified time.
 -  **Reboot**—The encoder will reboot at the specified time.

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- Event Name—Displays the name of the event.
- Start time—The time the event started.
- End time—The time the event completed.
- State—Status of the completed event: Completed or Failed. A manually stopped event will have a status of Completed.
- Error description—If the event failed, displays the description of the error.
- Encoder/Encoder Group—Encoders that were used for the event.
- Redundancy—Primary or Backup
- Preset—For an encoder, name of the preset that was used for the event.
- Archive location—File path and name for any defined archive files for the stream or streams.

Streaming Event Context Menu

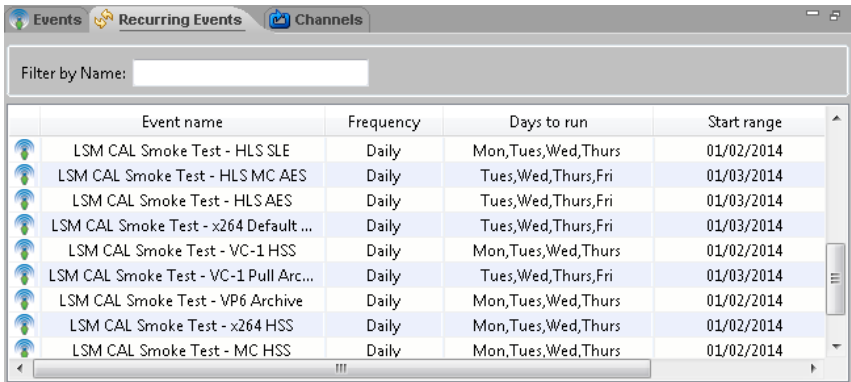
Right-click the event to display a context menu of available actions.



Clone Event—Displays the Clone a Streaming Event dialog box to create one or more copies of the event. Creating cloned events is similar to the way that recurring events work. To clone an event, enter the number of exact copies of the event to create. Each of these events is created in a draft state because the times and nodes are the same as the event used to create them. Click **Next** to display all the cloned events. Click the event values to edit them, and then click **Finish** to schedule the cloned events.

View System Logs—Displays the list of system logs specific to this event.

Recurring Events



The screenshot shows a window titled 'Events' with tabs for 'Events', 'Recurring Events', and 'Channels'. The 'Recurring Events' tab is active. Below the tabs is a 'Filter by Name:' text box. The main area contains a table with the following data:

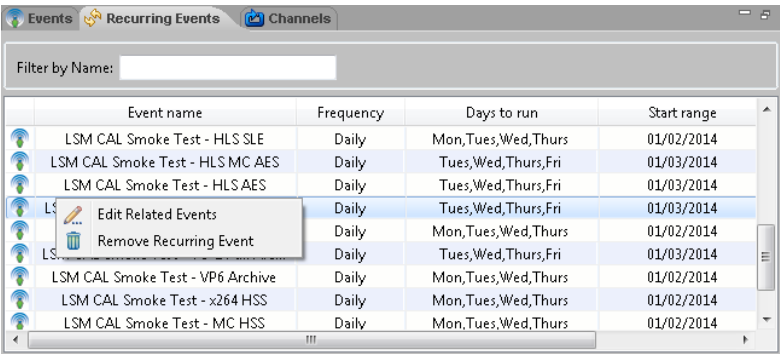
	Event name	Frequency	Days to run	Start range
	LSM CAL Smoke Test - HLS SLE	Daily	Mon,Tues,Wed,Thurs	01/02/2014
	LSM CAL Smoke Test - HLS MC AES	Daily	Tues,Wed,Thurs,Fri	01/03/2014
	LSM CAL Smoke Test - HLS AES	Daily	Tues,Wed,Thurs,Fri	01/03/2014
	LSM CAL Smoke Test - x264 Default ...	Daily	Tues,Wed,Thurs,Fri	01/03/2014
	LSM CAL Smoke Test - VC-1 HSS	Daily	Mon,Tues,Wed,Thurs	01/02/2014
	LSM CAL Smoke Test - VC-1 Pull Arc...	Daily	Tues,Wed,Thurs,Fri	01/03/2014
	LSM CAL Smoke Test - VP6 Archive	Daily	Mon,Tues,Wed,Thurs	01/02/2014
	LSM CAL Smoke Test - x264 HSS	Daily	Mon,Tues,Wed,Thurs	01/02/2014
	LSM CAL Smoke Test - MC HSS	Daily	Mon,Tues,Wed,Thurs	01/02/2014

Recurring events are displayed if the event that has been defined to recur has spawned scheduled events that are not yet completed. Each individual scheduled event spawned by the recurring event will be listed in the Scheduled Events monitor.

The following information is shown for each recurring event:

- Event Type icon—Indicates the type of event that is scheduled:
 Streaming—The encoder will encode one to four streams, according to the preset.

Right-click the event to display a context menu of available actions.



The screenshot shows the same 'Recurring Events' window as before, but with a right-click context menu open over the 'LSM CAL Smoke Test - HLS AES' row. The menu has two options: 'Edit Related Events' and 'Remove Recurring Event'.

	Event name	Frequency	Days to run	Start range
	LSM CAL Smoke Test - HLS SLE	Daily	Mon,Tues,Wed,Thurs	01/02/2014
	LSM CAL Smoke Test - HLS MC AES	Daily	Tues,Wed,Thurs,Fri	01/03/2014
	LSM CAL Smoke Test - HLS AES	Daily	Tues,Wed,Thurs,Fri	01/03/2014
	LSM CAL Smoke Test - x264 Default ...	Daily	Tues,Wed,Thurs,Fri	01/03/2014
	LSM CAL Smoke Test - VC-1 HSS	Daily	Mon,Tues,Wed,Thurs	01/02/2014
	LSM CAL Smoke Test - VC-1 Pull Arc...	Daily	Tues,Wed,Thurs,Fri	01/03/2014
	LSM CAL Smoke Test - VP6 Archive	Daily	Mon,Tues,Wed,Thurs	01/02/2014
	LSM CAL Smoke Test - x264 HSS	Daily	Mon,Tues,Wed,Thurs	01/02/2014
	LSM CAL Smoke Test - MC HSS	Daily	Mon,Tues,Wed,Thurs	01/02/2014


Edit Related Events—Displays the Event Details dialog box with all scheduled events spawned by this recurring event. Use this dialog box to individually modify or remove the events.

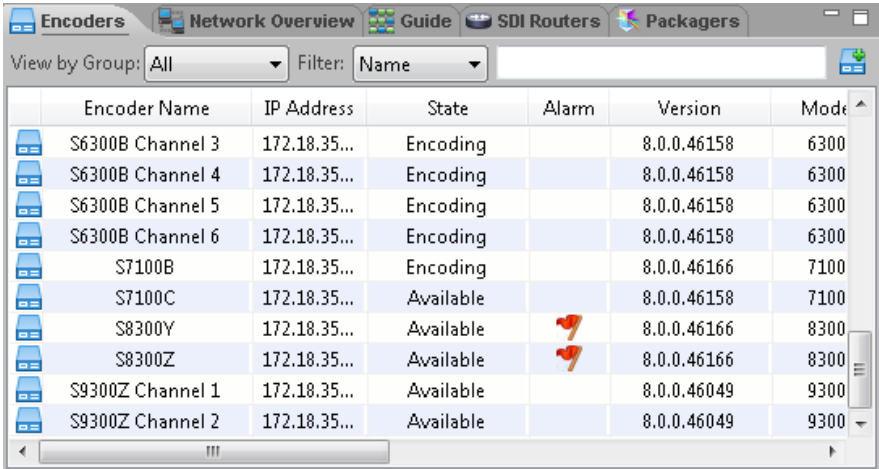
Remove Recurring Event—Deletes all scheduled events spawned by the recurring event and deletes the recurring event.

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- Event Name—Displays the name of the parent event.
- Frequency—Displays the frequency at which an event will recur: Daily, Weekly, or Monthly.
- Days to run—Displays the days of the week the event will run for Daily events.
- Start range—The starting date when the event is scheduled to run.
- End range—The last date when the event may run.

Encoders

The Encoders tab lists the encoder nodes and information specific to each. Use the View by Group or Filter drop-down lists to restrict the list of encoders. Click the **New Encoder**  icon to add a new encoder.



The screenshot shows the 'Encoders' tab in a software interface. At the top, there are tabs for 'Encoders', 'Network Overview', 'Guide', 'SDI Routers', and 'Packagers'. Below the tabs, there are two dropdown menus: 'View by Group:' set to 'All' and 'Filter:' set to 'Name'. A search bar is also present. The main area displays a table of encoder nodes. Each row has a small icon on the left, followed by the Encoder Name, IP Address, State, Alarm status, Version, and Mode. The table lists several encoders, including multi-channel models (S6300B, S9300Z) and single-channel models (S7100B, S7100C, S8300Y, S8300Z). The states are 'Encoding' or 'Available'. Some 'Available' entries have a red heart icon in the Alarm column.

	Encoder Name	IP Address	State	Alarm	Version	Mode
	S6300B Channel 3	172.18.35...	Encoding		8.0.0.46158	6300
	S6300B Channel 4	172.18.35...	Encoding		8.0.0.46158	6300
	S6300B Channel 5	172.18.35...	Encoding		8.0.0.46158	6300
	S6300B Channel 6	172.18.35...	Encoding		8.0.0.46158	6300
	S7100B	172.18.35...	Encoding		8.0.0.46166	7100
	S7100C	172.18.35...	Available		8.0.0.46158	7100
	S8300Y	172.18.35...	Available		8.0.0.46166	8300
	S8300Z	172.18.35...	Available		8.0.0.46166	8300
	S9300Z Channel 1	172.18.35...	Available		8.0.0.46049	9300
	S9300Z Channel 2	172.18.35...	Available		8.0.0.46049	9300

Right-click the encoder entry in the Encoders monitor to display a context menu with available configuration options. For more information on the options provided, see “Encoder Configuration” on page 236.

The following information is shown for each encoder:

- Encoder name—Name of the encoder. For multi-channel encoder models, each channel will have its own name entry and is treated as a separate node, except for cases that affect multiple channels, such as a reboot and update.
- IP address—IP address of the encoding node.

Note: To change the IP address of an encoder node, see the Cisco AnyRes Live Family User Guide for instructions. To view the guide, open the web interface for the encoder and click **Help**.
- State—Possible states are:
 - Available—The encoder is available for a streaming event.
 - Scheduled—The encoder has an upcoming scheduled event within 30 minutes.
 - Encoding—The encoder is encoding either a manual start event or a scheduled streaming event that was started within the LSM.

Stopped—The encoder has just finished encoding.

Updating/Rebooting—The encoder is running an update or reboot event.


Searching/Connecting—The server is attempting to establish a connection with the encoder.

Failed—The LSM/Server cannot communicate with the encoder.

Offline—The encoder has been taken offline by a user. While offline, the encoder cannot be scheduled for an event.

Pending—The encoder is encoding an event that was started outside of the LSM. The encoder can also be in Pending state if communication was interrupted between LSM and the encoder during an active streaming event, which resulted in the encoder being listed in Failed state and the event being listed as Failed in the LSM. If communication between LSM and the encoder is restored, the LSM lists the encoder in Pending state because it is still running the event that was started before the interruption. When an encoder transitions to Pending state, the LSM will attempt to match a recently failed event to the event running on the encoder. If successful, the LSM will restore the event as an Active event, will display any associated Channel as Running, and will display the encoder state as Encoding. See “Event Recovery” on page 222 for additional details.

Previewing—An input preview window has been opened for this encoder.

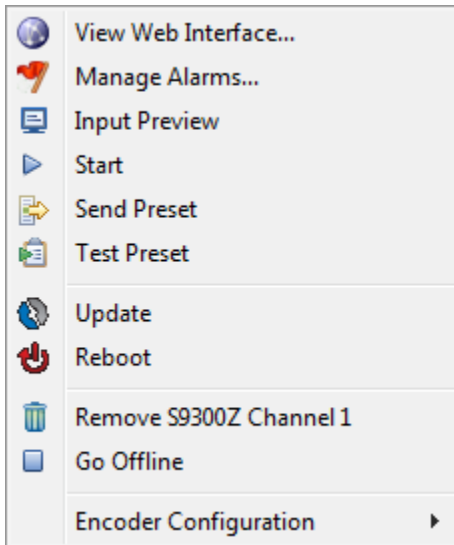
- Alarm—Displays an alarm warning  icon if any alarms are active on the encoder node. Click the alarm flag to display the encoder alarms. To ignore any specific alarms if they are not critical, check the **Ignore** check box for that alarm, and then click **Apply**. For recommended actions for alarms, see “Encoder Alarms” on page 276.
- Version—Software version of the encoder node.
- Model—Model number of the encoder node.
- Input(s)—Resolution and Frame rate of the detected input video. If this field has dashes, then this source has not been detected by the encoder.
- Preset—Current preset of the encoder, if currently encoding.
- View Encoding Stream—If available for current stream type, click the link to view the current encoding stream.
- Web Interface—Opens the AnyRes Live web interface.

Encoder Configuration

Right-click the encoder entry in the Encoders monitor to display a context menu with available configuration options.

Available/Scheduled/Stopped

If the encoder is Available, Scheduled, or Stopped, you will see the following menu.



Use this menu to perform the following actions related to the encoder:

View Web Interface—Displays the web interface for this encoder in a browser.

Manage Alarms—Displays the alarms for this encoder. You can optionally set specific alarms to be ignored.

Input Preview—For SDI encoders, displays the input video for this encoder. Input preview will create a temporary active event that will be removed after the preview window is closed.

Start—Opens the Start dialog box for the encoder. Choose a preset and click **Start Encoder** to create a streaming event to start now, with an indefinite end time.

Send Preset—Opens the Send Preset dialog box for the encoder. Choose a preset and click **Send Preset** to send it to this encoder. Note that this action does not make the preset the active preset on the encoder, it just sends the preset to the encoder.

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Test Preset—Opens a Test Preset dialog box for the encoder. Choose a preset to test on this encoder, and then click **GO** to begin the test. If successful, and if the player supports the output format, output video will be displayed for verification purposes only. Click **End Test** to stop the preset test.

Update—Opens an Update dialog box for the encoder. The current Encoder Update file is displayed. If this file is the correct update file, click **Update**, or click **Cancel** and choose **File > Import > Encoder Update File** to import a new update file, and then return to this option.

Reboot—Opens a Reboot dialog box for the encoder. Click **OK** in the dialog box to start a reboot of this encoder.

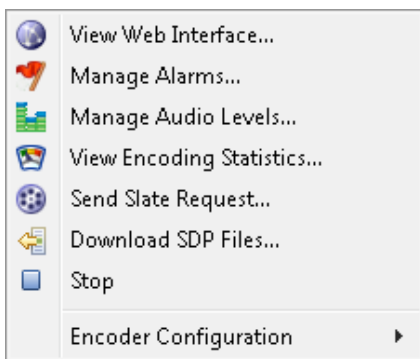
Remove—Removes this encoder from the LSM.

Go Offline—Makes this encoder unavailable to the LSM for encoding.

Encoder Configuration—Configures SNMP settings for this encoder. This will allow SNMP traps to be sent from the encoder node.

Encoding

If the encoder state is Encoding, you will see the following menu.



Use this menu to perform the following actions related to the encoder:

View Web Interface—Displays the web interface for this encoder in a browser.

Manage Alarms—Displays the alarms for this encoder. You can optionally set specific alarms to be ignored.

Manage Audio Levels—For SDI encoders only, displays a dialog box displaying audio channels. At the top of the dialog box the current decibels of each channel are displayed in real time. Use the slider controls beneath each channel decibel display to add/subtract the relative amount of decibels from what is currently being output. For example, if channel 1 is displaying 50 dB, and channel 1 is set to 10, the overall dB level should increase to 60 dB. The Manage Audio Levels option is not available for IP Input AnyRes Live models.

View Encoding Statistics—Displays the encoding statistics for this encoder.

Send Slate Request—See “Slate Request” on page 240 for details on sending a slate request to replace video with an image file at a specified time.

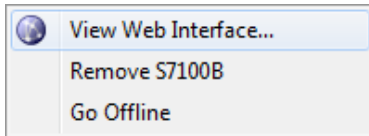
Download SDP Files—For RTP/3GPP Output, use this option to choose the location to download the primary (and optional redundant) SDP files that can be used by client software to view the stream or streams.

Stop—Stops the active event. The event will be listed in the completed events monitor and details of the event will be retained.

Encoder Configuration—Configures SNMP settings for this encoder. This will allow SNMP traps to be sent.

Updating/Searching/Connecting/Failed

If the encoder is Updating, Searching, Connecting, or Failed, you will see the following menu.



Use this menu to perform the following actions related to the encoder:

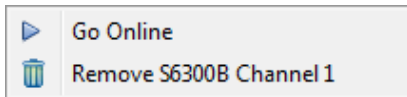
View Web Interface—Displays the web interface for this encoder in a browser.

Remove—Removes this encoder from the LSM.

Go Offline—Makes this encoder unavailable to the LSM for encoding.

Offline

If the encoder is Offline, you will see the following menu.



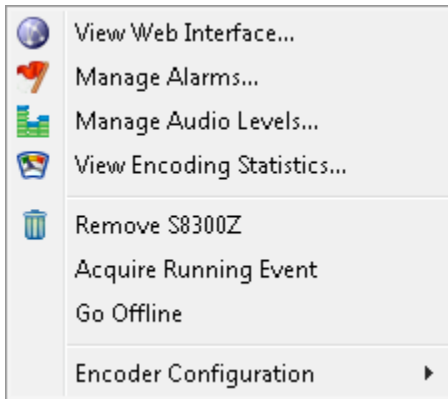
Use this menu to perform the following actions related to the encoder:

Go Online—Makes this encoder available to the LSM for encoding.

Remove—Removes this encoder from the LSM.

Pending

If the encoder is Pending, you will see the following menu.



Use this menu to perform the following actions related to the encoder:

View Web Interface—Displays the Web interface for this encoder in a browser.

Manage Alarms—Displays the alarms for this encoder. You can optionally set specific alarms to be ignored.

Manage Audio Levels—For SDI encoders only, displays a dialog box displaying audio channels. At the top of the dialog box, the current decibels of each channel are displayed in real time. Use the slider controls beneath each channel decibel display to add/subtract the relative amount of decibels from what is currently being output. For example, if channel 1 is displaying 50 dB, and channel 1 is set to 10, the overall dB level should increase to 60 dB. The Manage Audio Levels option is not available for IP Input AnyRes Live models.

View Encoding Statistics—Displays the encoding statistics for this encoder.

Remove—Removes this encoder from the LSM.

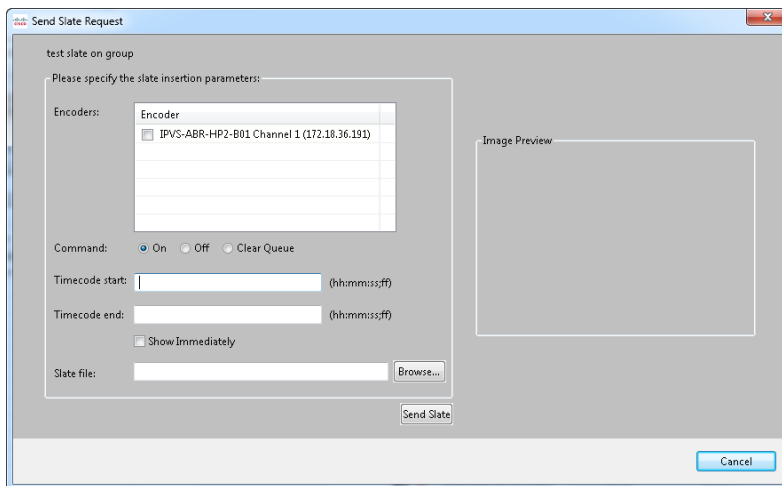
Acquire Running event—Imports the preset currently applied on the encoder, and creates an active event for the running encoder, unless a failed event matches the acquired event. In the case of a match to a failed event, the LSM recovers the event by moving it from the failed list to the active list, displaying any associated Channel as Running, and changing the encoder state to Encoding. If the acquired event does not match any failed events, the imported preset will be added to the Acquired Presets encoder preset group, with “acquired” appended to the preset name. The active event will have an indefinite end time unless a corresponding scheduled event exists for the event name. No changes are made to the status of the encoder.

Go Offline—Makes this encoder unavailable to the LSM for encoding.

Encoder Configuration—Configures SNMP settings for this encoder. This will allow SNMP traps to be sent.

Slate Request

Choose **Slate Request** from the context menu of an active event or an encoder that is encoding. For an active event, this option displays the Send Slate Request dialog box.



To replace input video with a slate file image on all output streams of any or all encoders associated with an event, choose one or more encoders, choose the On command, and provide the path or browse to the slate file. The Slate File field must include the path to the existing image file either on the encoder (recommended) or a network share accessible by the encoder. The slate file may be 24-bit BMP, 24-bit JPEG, .png, .tif, or .gif. Once the slate file is specified, the image will appear in the Image Preview.

To insert the file immediately, check the **Show Immediately** check box. If another slate is active, it will be replaced by the new slate.

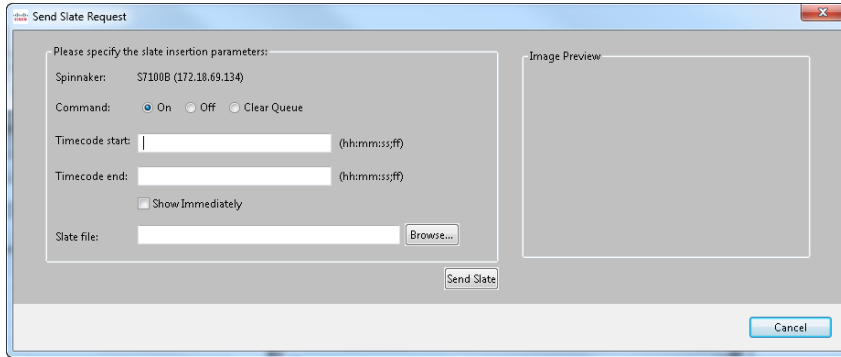
If provided, Timecode must be in 24-hour hh:mm:ss;ff format. The frame number (;ff) is optional. Once all fields in use have been specified, click **Send Slate** to queue the slate. If both start and end timecodes are specified, two separate commands will be generated; the On command will insert the slate at the specified time, and an Off command will be queued for the end time.

Choose the Off command to turn off the slate and restore video. To remove the file immediately, check the **Remove Immediately** check box. To remove at a specified time, provide the timecode end field. Once all fields in use have been specified, click **Send Slate** to remove the slate.


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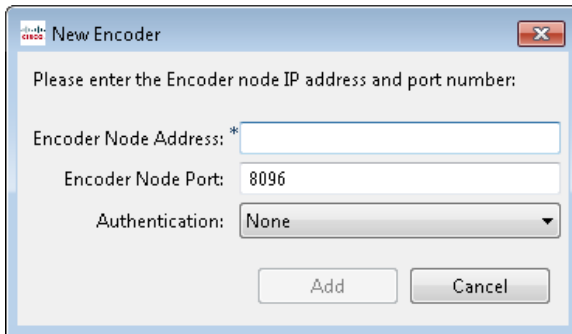
Choose the Clear Queue command to clear all slate requests from the queue. To clear the queue immediately, check the **Cancel Immediately** check box. To clear the queue at a specified time, provide the time and click **Send Slate** to delete all scheduled slate commands.

The Slate Request option for an encoder node will display the Send Slate Request dialog box.



Add Encoder Node

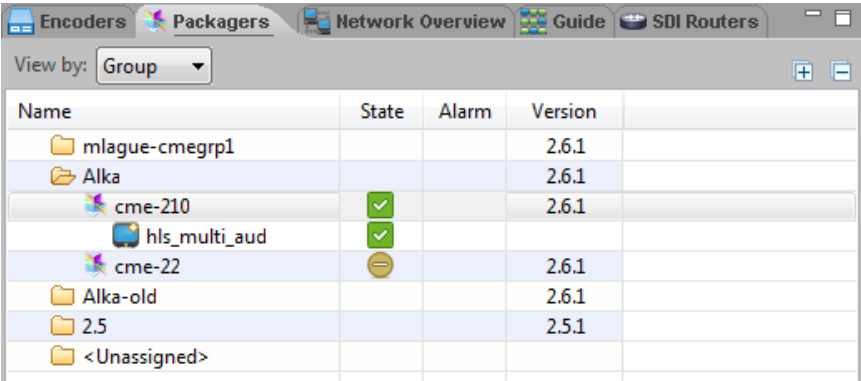
Click the **New Encoder**  icon to manually add a node to the system.



1. Enter the IP address for the node.
2. Enter the network port for the node. The default port number for an encoder node is 8096.
3. From the Authentication drop-down list, choose **None** for no authentication, choose **Username** for the encoder username and password, or choose **Windows** authentication for an encoder inside the domain.

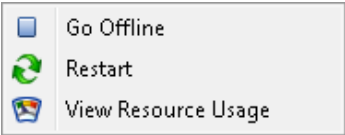
Packagers

The Packagers tab lists the Packager groups and packagers, as well as information specific to each packager. You can view by packager group, or view by a list of all packagers.



Name	State	Alarm	Version
mlague-cmegrp1			2.6.1
Alka			2.6.1
cme-210	✓		2.6.1
hls_multi_aud	✓		
cme-22	⊖		2.6.1
Alka-old			2.6.1
2.5			2.5.1
<Unassigned>			

For each packager, right-click the corresponding row to display a context menu of available options.

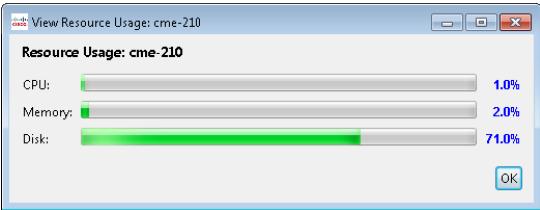


Use this menu to perform the following actions related to the packager:

Go Offline—Changes the packager state to Offline. If any Channels are currently assigned to this packager, they will be relocated to an available packager if possible.

Restart—Restarts the packager. This action will restart the Packager service that is running on its host machine. If any Channels are currently assigned to this packager, they will be relocated to an available packager if possible.

View Resource Usage—Displays the View Resource Usage dialog box.



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CPU—The CPU usage displays the percentage of CPU capacity in use on the server on which the packager is running. This usage is a composite of all CPUs/cores on the machine.


Memory—The Memory Usage displays the percentage of memory in use on the server on which the packager is running.

Disk—The Disk Usage displays the percentage of disk space in use on the server on which the packager is running.

The following information is shown for each Packager tab entry:

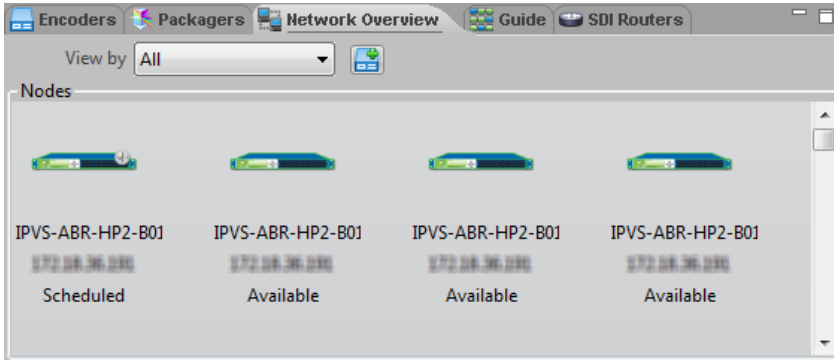
- **Packager Group name, Packager name, or Channel name**—Name of the group, packager, or running Channel on the packager.
- **State**—Current state of the packager group, packager, or Channel. Hover your mouse cursor over the icon to see the specific state name. The following table lists the possible states.

State	State Icon	Description
Available		The packager is online and available
No Heartbeat, or Unsupported		No Heartbeat indicates that the packager has been turned online by the user, but the LSM cannot communicate with it (the packager is not responding to heartbeat message requests). Unsupported indicates that the packager software version is unsupported in the group.
Degraded		For a Channel, Running with Alarms, or Unknown.
CAP versions vary		For a packager group, software versions of the packagers within the group are different. Packagers with versions other than the group version are not supported.
Offline		The packager has been turned offline by the user

- **Alarm**—Displays an alarm warning  icon if any alarms are active on the packager. See the context menu on the Channels tab to view alarms. For recommended actions for alarms, see “Appendix E: Troubleshooting” on page 274.
- **Version**—Software version of the packager.

Network Overview

The Network Overview tab provides a pictorial view of the encoder nodes and their state. Use the View by drop-down lists to restrict the list of nodes. To add an encoder node, click the **New Encoder** icon. For more information on adding the node, see “Add Encoder Node” on page 241.



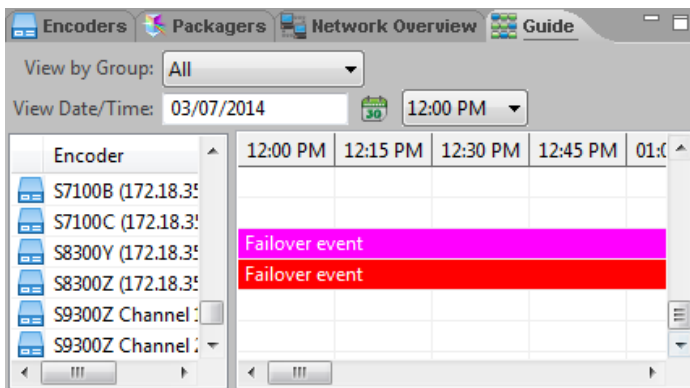
For each encoder node, the network overview displays a picture of an encoder in a color that represents its state, sometimes with an additional symbol that indicates that the encoder is encoding. In general, the green images represent an encoder that is working and communication is good, while the red images indicate that communication cannot be established. The encoder color and symbol indicate the state of the node as follows:

- **Green:** The encoder is connected and available to be assigned events by the LSM Server Node. States: Available, Stopped, Pending.
- **Green with green encoding symbol:** The encoder is encoding an event. State: Encoding.
- **Green with gray encoding symbol:** The encoder is scheduled to begin encoding an event within the next 30 minutes. State: Scheduled.
- **Yellow:** The node is “Searching”, “Connecting” or “Updating”, which means either the LSM Server Node has not yet completed establishing communication with it, it is updating or rebooting, or the status is unknown. States: Searching, Connecting, Updating.
- **Red:** The encoder node has failed, and the server cannot communicate with it. State: Failed. For information on what could resolve the failed state, see “Appendix E: Troubleshooting” on page 274.
- **Gray:** The encoder is connected to the server, but has manually been taken offline and cannot be used until it is manually placed back online. State: Offline.

Right-click the node icon to display the appropriate encoder configuration context menu for this encoder node. For information on the context menu options for each encoder state, see “Encoder Configuration” on page 236.

Guide

The Guide tab provides a pictorial view of the encoder schedules. Primarily, the Guide tab is useful for overview of the availability of encoders. If you know that you have an event at a particular time, use this tab to see which encoder is free at that time. Use the View by Group drop-down list to restrict the list of encoders by encoder group.



Right-click the encoder entry in the Encoder table to display a context menu with available configuration options. For more information about these options, see “Encoder Configuration” on page 236.

The following colors represent the event status of scheduled and active events:

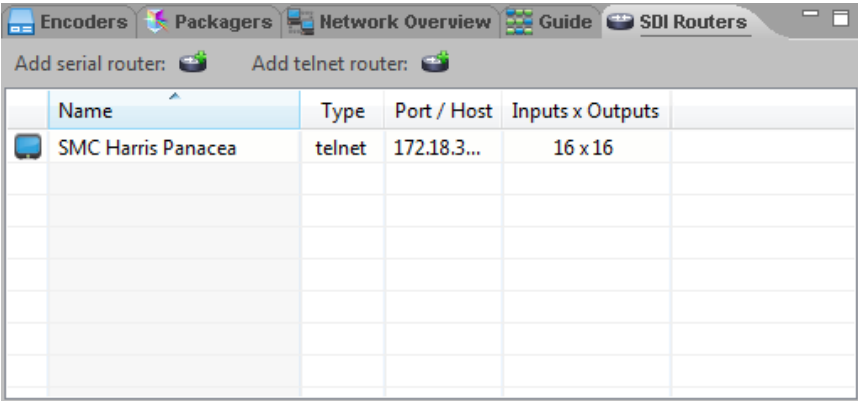
- Green: The event is active and is successfully encoding.
- Red: The event is active but has failed.
- Blue: The event is scheduled.
- Magenta: The encoder is a backup for an active event.
- Cyan: The encoder is a backup for a scheduled event.

Choose an unscheduled period of time for an encoder on the Guide tab and right-click to create an update, reboot, or streaming event for the specified time. Right-click a scheduled event time to display the streaming event context menu. For more information on the options provided by this context menu, see page 227.

SDI Routers

If more than one SDI Router is attached to the LSM Server, we highly recommend that the level settings on the SDI Routers differ, especially when added via Telnet or through a Harris Edge Router Protocol Translator. For information about how to configure the level of your SDI router, see the user documentation for your router.

The SDI Routers tab lists the SDI routers and information specific to each.



Name	Type	Port / Host	Inputs x Outputs
SMC Harris Panacea	telnet	172.18.3...	16 x 16

The following information is shown for each router:

- Router name—Name of the SDI router.
- Type—Serial or Telnet.
- Port/Host—Serial port to which a serial router is connected, or host of a Telnet router.

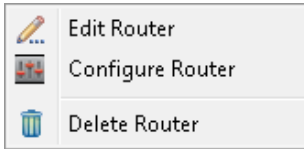
Note: A serial router must be attached via serial port to the machine that is running the LSM Server.

- Inputs x Outputs—Number of inputs and outputs for the router.

Router Configuration

Right-click the router entry in the SDI Routers monitor to display a context menu with available configuration options.

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Use this menu to perform the following actions related to the router:

Edit Router—Modifies SDI Input source names and associates encoders with router outputs.

Configure Router—Maps the SDI inputs of the router to output encoders.

Delete Router—Removes this router from the LSM.

Edit Router

Choose **Edit Router** from the context menu of a router, which will display the Edit Router dialog box:

Edit Router - SMC Harris Panacea

Router

Name: SMC Harris Panacea

Host/IP Address: 172.18.35.101

IP Port: 23

Username: leitch

Password: *****

Level: 0

SDI Inputs:

Name	Address
1920x1080	0
720p60 + VITC	1
480i	2
N/A	3
N/A	4
N/A	5

Directions:

Click the entry in the name column in the SDI Inputs table to modify the SDI input port names.

Please select a valid Cisco Media Processor to associate with each SDI output port in the SDI Outputs table by clicking the Encoder

SDI Outputs:

Name	Address	Encoder
S7100B (172.18.35....	0	S7100B (172.18.35.134)
Output 2	1	
172.18.35.59 (172.1...	2	S8300Z (172.18.35.59)
172.18.35.64 (172.1...	3	S8300Y (172.18.35.64)
S9300Z Channel 1 ...	4	S9300Z Channel 1 (172.18.35.69)
S9300Z Channel 2 ...	5	S9300Z Channel 2 (172.18.35.69)

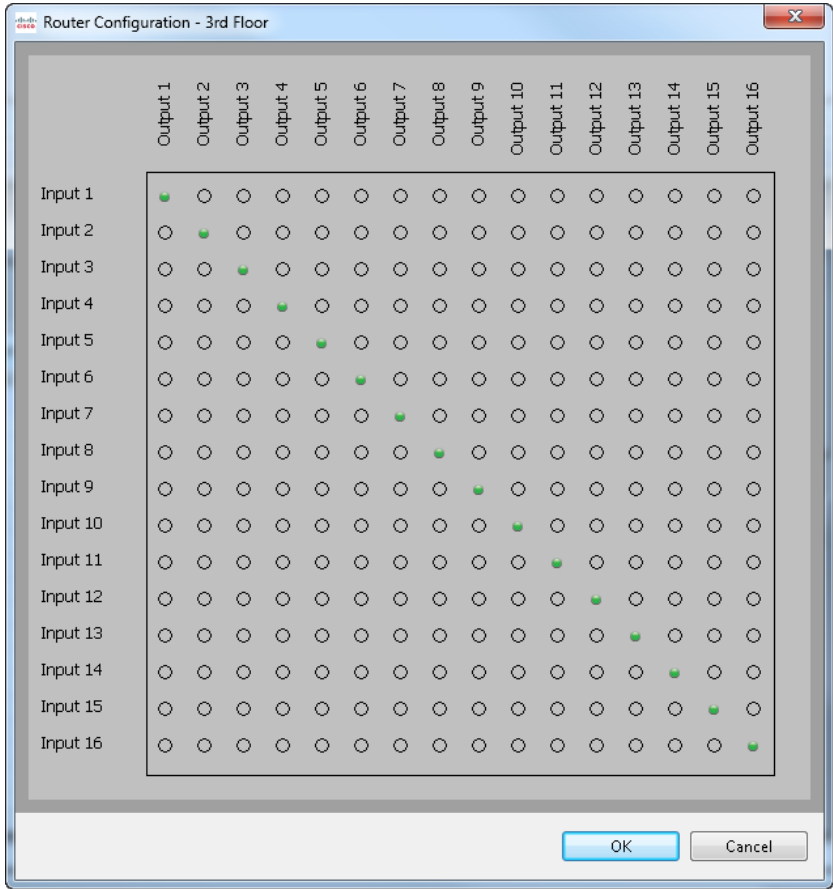
Update Cancel

For a serial router, view the serial port, baud rate, data bits, parity, stop bits, and flow control settings. For a Telnet router, view the host/IP address, IP Port, username, password, and level. To modify these settings, you must delete the

router and then add the router with the new settings. For the SDI Inputs, modify the SDI input port names to identify each source input by clicking the default input name and entering the new input name. For the SDI Outputs, associate an encoder with an output by clicking the entry in the Encoder column that corresponds to the output, and then choosing an encoder from the drop-down list. A particular encoder can only be associated with one output.

Configure Router

To configure a router, choose Configure Router from the context menu of the router to display the Router Configuration dialog box.



For each output, choose an input to map to that output by clicking on the circle at the intersection of the two. A particular output may only be mapped to one input, but the same input may be mapped to multiple outputs. The green dot represents

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that the input on that row has been mapped to the output on that column. Click **OK** when all outputs have been mapped.


Add a Serial Router

The LSM currently only supports the Harris Panacea line of serial routers. Additional routers may be added in future releases of the product.

To add your Harris Panacea router, follow the steps below:

1. Close any other terminal application tools on the LSM Server, such as Hyper terminal.
2. Turn off the power to the router by unplugging the power cable.
3. Verify that the Baud rate is set to 115200 on the SDI router (unscrew the front panel, and set the DIP switches 7 and 8 on SW3 to the up position).



4. Open services on the LSM system and restart the AnyRes Live Server Node service.
5. Turn on the power to the router by plugging in the power cable.
6. Log in to the LSM, and click the **Add serial router**  icon to manually add the router to the system.

New Router

Router

Name:

Serial Port:

Baud Rate:

Data Bits:

Parity:

Stop Bits:

Flow Control:

Note: this product only supports a baud rate of 115200 bps. Please refer to the 'SDI Routers - Add a Serial Router' section of the Cisco Linear Stream Manager User Guide for instructions on how to modify the baud rate of your SDI router.

Add Cancel


7. Enter the name for the router.
8. Choose the serial port to which the router is connected.

Note: The router must be attached via serial port to the machine that is running the LSM Server.

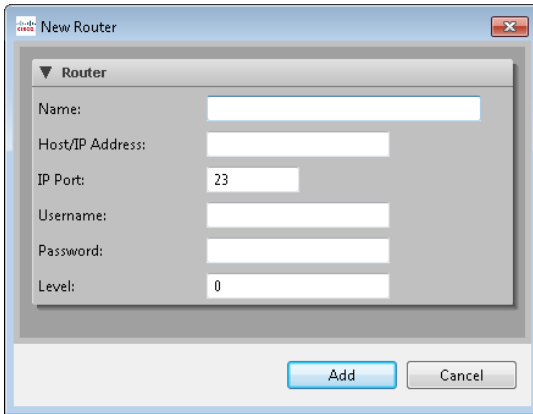
9. Choose the router baud rate from the drop-down list. Currently, only a baud rate of 115200 bps is supported. The baud rate set on the router must match this value or the LSM will not be able to communicate with the router.
10. Choose the data bits for the router from the drop-down list. The range is 5 to 8, and we recommend a value of 8.
11. Choose the parity bit for the router. Valid values are None, Odd, Even, Mark, and Space, and we recommend a value of None.
12. Choose the stop bits for the router. Valid values are 0, 1, 1.5, and 2. We recommend a value of 1.
13. Choose the flow control for the router. We recommend that you choose Request to Send.
14. Click **OK**.

The Router should be added to the list of available routers.

Add a Telnet Router

1. Click the **Add telnet router**  icon to manually add the router to the system.

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2. Enter the name for the router.
3. Enter the Host/IP address for the router. If the SDI router is attached to a Harris Edge Protocol Translator, enter the IP address of the Edge Protocol Translator.
4. Enter the IP Port for the router. The default is the standard Telnet port. If the SDI router is attached to a Harris Edge Protocol Translator, enter the port for the Edge Protocol Translator.
5. Enter the Username and Password. If the SDI router is attached to a Harris Edge Protocol Translator, enter the login credentials of the Harris Edge Protocol Translator.
6. Enter the Level of the router. The Level is configured at the router. Default configuration is 0. The value entered here must match the configuration at the router. If you are using a Harris Edge Protocol Translator with multiple SDI routers, each SDI router must be configured with a unique Level. See the SDI router user manual for information about how to configure the level.
7. Click **OK**.

The Router should be added to the list of available routers.

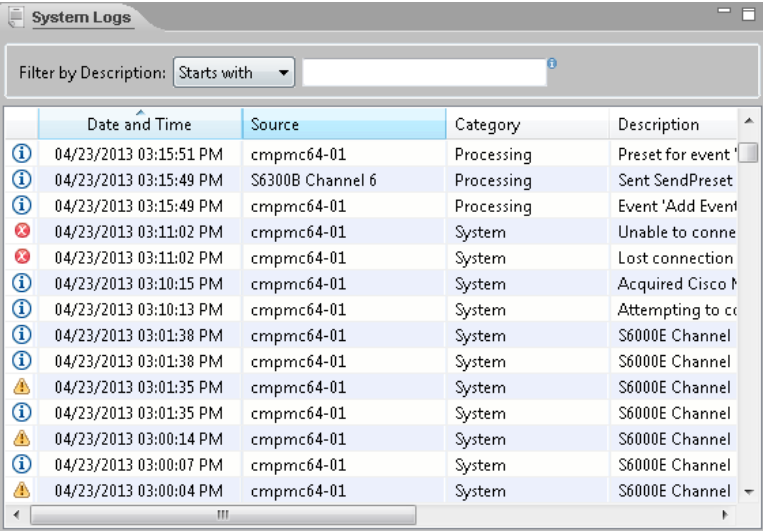
System Logs

The System Logs tab displays information, warning, and error events for encoders and packagers managed by the LSM.

The Filter by Description field provides several filtering options. From the drop-down list, choose Starts with or Ends with to restrict the display of logged events to those that begin or end with a particular character string. Choose Contains to display events that

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contain the entered character string. Choose Has Pattern to use * and ? wildcards with the string to display events that match the entered pattern.



	Date and Time	Source	Category	Description
i	04/23/2013 03:15:51 PM	cmpmc64-01	Processing	Preset for event
i	04/23/2013 03:15:49 PM	S6300B Channel 6	Processing	Sent SendPreset
i	04/23/2013 03:15:49 PM	cmpmc64-01	Processing	Event 'Add Event'
x	04/23/2013 03:11:02 PM	cmpmc64-01	System	Unable to connect
x	04/23/2013 03:11:02 PM	cmpmc64-01	System	Lost connection
i	04/23/2013 03:10:15 PM	cmpmc64-01	System	Acquired Cisco M
i	04/23/2013 03:10:13 PM	cmpmc64-01	System	Attempting to connect
i	04/23/2013 03:01:38 PM	cmpmc64-01	System	S6000E Channel
i	04/23/2013 03:01:38 PM	cmpmc64-01	System	S6000E Channel
w	04/23/2013 03:01:35 PM	cmpmc64-01	System	S6000E Channel
i	04/23/2013 03:01:35 PM	cmpmc64-01	System	S6000E Channel
w	04/23/2013 03:00:14 PM	cmpmc64-01	System	S6000E Channel
i	04/23/2013 03:00:07 PM	cmpmc64-01	System	S6000E Channel
w	04/23/2013 03:00:04 PM	cmpmc64-01	System	S6000E Channel

The following information is shown for each event:

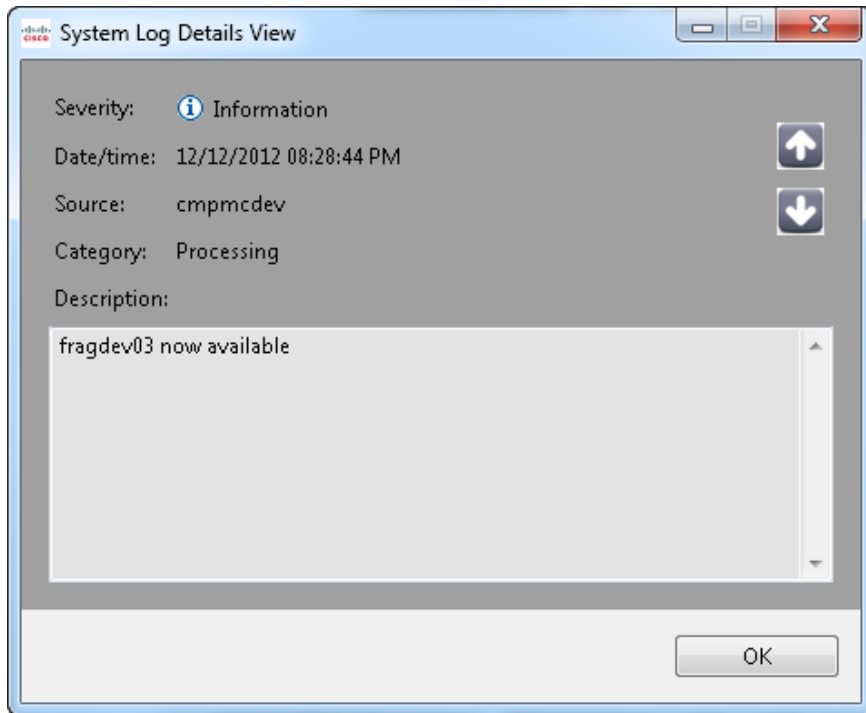
- Icon—Icon that shows the severity level of the events:
 - i Informational—Just information, no action needed
 - w Warning—Potential issue, may require user attention
 - x Error—Likely a problem, requires user attention
- Date and time—Date and time the event happened
- Source—Originator of the event (server, node)
- Category—System or Application
 - System—Anything occurring at a “system” level, such as services started or stopped, nodes detected or lost, or communication issues
 - Processing—Anything having to do with the operation of the LSM, the LSM Server Node, or the AnyRes Live node programs. (that is, event started, completed, failed, and so forth)
- Description—High level description of the event

To communicate system log data to support personnel, select one or more rows in the system log, right-click to display a context menu, and choose **Copy** to copy the log data

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to the system clipboard. This log data can then be pasted in an email message or spreadsheet.

Click the severity icon for the logged event to display the event details. An example of a system log details dialog box follows.



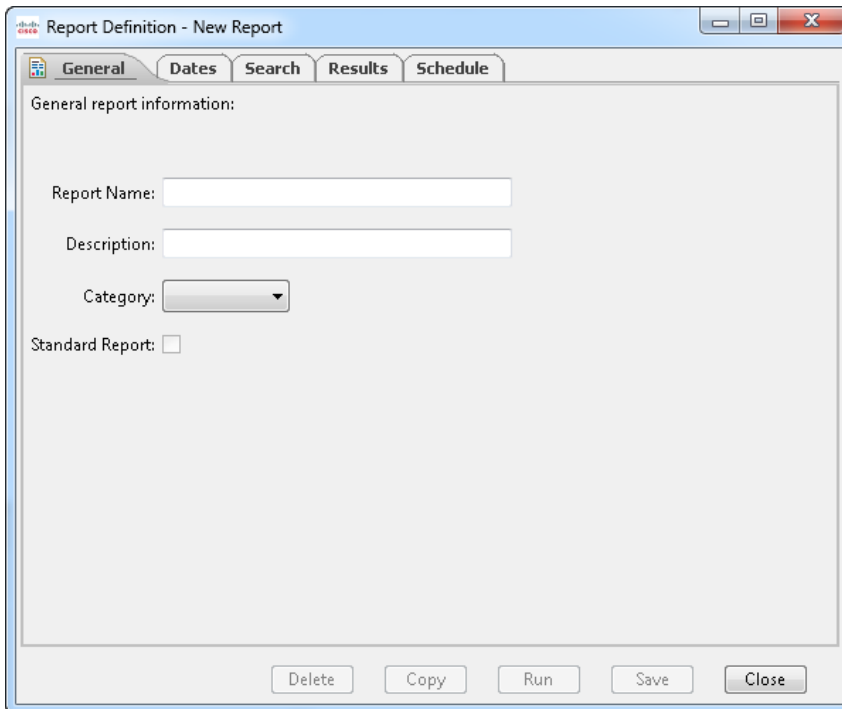
Overview

Reports 8

Reports can be run on data related to encoder nodes, events, and logs. A report can be run to view immediately, saved, or scheduled to generate a file that can be emailed or stored in a network location.

Create a Report

To add a new report, choose the **New Report**  icon from the Reports tab, which will open the Report Definition dialog box.



The image shows a screenshot of the 'Report Definition - New Report' dialog box. The dialog has a title bar with the Cisco logo and the text 'Report Definition - New Report'. Below the title bar are five tabs: 'General', 'Dates', 'Search', 'Results', and 'Schedule'. The 'General' tab is selected. The main area of the dialog is labeled 'General report information:' and contains the following fields:

- Report Name:** A text input field.
- Description:** A text input field.
- Category:** A dropdown menu.
- Standard Report:** A checkbox.

At the bottom of the dialog, there are five buttons: 'Delete', 'Copy', 'Run', 'Save', and 'Close'.

General Tab

On the General tab, enter the Report Name, optional Description, and Category. Report categories include Events, Encoders, and Logs. Choices on other tabs will be determined by the report category.

Dates Tab

Report Definition - New Report

General Dates Search Results Schedule

Report date and time range information:

Date and time field: Start Time

Predefined

☐ Last 24 hours

Custom

☒ Start: 6:02:21 PM End: 7:02:21 PM

May, 2013							May, 2013						
Sun	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Mon	Tue	Wed	Thu	Fri	Sat
28	29	30	1	2	3	4	28	29	30	1	2	3	4
5	6	7	8	9	10	11	5	6	7	8	9	10	11
12	13	14	15	16	17	18	12	13	14	15	16	17	18
19	20	21	22	23	24	25	19	20	21	22	23	24	25
26	27	28	29	30	31	1	26	27	28	29	30	31	1
2	3	4	5	6	7	8	2	3	4	5	6	7	8

Delete Copy Run Save Close

The Dates tab allows report results to be restricted to those that fall within a defined date/time range. The Dates tab is only relevant if the report category is Events.

For the Events category, choose a predefined time range, use the date and time field for the report for the Start Time or End Time.

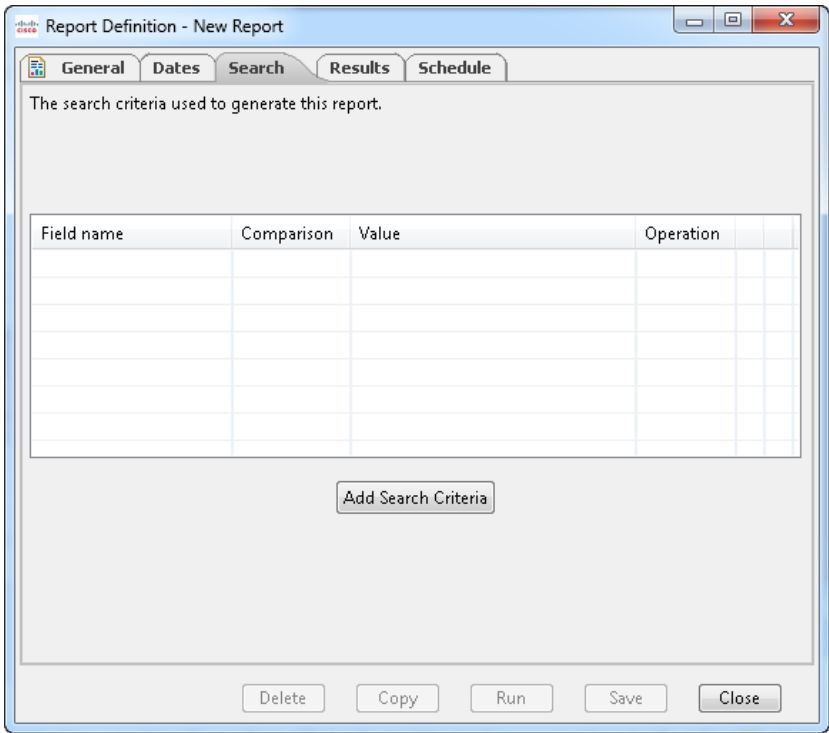
Predefined

Predefined date ranges for reports, such as Last 24 hours and Last 7 days, allow easy selection of common report date ranges.

Custom

Custom date ranges for reports allow specification of any range of dates and times for the report. Choose a start and end time by typing in the time box or using the arrows to adjust the time. To choose a start and end date, use the arrows to arrive at the correct month, and then click the desired date.

Search Tab



The Search tab specifies the search criteria that will be used to generate the report. Use search criteria to narrow the contents of the report to specific information, such as customer name in a jobs report. Click **Add Search Criteria** to open the New Report Search Criteria Entry dialog box.

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New Report Search Criteria entry

Please enter the Report Search Criteria details:

Field: Archive File

Comparison Operation: =

Value: Date

Logical Operation: AND

OK Cancel

The search field list from the drop-down list will consist of choices related to the report category. Choose a field, the comparison operation, the value to compare to that field, and the logical operation to be used between this field and other search criteria.

Only the comparison operation choices appropriate to the report category and field name chosen will be available in the search criteria. The following comparison operations are available for the search criteria.

Comparison operator	Description
=	Equal to
<>	Not equal to
>	Greater than
>=	Greater than or equal to
<	Less than
<=	Less than or equal to
Contains	The data contains the character string (case-insensitive) specified in the value

Results Tab

The screenshot shows a window titled "Report Definition - New Report" with a Cisco logo in the top-left corner. The window has five tabs: "General", "Dates", "Search", "Results" (which is selected), and "Schedule". Below the tabs, the text "Choose the data to display:" is followed by a table with three columns: "Field", "Sort", and "Summary".

Field	Sort	Summary
Assigned Encoder	Descending	
Event State		

At the bottom of the window, there are five buttons: "Delete", "Copy", "Run", "Save", and "Close".

The Results tab defines the fields that will be shown in the report. The report category determines the available fields.

Choose the fields in the order that you want the columns to appear. Choosing the first field causes another drop-down list to become available for the subsequent field. You can also designate one field to be sorted in either ascending or descending order. At least one field must be presented in non-summary format.

To remove a field from a report, choose the blank field from the drop-down list.

Schedule Tab

Report Definition - New Report

General Dates Search Results **Schedule**

Report schedule and delivery:

Time

Frequency: Weekly

Starting: 7:02:21 PM

May, 2013

Sun	Mon	Tue	Wed	Thu	Fri	Sat
28	29	30	1	2	3	4
5	6	7	8	9	10	11
12	13	14	15	16	17	18
19	20	21	22	23	24	25
26	27	28	29	30	31	1
2	3	4	5	6	7	8

Delivery

Path:

Email:

Format: CSV

Delete Copy Run Save Close

Frequency

To schedule automatic generation of reports, first specify the time parameters. Options for **Frequency** include the following:

- No schedule—This report will not be generated on a time basis, but can still be run on demand.
- Once—Run this report one time at the specified starting time.
- Hourly—Run this report hourly, beginning at the specified starting time.
- Daily—Run this report daily at the specified starting time.
- Weekly—Run this report weekly, at the specified starting time on the specified day of the week.
- Monthly—Run this report monthly, at the specified starting time on the specified day of the month. If the day of the month is greater than 28, and the current month has fewer days than specified, then run this report on the last day of the current month.

Delivery

In the **Delivery** area, enter a location where the report file will be stored, an email address to send the report file, or both. The report file name is the report name concatenated with a date string. The file extension is the data format (.csv or .xml) chosen in the **Format** drop-down list, which indicates either CSV (Comma Separated Variables) or XML format.

Storing a File

To store the report file when the scheduled report is run, enter or browse to the **Path** indicating where to store the report results.


Emailing a file

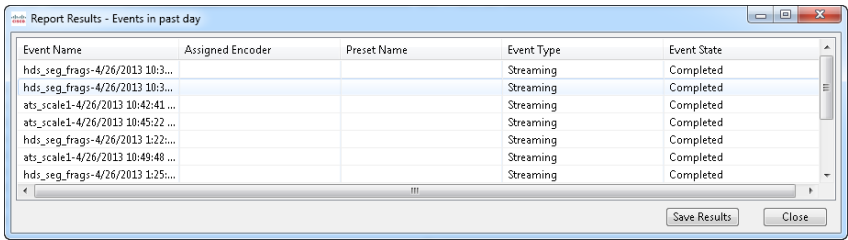
To email the report file when the scheduled report is run, enter the **Email address** indicating where to send the report results. The report name will be included in the email subject line, and the report will be an attachment.

Saving the New Report

Once report is defined, save the report by clicking **Save**. At a minimum, a new report must have a Report Name and Category in the General tab, and at least one field defined in the Results tab.

Run a Report


To run a report, click the **Run**  icon for the report definition, or open the report definition from the Reports tab and click **Run** at the bottom of the report definition. The report results will be displayed for viewing. The following is an example of a report output.

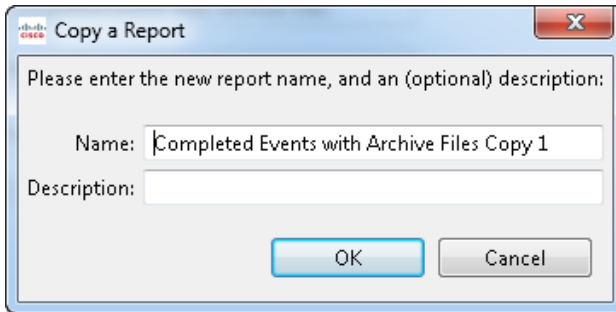


Event Name	Assigned Encoder	Preset Name	Event Type	Event State
hds_seg_frags-4/26/2013 10:3...			Streaming	Completed
hds_seg_frags-4/26/2013 10:3...			Streaming	Completed
ats_scale1-4/26/2013 10:42:41 ...			Streaming	Completed
ats_scale1-4/26/2013 10:45:22 ...			Streaming	Completed
hds_seg_frags-4/26/2013 1:22...			Streaming	Completed
ats_scale1-4/26/2013 10:49:48 ...			Streaming	Completed
hds_seg_frags-4/26/2013 1:25...			Streaming	Completed

You can also save the report results to a file by clicking **Save Results** and entering the location to save the file. The report file name is the report name concatenated with a date and time string, and the file extension is .csv (Comma Separated Variables).

Copy a Report

Any standard or custom report can be copied to become the initial definition of a new report. To copy a report, open the report definition from the Reports tab by clicking the **Edit**  icon. Click **Copy** at the bottom of the report definition to open the Copy a Report dialog box.



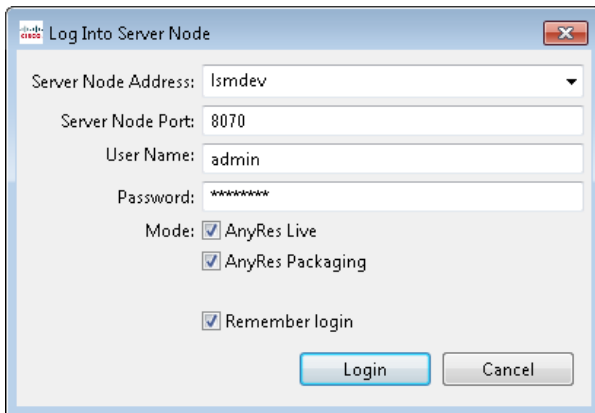
Enter the name of the copied report and an optional description. Click **OK** to create the copy. The new report name is listed on the Reports tab. Edit the new report to modify its definition, and then save it.

Standard Reports

Standard reports are provided by the Cisco Linear Stream Manager (LSM) for commonly useful report data. They may also be copied as a starting point for a custom report. Standard reports cannot be deleted, scheduled, or edited.

Logging In **User Administration** 9

From the User menu, choose **Log In** to manually log into the server.



The image shows a 'Log Into Server Node' dialog box with the following fields and options:

- Server Node Address: A drop-down menu showing 'lsmdev'.
- Server Node Port: A text field containing '8070'.
- User Name: A text field containing 'admin'.
- Password: A text field containing '*****'.
- Mode: Three checked checkboxes: 'AnyRes Live', 'AnyRes Packaging', and 'Remember login'.
- Buttons: 'Login' and 'Cancel'.

Enter the address and port for the Cisco Linear Stream Manager (LSM) Server Node, and your username and password on that server. If you use more than one server address, any address you have used can be chosen from the drop-down list of addresses.

The LSM ships with two default usernames, admin and user1. We recommend that you change the password immediately the first time you log in. See “Changing a Password” on page 263.

To log in with Administrator authority, use the following user name and password:

User name: admin

Password: password

To log in with User authority, use the following user name and password:

User name: user1

Password: password

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The Mode check boxes determine whether the LSM displays panes that apply to management of AnyRes Live encoders, AnyRes Packagers, or both. Check the **AnyRes Live** check box to only display encoder-related areas of the LSM, check the **AnyRes Packaging** check box to only display packager-related areas of the LSM, or check both check boxes to display all areas of the LSM.

Optionally check the **Remember Login** check box to display this username and password in these fields for subsequent logins. If you uncheck this check box on a subsequent login, the saved username and password are wiped out.

Click **Login** to log in to the LSM Server.

Logging Out

From the User Menu, choose Log Out to log off the LSM server.

Changing a Password

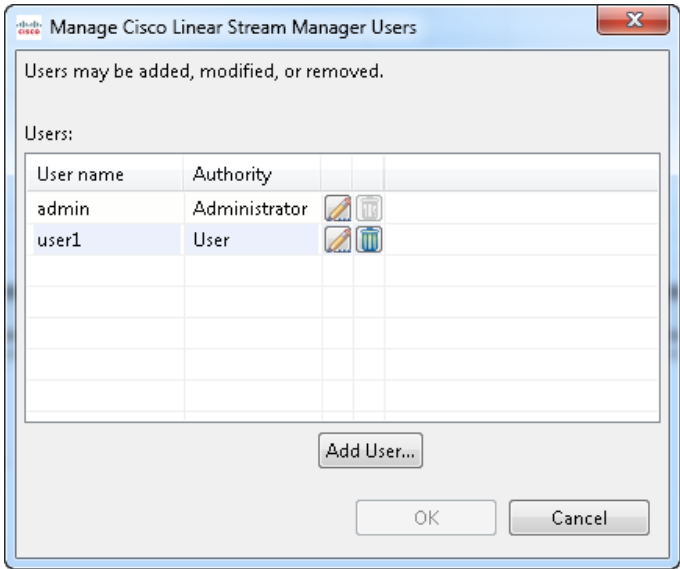
Once you have logged in, you can change the password for the user name by choosing Change Password from the User menu, which will display the Change Password dialog box.

A screenshot of a 'Change Password' dialog box. The dialog has a title bar with the Cisco logo and the text 'Change Password'. Inside, it says 'Please enter your new password:'. There are three text input fields: 'Current Password:', 'New Password:', and 'Confirm New Password:'. At the bottom, there are two buttons: 'OK' and 'Cancel'.

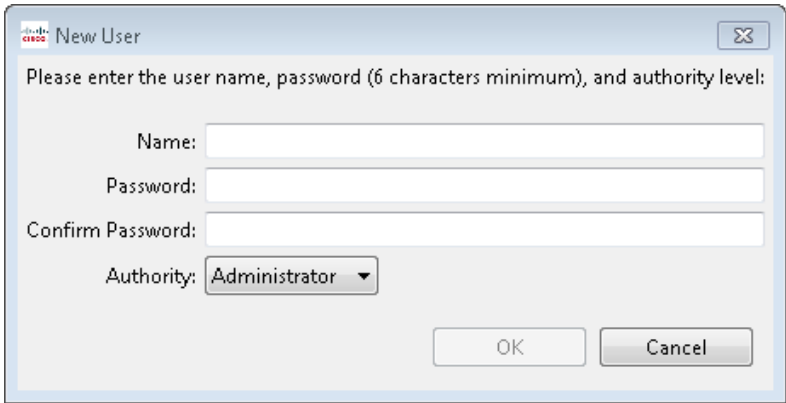
Enter the new password, confirm it, and click **OK** to change the password.

Manage Users

Choose Manage Users to add, modify, or remove users.



To add a user, click **Add User** to display the New User dialog box.



Setting Up AnyRes Live to Write to a Network Drive

Appendix A: Technical Guide

A

To enable AnyRes Live to archive or write a file to a network drive, the Encoding Service needs to have read and write capabilities on the network drives on your network.

To do this, you will need to log into AnyRes Live using remote desktop, and then follow these steps:

1. From the Windows Startup menu, right-click **My computer** and choose **Manage**.
2. In the left pane, expand **Services and Applications**, and then click **Services**.
3. Right-click **Encoding Service A** and choose **Properties**.
4. In the Properties window, click the **Log On** tab.
5. On the Log On tab, click the **This account** radio button and enter the username and password that has read and write permissions on your network drives. Click **OK**.
6. After changing the logon account for an AnyRes Live encoder, you should restart the service by closing the Properties window and clicking the **Restart** hyperlink in the Services applet.

After you complete the procedure to allow AnyRes Live to write to a network drive, you must also add the username to AnyRes Live as an administrator. Follow these steps:

1. From the Windows Startup menu, right-click **My computer** and choose **Manage** (or choose **Control Panel > Computer Management**).
2. In the left pane, expand **Local Users and Groups**, and then click **Groups**.
3. Right-click **Administrators** and choose **Add to Group**.
4. In the Administrators Properties dialog box, click **Add**.
5. In the Enter the Object Names to Select field, enter the same username that you specified for the logon account.
6. When prompted, enter the username and password.

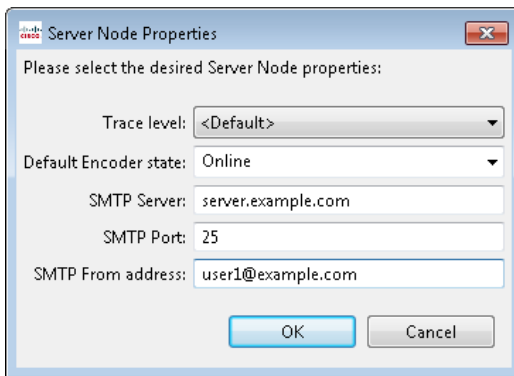
CISCO SYSTEMS, INC.

7. Click **OK**.

.

Appendix B: Server and Node Server Options Options **B**

Choose **Properties > Server Node** from the View menu to display the Server Node Properties dialog box.



View or modify the following Cisco Linear Stream Manager (LSM) Server Node properties:

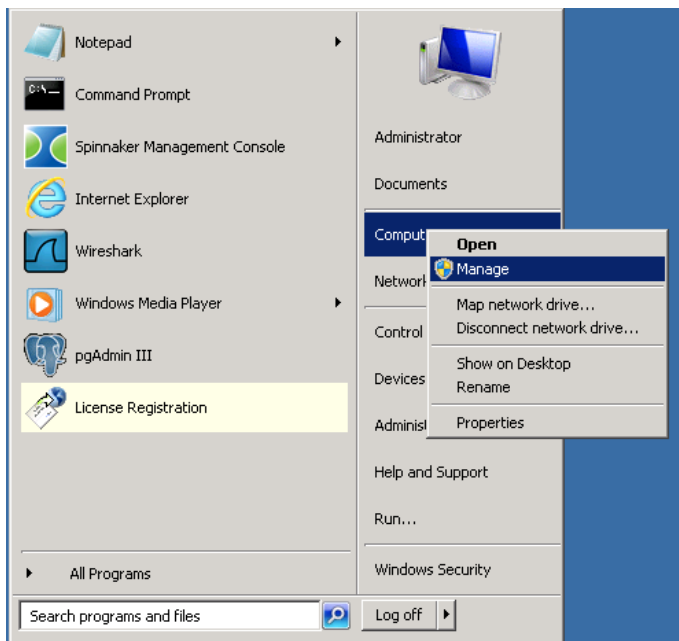
- **Trace level**—Choose Default, Off, Error, Warning, Informational, Debug, or All. Trace levels are inclusive, and will trace all higher levels. Default resets the LSM Server Node to the original trace level.
- **Default Encoder state**—Online or offline. This is the state an encoder node will be set to when first detected.
- **SMTP Server**—SMTP (Simple Mail Transfer Protocol) server name, either host name or dotted-decimal IP address.
- **SMTP Port**—SMTP port number, usually 25.
- **SMTP From Address**—the address to use in the From field when the server node sends automated email.

Node Options

If the encode and server nodes are a part of a Windows Domain, you may need to enable the Cisco Linear Stream Manager Server and Cisco AnyRes Live node services to log in to your network with a username and password that has read and write capabilities to the network drives on your network.

To do this, you will need to log into the server node or AnyRes Live node locally or using remote desktop, and then follow these steps:

1. From the Windows Startup menu, right-click **My computer**, and then choose **Manage** (or choose **Control Panel > Computer Management**).



2. In the left pane, expand **Services and Applications**.
3. Click **Services**.
4. For AnyRes Live Nodes, right-click **Spinnaker Messaging Service** and click **Properties**. For Server nodes, right-click **Inlet Spinnaker Server Node** and click **Properties**.
5. In the Properties window, click the **Log On** tab.

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6. On the Log On tab, click the **This account** radio button and enter the username and password that has read and write permissions on your network drives. Click **OK**.
7. Restart the service by closing the Properties window and clicking the **Restart** hyperlink in the Services applet.

Appendix C: LSM Database Backup and Restore

The Cisco Linear Stream Manager (LSM) Server will automatically back up the application database after the start of the LSM Server Node. Additional database backups will be taken every six hours. We recommend that you periodically transfer a database backup to a remote file location should it be needed for recovery purposes.

Database backups are located in the following directory:

\Program Files\Cisco Systems\LSM Server Node\backups (or \Program Files(x86)\Cisco Systems\LSM Server Node\backups)

The backups are created with the following name structure:

lsm_db_releaseNumber_yyyymmdd_hhmm.backup

In the backup name, yyyymmdd_hhmm is the current timestamp at start of backup. For example, a backup of release 7.2.0.40103 on September 9, 2013 at 3:15pm would be backed up to lsm_db_7.2.0.40103_20130909_1515.backup. The timestamp is created with a 24-hour clock, and uses local time.

The system, by default, will keep 7 days of backups. Therefore, by default, up to 28 of the most recent database backups (4 backups a day x 7 days) will be kept. Once the limit is reached, the system will start deleting backups, always deleting the oldest backup after the current backup completes successfully.

The automated backups are accomplished via the *pg_dump* utility bundled with Postgres, and can be restored either via the *restore* script located in the main application directory, or via the *pg_restore* or *pgAdmin* utilities provided with Postgres.

Database Restore

To restore a backup of the LSM database, use the Windows command script called *restore.cmd*, installed in the primary application directory (\Program Files\Cisco Systems\LSM Server Node or \Program Files(x86)\Cisco Systems\LSM Server Node) of an LSM Server Node installation. The script takes a single argument, the path to a Postgres backup file to be restored. Execute the script on the command line by following this example:

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```
restore.cmd .\backups\lsm_db_7.2.0.40103_20130909_1515.backup
```


This example illustrates the restoration of one of the automated backups (stored in the \backups directory in the main application directory). When invoked, the script will first ensure the environment it needs to function properly is set up, and then backs up the current LSM database by renaming it to spinnaker_YYYY_MM_DD_HHMM, where YYYY_MM_DD_HHMM is the current time to the nearest minute. The script then restores the specified backup file to the spinnaker database. If the backup file is corrupt or otherwise fails to restore, the script deletes the new (faulty) database and renames the backup database to spinnaker, the original database name.

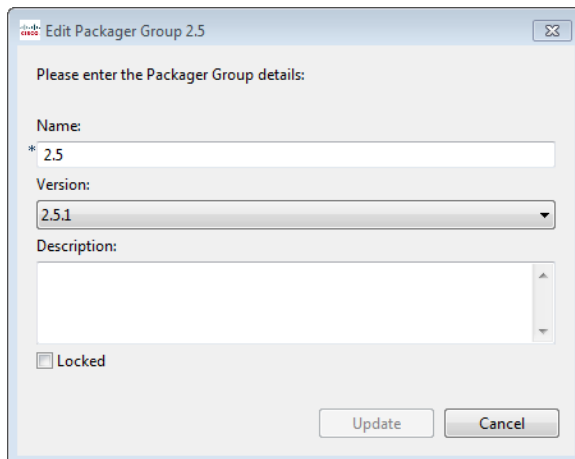
Appendix D: Upgrading Cisco AnyRes Packager Upgrade Procedure

To upgrade Cisco AnyRes Packager, follow these steps:

1. Upgrade both the Cisco Linear Stream Manager (LSM) Server and Client. See the *Cisco Linear Stream Manager Quick Start Guide* for additional details.
2. Upgrade each packager. See “AnyRes Packager Application Installation” in the *Cisco AnyRes Packager Installation and Configuration Guide* for additional details.

For a locked group, as each packager goes offline, the Channels on that packager will stop operating. Note that if Channel redundancy was enabled, the redundant instance on another packager will continue to operate; therefore, overall video service would still be available. After the packager is upgraded and reachable again by the LSM, the Channels will be sent back down to the packager and will become active. If the group is unlocked during upgrade and the group includes multiple packagers, Channels on the upgrading offline packager will be relocated to another online packager.

3. Edit the packager groups that include upgraded packagers. Click the **Edit**  icon next to the group name on the Packager Groups tab of the Configure tab to display the Edit Packager Group dialog box.



Edit Packager Group 2.5

Please enter the Packager Group details:

Name:
* 2.5

Version:
2.5.1

Description:

☐ Locked

Update Cancel

4. From the drop-down list, choose the release version for the packager group, and then click **Update**.

Note: If you are upgrading to LSM 9.0, the version of existing packager groups is unaffected. Packagers in this group may be upgraded to higher versions, but new features will not be enabled until the group is updated to the version that supports the new features. Before raising the group version to a higher software version, each packager in the group must be upgraded to that version or later.

Troubleshooting Scenarios

Appendix E: Troubleshooting

I am having a problem with my encoder node showing a “failed” state.

Check the following:

1. If the encoder was available previously, try taking it offline and then back online. This will cause the server to initiate communication again.
2. Clocks—The encoder and server must have a time that is within 5 minutes of each other. Otherwise, the lower level protocol will reject the messages. We recommend that the server and the encoder use the same network time protocol server.
3. DNS—The API between the encoder and server uses DNS authentication. To verify that the DNS name is correct, use a command prompt on the server to enter ping –a *encoder IP address*. The ping should respond with the DNS name of the encoder. Make sure that this name matches what the encoder is reporting as its DNS name.
4. Basic IP connection—Can they ping each other?
5. Firewall/router—Make sure that the firewalls on the encoder and the server, as well as routers in the network between them, have the correct port (8096) open.
6. Username—If you have changed the Spinnaker Message Service to log on as another user other than “admin”, then the configuration file on the server needs to be updated with the username and password that is being used.
7. Are both machines powered up? On the server in the services list, do you see that the Inlet Spinnaker Service is started? On the encoder, do you see that the Spinnaker Message Service is started?

The LSM cannot connect to the AnyRes Live server.

Check the following:


1. Basic IP connection—Can they ping each other?
2. Firewall/router—Make sure that the firewalls on the Cisco Linear Stream Manager (LSM) and the server, as well as routers in the network between them, have the correct port (8070) open.

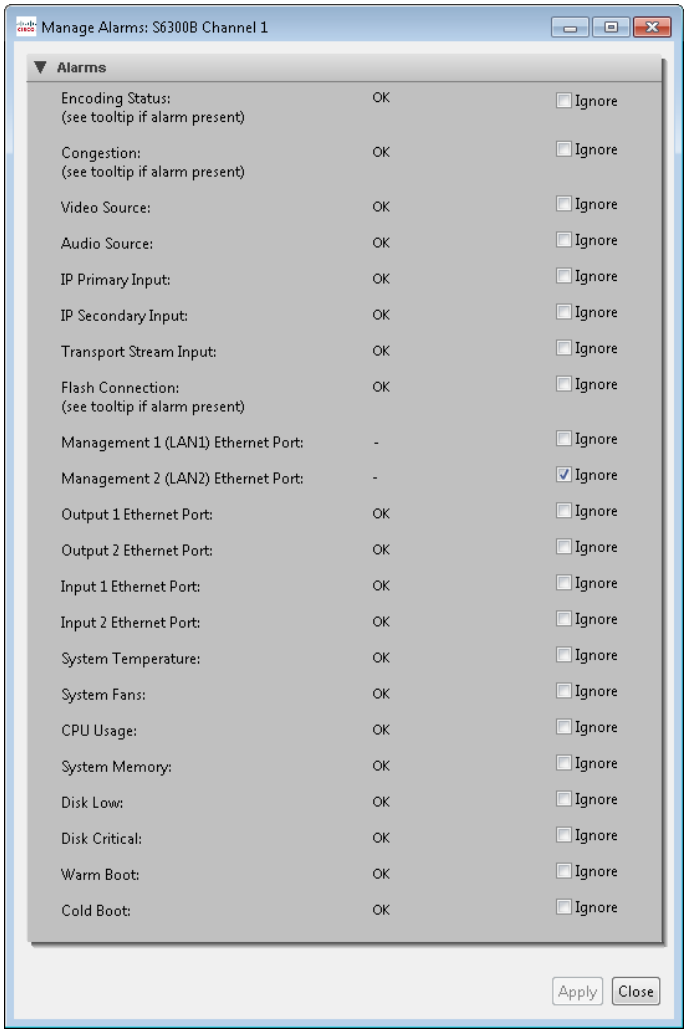
The server and/or encoder nodes do not seem to have access to network share drives.

With Windows domains the Inlet services (encoder node and server nodes) will need to “log on” to the domain with a username and password that has read and write privileges on the network. See “Appendix B: Server and Node Options” on page 267 for more information.

Note: Other troubleshooting information can be found in the Cisco AnyRes Live Family User Guide. To view the guide, open the AnyRes Live web interface and click Help.

Encoder Alarms

Click the Alarm flag warning  icon on the Encoders tab to display the encoder alarms.



The Manage Alarms dialog box displays the most recently updated status of alarms that are constantly being monitored. The system alarms are global alarms for the AnyRes Live encoder, but encode, audio and video alarms are independent for encoders with two channels. Also, if the video source is lost, audio will be lost as well. The following table details the alarm triggers and recommended actions.

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Alarm	Indicator	Recommended Action
Encoding Status	ALARM	An encoding error has occurred. Check the Message field to view the error message.
License	ALARM	Make sure that the HASP license is configured properly on the system.
Congestion	ALARM	Congestion has been detected and data is potentially being lost. Congestion could be either input congestion or output congestion. Generally, congestion means that either the CPUs, network bandwidth, or the network cannot keep up with the amount of input packets. Check the queue size encoding statistic for information that may help determine the source of the congestion.
Flash Connection	Status Message	The connection to the Flash Media Server has been lost during an encode with RTMP output enabled. Make sure that the Server URL and authentication credentials are entered correctly. If the connection still fails, the Flash Media Server may be down or unreachable. Connection retry alarms will be displayed based on the configured retry count and delay. Once all retries are exhausted, the retries exhausted alarm will be displayed.
Video Source	ALARM	Make sure that video is active on the source currently specified on the Input page and that the cable for that source is properly connected. Also, make sure that your video source equipment is powered on and working properly.
Audio Source	ALARM	Audio has been silent on the specified audio source for at least 15 seconds during encoding. Make sure that both audio and video are active on the source currently specified on the Input page and that the cable for that source is properly connected. For stereo analog audio, make sure that audio is active on both left and right. This alarm will return to OK after 1 second of audio is detected on the source during encoding.

Alarm	Indicator	Recommended Action
Dropped Frame	ALARM	The number of dropped frames has exceeded the threshold specified in the Alarm Options area of the Global Configurations page. The conditions causing the dropped frames may be temporary and no action may be required. If the alarm persists, we recommend reducing the complexity of encoding. Click the Clear Alarms link for this alarm to manually clear the Dropped Frame alarm. This alarm will not clear automatically.
IP Input (IP Source Encoders Only)	ALARM	No network packets are coming in from the multicast address and port specified on the Input page for this channel. Make sure that video is active on the source currently specified on the Input page and that the cable for that source is properly connected. Also, make sure that your video source equipment is powered on and working properly.
Transport Stream Input (IP Source Encoders Only)	ALARM	First Priority error conditions have been detected on the source input transport stream, as specified in the ETSI TR101290 specification. First Priority errors on the input will affect the result of the encoded output. The alarm will be cleared automatically when no input stream errors have been detected for approximately 10 seconds.
Ethernet Port	ALARM	Check to make sure that the cable is properly connected.
System Temperature	HIGH	Contact Service personnel.
System Fans	FAILURE	Contact Service personnel.
CPU Usage	HIGH	Change options to make the encoding simpler.
System Memory	LOW	Change options to make the encoding simpler.
Disk Low	LOW	Check if encodes are being saved to the local disk. Remove previously saved encoded files off the AnyRes Live encoder. This alarm is triggered if the available disk space is < 2GB.
Disk Critical	LOW	Check if encodes are being saved to the local disk. Remove previously saved encoded files off the AnyRes Live encoder. This alarm is triggered if the available disk space is < 1GB.

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Alarm	Indicator	Recommended Action
Warm Boot or Cold Boot	ACTIVE	A system reboot is in progress. This alarm will display OK 10 seconds after the encoding service starts.

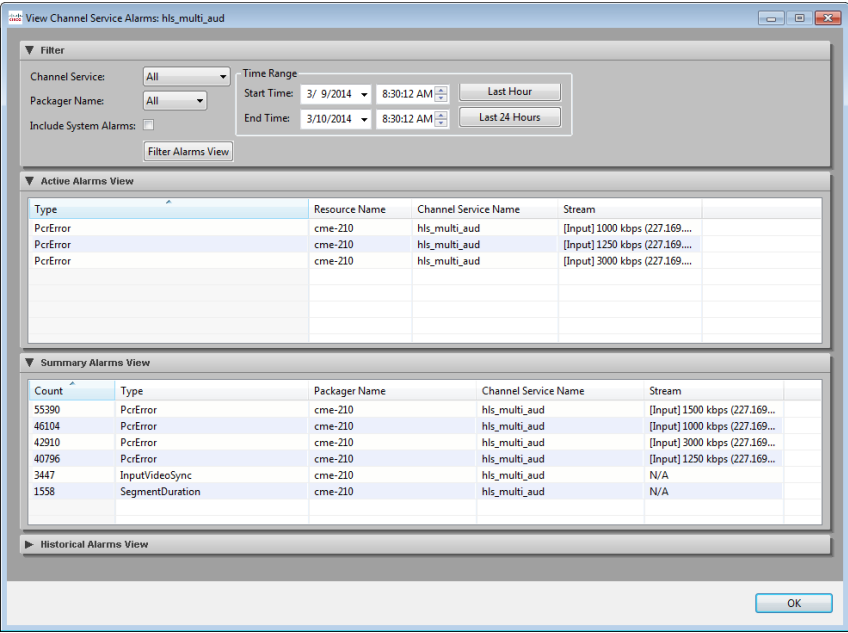
If these steps do not resolve the alarm, contact service personnel.

Note: Other troubleshooting information can be found in the Cisco AnyRes Live Family User Guide. To view the guide, open the web interface for the encoder and click **Help**.

Packager Alarms

On the Channels tab on the Monitor tab, right-click the Channel to display the context menu, and then choose **Channel Service > View Alarms** to display the Channel service alarms. You can choose to filter the displayed alarms by Channel service or packager, or restrict the time range of displayed alarms. Check the **Include System**

Alarms check box to display these alarms in the list. After modifying filter options, click **Filter Alarms View** to update the alarms.



Active Alarms View

The active alarms view displays the alarms that are currently active. The following information is shown for each active alarm:

- **Type**—Displays the type of alarm that was generated. See the table of alarm types below.
- **Resource Name**—Displays the name of the resource that contained the Channel.
- **Channel Service Name**—Displays the Channel service that generated the alarm.
- **Stream**—Displays the Channel stream.

Summary Alarms View

The summary alarms view displays the alarms that have been generated during the displayed time period. The following information is shown for each summary alarm:



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- Count—Displays the count of times the alarm has been generated during the displayed time period.
- Type—Displays the type of alarm that was generated. See the table of alarm types below.
- Packager Name—Displays the name of the packager that contained the Channel.
- Channel Service Name—Displays the Channel service that generated the alarm.
- Stream—Displays the Channel stream.

Historical Alarms View

The historical alarms view displays the history of alarms that have been generated and cleared during the displayed time period. The following information is shown for each historical alarm:

- Date and Time—Displays the time that the alarm was generated or cleared.
- State—Displays whether the alarm was raised or cleared.

State Icon	Description
	Alarm Cleared
	Alarm Raised

- Type—Displays the type of alarm that was generated.
- Packager Name—Displays the name of the packager that contained the Channel.
- Channel Service Name—Displays the Channel service that generated the alarm.
- Stream—Displays the Channel stream.

Alarm Descriptions

The following tables detail the alarm triggers and descriptions.

System Alarms

Type	Description/Recommended Action
SystemCpuHigh	The CPU usage has exceeded the threshold of 80% load. Reduce the number of Channels or output streams.
SystemMemoryHigh	The memory usage for the system has exceeded the threshold of 80% usage. Reduce the number of Channels or output streams.
SystemInterfaceDown	A configured Ethernet interface is not available for use.

Input Alarms

Type	Description/Recommended Action
InputUnderflow	The packager is not receiving input multicast traffic for audio or video. Make sure that that video is active on the source and that the cable for that source is properly connected.
InputOverflow	The input buffer fullness has exceeded the threshold of 50% of 18 MB.
InputVideoSync	Two or more of the input streams are not in sync with each other.
SyncByteError	Sync byte not compliant with TS specification.
TransportError	Transport error indicator in TS input.
CrcError	CRC error occurred in CAT, PAT, PMT, NIT, EIT, BAT, SDT, or TOT table.
PatError	PAT not present in defined interval.
PmtError	PMT not present in defined interval.
CatError	CAT not present per TS specification.
CcError	Incorrect packet order, packet occurs more than twice, or lost packet.
PidError	The packager has not received information about this PID in more than 5 seconds. This checks for video and audio elementary streams and would be an indication that the packager is not receiving any data on those PIDs.

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Type	Description/Recommended Action
PcrError	The interval between PCR values has exceeded the threshold of 40 ms. The PCR value is used to generate the clock for the stream. It is updated regularly to alleviate potential for jitter or drift.
PtsError	The time between PTS values has exceeded the threshold of 700 ms. This error could indicate an issue if the PTS for the IDR frame that starts a fragment is missing.
InputEbpError	EBP has not been seen on the Channel for over 30 seconds when the Channel is configured in EBP mode.
ConfigMismatch	Input stream and Channel configuration do not match.
InputMissingAudio	The input stream is missing audio data for one or more audio PIDs. The first audio PID data will be substituted for the missing audio PID or PIDs.
InputMissingAudioNotReplaced	The input stream is missing audio data for one or more audio PIDs, and the packager is unable to replace the data.

Output Alarms

Type	Description/Recommended Action
Os1NotConnected, Os2NotConnected	Cannot send to origin server or to redundant origin server.
OutputOverflow	Either cannot send to origin server or connection is slow.
Segment Duration	One or more segment durations differ from other segment durations, and client playback may have problems. Check whether incoming streams are correctly aligned with each other.
GenericNotConnected	Connection lost to origin server.
PartitioningFailure	Segments/fragments are not being cut by the packager.
OutputEncryptionFailed	A publisher is configured for encryption, but has failed to encrypt a segment or fragment.

Type	Description/Recommended Action
OutputStreamAlignment	The output media segments being produced are misaligned. The playback of this media may be discontinuous when switching between bit rates.
OutputInitFailure	The output publisher has failed to initialize. The publishing stream and type that has failed to initialize will not produce output.

DRM Alarms

Type	Description/Recommended Action
KeyRotationFailed	A key rotation request has failed. This alarm is a global alarm raised by the packager.
DrmServerNotConnected	Communication has been interrupted between the Cisco Keystore Server and the LSM Server.
DrmServerOutOfSync	The LSM and the DRM server are not time synchronized. Ensure that both servers are synchronized against a local NTP server.

The following glossary covers terms mentioned in this guide, and is adapted from several online sources.

Glossary G

Term	Definition
.avi	The file name extension for a video file in Audio Video Interlaced (AVI) format.
.avs	The file name extension of an AviSynth script file.
.dv	The file name extension of a DV stream.
.gxf	The file name extension of a General Exchange Format file.
.mp3	The file name extension of an audio file in MPEG Layer 3 format.
.mp4	The file name extension of a video file in MPEG-4 part 14 format.
.prx	The file name extension of a Windows Media Profile file.
.trp	The file name extension of an mpeg-2 transport stream.
.ts	The file name extension of a transport stream MPEG-2 video stream.
.wav	The file name extension for a sound file in WAV format.
.wma	The file name extension of an audio file in Windows Media Format.
.wmv	The file name extension of a video file in Windows Media Format.
.yuv	The file name extension of a raw YUV file.
Audio Video Interleaved (AVI)	Digital media file format for storing sound and video.
bit rate	Number of bits transferred per unit of time, typically expressed in bits per second.
buffer	Area of computer memory that briefly holds data before that data is used on the receiving computer. Buffering protects against the interruption of data flow.
caption	This text accompanies images or videos, either as a supplemental description or a transcript of spoken words.

Term	Definition
codec	Abbreviation for compressor/decompressor. Software or hardware used to compress and decompress digital media.
compression	Process for removing redundant data from a digital media file or stream to reduce its size or the bandwidth used.
constant bit rate (CBR)	A data stream whose bit rate remains nearly uniform for the duration of the stream.
differentials	Indicators of transitions between compressed frames.
encode	Converting audio and video content to a specified digital format.
encoder	Device that converts live or prerecorded audio and video content to a specified digital format. Content is usually compressed during encoding. Windows Media Encoder is an example of an encoder. Also used as a generic name to refer to AnyRes Live.
frame	One of many sequential images that comprise video.
frame rate	Number of video frames displayed per second. Higher frame rates usually produce smoother movement in the picture.
genlock (generator locking)	A technique to synchronize video sources with a reference signal from a signal generator or the video output of another source.
interlace	To display a video frame in two fields. One field contains the even lines of the frame; the other field contains the odd lines. During playback, the lines in one field are displayed first; the lines in the second field are displayed next.
inverse telecine	Process that removes the frames that were added when 24 fps film was converted to 30 fps video.
key frame	Video frame containing all data needed to construct an image without reference to previous frames. Also known as an I frame.
metadata	Information about digital media content such as the artist, title, album, producer, and so forth.

CISCO LINEAR STREAM MANAGER

Term	Definition
multi-channel audio	An audio reproduction system that processes several, typically more than two, channels of sound. For example, 5.1 multi-channel audio refers to a surround sound system in which there are five primary channels and a subwoofer channel.
multiple bit rate (MBR)	Data stream in which the same content is encoded at several different bit rates to optimize content delivery.
Progressive Segmented Frame (PsF)	Standard frame types are p—progressive, i—interlaced, and PsF—progressive segmented frame. A PsF frame is a progressive frame that sent as two interlaced fields. This increases slow frame rates, such as 24 fps, above the human flicker perception frequency. 24 fps becomes 48 fps.
telecine	Film-to-video conversion system that adds frames to video to compensate for the differences in frame rates between film and video.
timeline	The area of the user interface that shows the timing and arrangement of frames, files or clips that make up a project.
variable bit rate (VBR)	Data stream in which the bit rate fluctuates, depending upon the complexity of the data.
WAV	Digital media file format for storing sound.
Windows Media Audio (WMA)	Audio file or stream in Windows Media Format. The audio content of the file or stream is encoded with one of the Windows Media Audio codecs.
Windows Media file	A file that contains audio, video, or script data. The content of the file is encoded with one of the Windows Media codecs.
Windows Media Format	Format of a digital media file or stream that was encoded with Windows Media codecs.
Windows Media Video (WMV)	Video file or stream in Windows Media Format. The video content of the file or stream is encoded with one of the Windows Media Video codecs.
Windows Media Video 9 Advanced Profile (WVC-1)	A codec that offers support for interlaced content.

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