MediaFirst Video Processing

Encoding Live Version 7.0

CONFIGURATION GUIDE

P/N 1010-00328-01 Rev01

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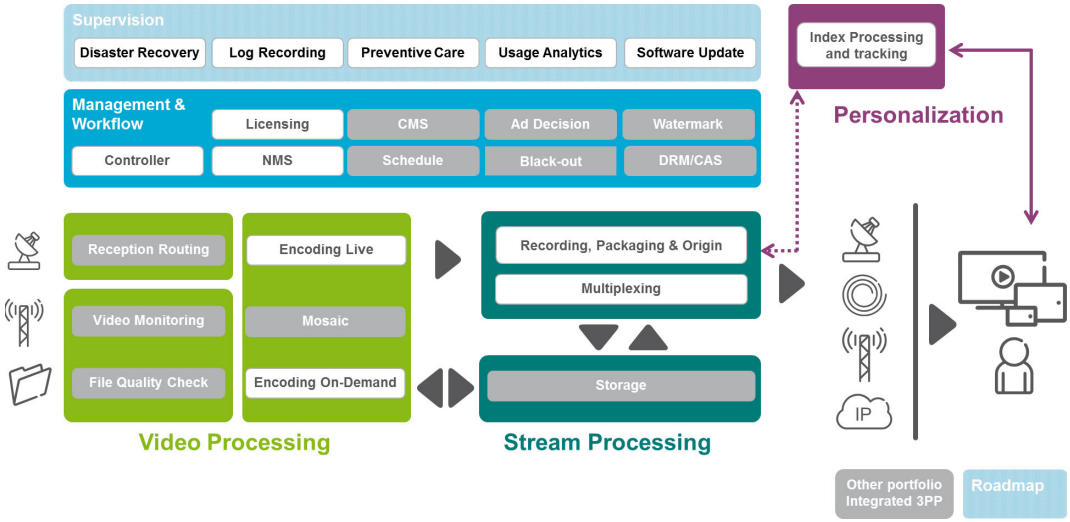
Figure 1. Solution overview



High Quality Encoding for Any Screen

MediaFirst Video Processing - Encoding Live brings together 25 years of video compression experience to deliver the highest quality, any-screen software applications for live video encoding and transcoding. No matter which compute platform is selected, custom hardware or standard servers, Ericsson's continued investment and focus on the latest compression technologies ensures that the MFVP Encoding live capabilities will deliver the best picture quality over bandwidth in all encoding environments.

Virtualized and Standard Server Deployments

With its 'Up!' compression mode, MFVP Encoding live for standard server and virtualized deployments improves video quality, saves bandwidth, and future-proofs operations. It offers an IP-centric and IT-oriented approach to video transcoding to all standards, including MPEG-2, AVC and HEVC, and supports traditional set-top boxes as well as Apple iOS,

Android and 3GPP smartphones, tablets, PCs with HTML2 browsers, Flash or Silverlight, gaming consoles and connected TVs.

MFVP Encoding live for standard server and virtualized deployments is ideal for any real-time broadcast application, including IPTV, cable, DTH, Internet and mobile. Thanks to its superior IP statistical multiplexing, operators can fit more MPEG-2 or MPEG-4 AVC (H.264) channels within their available bandwidth (up to 4:1 or 5:1 HD for cable or 8 HD channels in a DTH transponder). It also supports advanced services such as ad insertion and content protection for personal devices.

Extended functionalities

MFVP Encoding live performs all your head end functions in one single product: signal analysis, decoding, video overlay, content replacement, filter & conversion, loudness control, encoding, packaging and encryption. Replace your rack of video production equipment by a single powerful software-based platform!

Video compression

MFVP Encoding live provides the best video quality for your content in MPEG-2, H.264 and HEVC across all networks and devices. With the “Up!” compression mode, gain more granular control over your video compression settings to achieve the optimal quality.

Ericsson’s research into compression algorithms ensures that MFVP Encoding live benefits from continuous video quality improvement. For the same quality and for a given codec, the bandwidth required is reduced each year. This leads to big operational savings! More HD channels per QAM, increasing IPTV eligibility on your DSLAM deployment, reducing the number of satellite transponders or lowering your CDN bill.

Video Pre-processing

Thanks to its top-end preprocessing filters, MFVP Encoding live improves the quality of the source content to provide the best end user experience:

- The de-blocking, de-ringing and de-noising filters clean up and enhance compressed sources to remove the most visible compression artifacts
- The aspect ratio management dynamically adapts the output to 4:3 or 16:9 aspect ratios when dynamic changes are not supported

- The image settings modify the brightness and color settings
- The de-interlacing, cross-scaling and inverse telecine filters perform advanced conversion from interlaced to progressive video in any resolution. Ericsson's TrueMotion delivers smooth content to OTT players due to the unique progressive image and higher frame rate conversion.

Fully automated pre-processing optimizes the video quality and simplifies the workflow. It automatically configures its pre-processing filters based on input analysis.

Audio compression and pre-processing

MFVP Encoding live offers the latest standards in audio compression technologies with the support of Dolby Digital, Dolby Digital Plus, High Efficiency AAC and Surround Sound up to 5.1 channels.

It offers a built-in loudness control allowing compliance with loudness regulation standards.

Integrated tools for video quality monitoring

MFVP Encoding live monitors the MPEG-2 TS ETR 290, using the RTP headers and FEC information to correct input streams and raise alarms in case of anomalies, for both multicast sources in active/active mode. Furthermore, it has built-in video quality monitoring to ensure that operations are instantly aware of any drop of video quality on any input or output.

Content replacement

MFVP Encoding live lets you seamlessly switch between video sources at a scheduled time. Operators can now streamline their ad-replacement or content black-out workflows without any glitch.

Deploy anywhere

MFVP Encoding live can be deployed as an appliance on optimized Ericsson platforms, as software on bare bone servers, or in the cloud as a virtual instance. This gives more flexibility to manage operations and deployment.

Regular software upgrade are available to benefit from continuous processing, workflow and quality enhancements.

Support information

CAUTION It is not allowed to perform installation or modification of any software on the appliance as it could impact its proper functioning and could limit the warranty.

You may contact us for specific projects requiring customized options or specific development, available through our service organization. For contact information, see the Ericsson website at http://www.ericsson.com.

If you have support questions, contact your Ericsson regional support or send an email to your Sales contact.

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- **EMEA Region**:
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  South Korea +82 2 2005 4633
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Chapter 2

Quick start

This chapter explains how to quickly set parameters for encoding a stream.

Step 1: Setting the IP address[1]

If you are using G4 or G5

By default, the network management interface (Ethernet 6 on 4Caster G4 or Ethernet 1 on G5) is set with a static address: 10.0.0.1XX.

You can configure the network interfaces (Ethernet 1 to Ethernet 6) via the web interface *(see page 31)*.

If you are using G6 or X1

For G6 and G7 1000 & 2000 Series and on X1, please refer to the corresponding Installation Guide.

1. This section is not relevant for MFVP Encoding live, software version on HP or Cisco blade.

Step 2: Setting the input source

1. From any computer that has a network access to MFVP Encoding live, open a web browser and enter the following address: **`http://IP_address/`**

   …where *`IP_address`* is the IP address of MFVP Encoding live (connected to the management network).

   The MFVP Encoding live web interface is displayed.

2. Select the input type used to capture the video stream:
   - Select the input type from the drop down list.
   - If you choose **On-board**, select the input type and the standard
     See "On board input" on page 48.
   - If you choose **Ethernet**, enter the source IP address and port where your input source is broadcasting then select the audio and video PIDs detected in the stream.
     See "Ethernet input" on page 59.

   See Installation Guide, for more details on inputs.

3. Select the video and audio sources.

Step 3: Setting the encoding profile parameters

4. Create one or more profile(s) and set parameters *(see Chapter 7, Chapter 6, or Chapter 5 depending on the export types)*.
5. Click the  button at the profile level to start encoding a profile *(see page 40)*.

Chapter 3

Web interface basics

Accessing the web interface

To access MFVP Encoding live web interface, follow these steps:

1. From any computer that has a network access to MFVP Encoding live, open a web browser.
2. For the first connection, enter the following address: **`http://10.0.0.1XX/`**

   (where **XX** are the two last digits of the serial number located on the sticker at the rear of the chassis - See Installation Guide, Chapter 1, "Network connectors (management")

- On 4Caster G4 and on G5 2000 Series, by default, the network management interface is **Ethernet 6**.
- On G5 1000 Series, and on G6 and G7 1000 & 2000 Series, and on X1, by default, the network management interface is **Ethernet 1**

3. You can configure the network interfaces (Ethernet 1 to Ethernet 6) via the web interface (see "Configuring the Ethernet connectors" on page 31).

NOTE On G6 and G7 1000 & 2000 Series, and on X1, please refer to the Installation Guide for network interfaces configuration.

Web browser

The web browser must support HTML 4.0. We recommend Internet Explorer 8.0 or later.

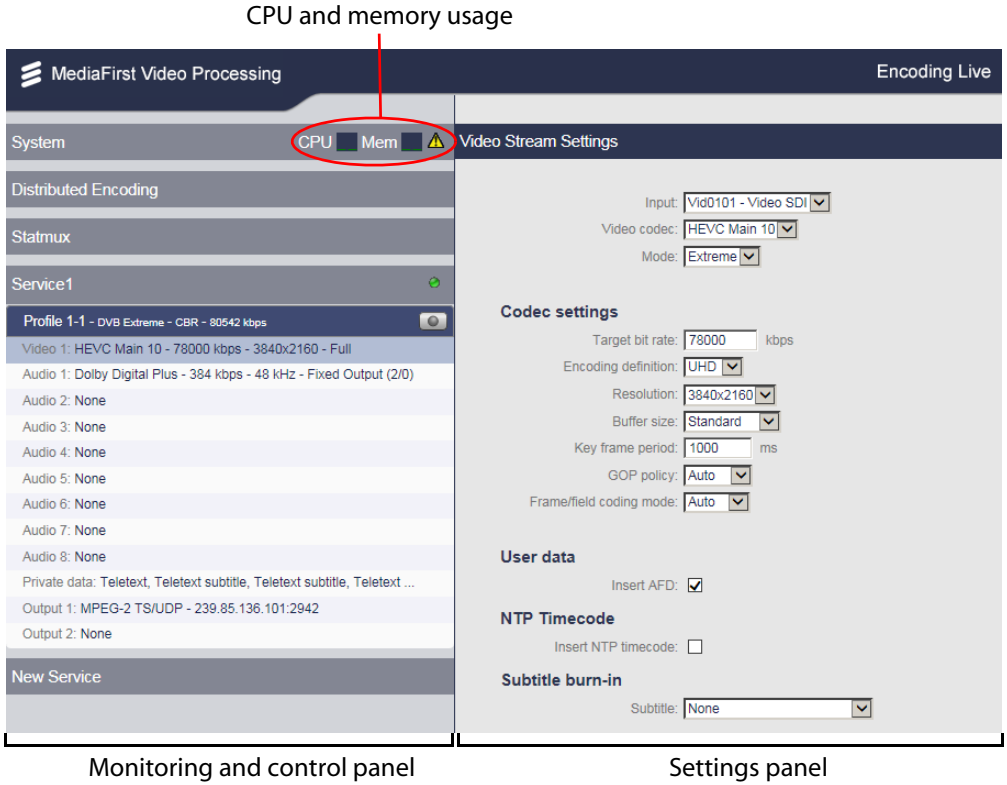
The web interface is automatically refreshed every 3 seconds to maintain consistency between the web interface and MFVP Encoding live.

Web interface overview

The MFVP Encoding live web interface is divided in two panels:

- The **monitoring and control panel**, on the left side. From this panel, you can view the current configuration of MFVP Encoding live and perform some actions such as start/stop a profile.
- The **settings panel**, on the right side. From this panel, you can display the settings of the selected tab and change the parameters.
  An error message may be displayed in the top black banner.

Figure 2. MFVP Encoding live web interface



The monitoring and control panel is divided in 3 main sub-panels:

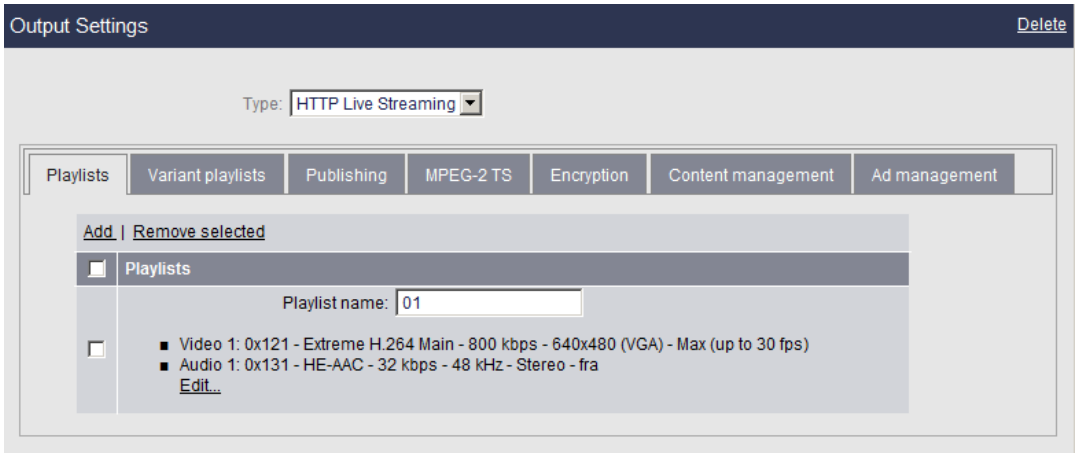
- **System**: This panel lets you configure system settings such as access control, view alarms and logs, import/export configuration parameters, manage credentials, update MFVP Encoding live, display system performance, manage DRM and PlayReady, shut down or restart MFVP Encoding live, or view MFVP Encoding live version information.
- **Service**: Each service includes at least one profile. A service corresponds to an audio/video input with a logical name. The number of managed services depends on the license. This panel lets you set the source, and the audio/video capture input parameters. MFVP Encoding live can manage one video input and up to eight audio inputs per service.
- **Profile**: A profile corresponds to particular encoding settings of one channel. This panel lets you configure the encoding parameters for a profile. The number of managed profiles depends on the license.
  MFVP Encoding live can manage up to 32 profiles per service. A profile can include up to 15 video streams.

Output configuration

For IPTV and Internet TV, output parameters are displayed in several tabs.

The number of tabs depends on the export type.

Figure 3. Output parameters: HTTP Live Streaming



You will have to fill all the tabs, one after the other.

Applying modifications

- In popup menus and combo boxes, modifications are applied immediately.
- In alphanumerical fields, modifications are applied:
  - when you click outside the field,
  - when you move to another parameter field, or
  - when you press the ENTER or SCAN key.
- In the **output** menus, some modifications are applied when you press the **OK** button
- In the **System** menus, some modifications are applied when you press the **Apply** button.

Configuring the Ethernet connectors

CAUTION This section and all the features related to Ethernet connectors configuration are not relevant for G6 and G7 1000 & 2000 Series and on X1. For these platforms, please refer to the corresponding Installation Guide.

The following sections explain how to configure the Ethernet connectors.
The network configuration depends on the presence of a DHCP server:

- **If you have a DHCP server,** MFVP Encoding live will obtain an IP address automatically from the DHCP server. You do not need to re-configure the network address.
- **If you do not have a DHCP server,** you have to configure the network interface(s) with a static address.

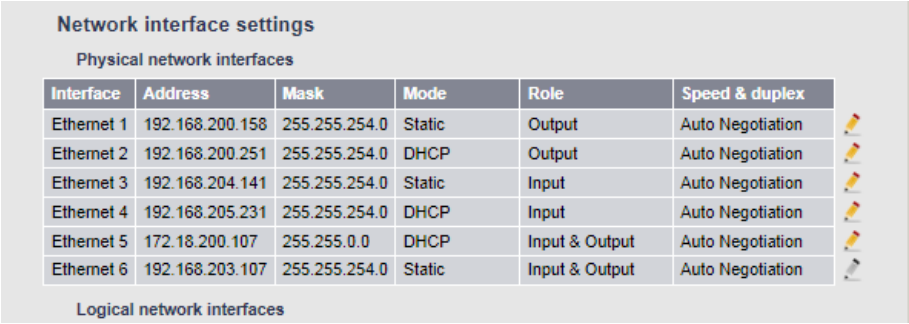
NOTE By default, the Ethernet interfaces use a DHCP configuration, except the management interface that is configured with a static IP address (10.0.0.1XX).

Configuring the physical network interface

To configure the Ethernet connectors, follow these steps:

1. Click the **System** tab, then click the **Configuration** subtab.

Figure 4. Configuration panel



Network interface settings

Physical network interfaces

Interface	Address	Mask	Mode	Role	Speed & duplex
Ethernet 1	192.168.200.158	255.255.254.0	Static	Output	Auto Negotiation
Ethernet 2	192.168.200.251	255.255.254.0	DHCP	Output	Auto Negotiation
Ethernet 3	192.168.204.141	255.255.254.0	Static	Input	Auto Negotiation
Ethernet 4	192.168.205.231	255.255.254.0	DHCP	Input	Auto Negotiation
Ethernet 5	172.18.200.107	255.255.0.0	DHCP	Input & Output	Auto Negotiation
Ethernet 6	192.168.203.107	255.255.254.0	Static	Input & Output	Auto Negotiation

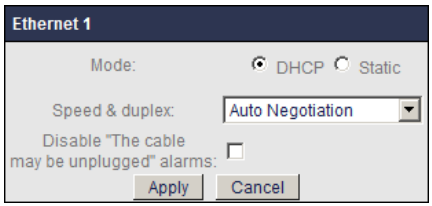
Logical network interfaces

2. In the **Network interfaces settings** panel, click the pencil button, next to the interface to be configured. A new panel is displayed, depending on the current configuration, DHCP or Static.

Configuring an Ethernet connector with DHCP

1. If you select **DHCP**, the following panel is displayed:

Figure 5. Configuring the Ethernet interface with DHCP



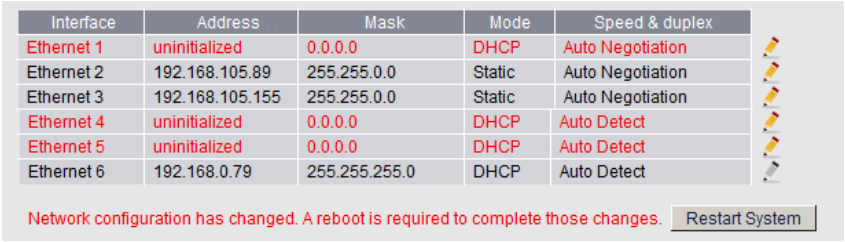
Select **Speed & Duplex**. Possible values are:

- Auto Negotiation
- 10 Mbps Half duplex
- 10 Mbps Full duplex
- 100 Mbps Half duplex
- 100 Mbps Full duplex
- 1.0 Gbps Full Duplex

NOTE Auto-negotiation allows MFVP Encoding live and a peer ethernet entity to share their network capabilities. Then the fastest common transmission mode is used by the two entities. To optimize the connections between two appliances, we recommend selecting another value.

Select the **Disable "the cable may be unplugged" alarms** checkbox if you want the alarms to be ignored when a physical cable is unplugged.

CAUTION If you modify the **Speed & Duplex** value, a warning message will ask you to restart MFVP Encoding live to apply the modifications. Click the **Restart System** button *(see graphic below)*.



Interface	Address	Mask	Mode	Speed & duplex
Ethernet 1	uninitialized	0.0.0.0	DHCP	Auto Negotiation
Ethernet 2	192.168.105.89	255.255.0.0	Static	Auto Negotiation
Ethernet 3	192.168.105.155	255.255.0.0	Static	Auto Negotiation
Ethernet 4	uninitialized	0.0.0.0	DHCP	Auto Detect
Ethernet 5	uninitialized	0.0.0.0	DHCP	Auto Detect
Ethernet 6	192.168.0.79	255.255.255.0	DHCP	Auto Detect

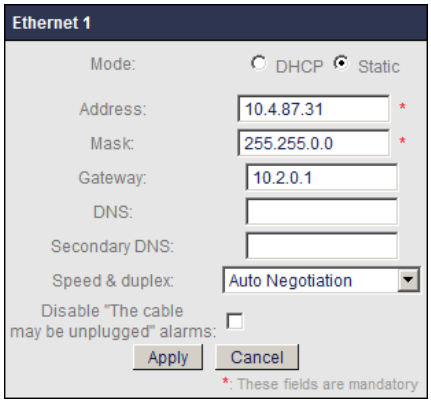
Network configuration has changed. A reboot is required to complete those changes. Restart System

2. Click the **Apply** button to validate your modifications.

Configuring an Ethernet connector with a static IP address

1. If you select **Static**, the following panel is displayed:

Figure 6. Configuring the Ethernet interface with static address



2. Enter the IP address, the network mask, the gateway, the DNS address and the Speed & duplex.

NOTES IP address and Mask are mandatory.

The gateway must be configured on ONLY one interface per subnetwork.

Configuring the logical network interface

MFVP Encoding live can output identical streams on two different interfaces, one interface being on the primary distribution network, the second interface being on the backup distribution network.

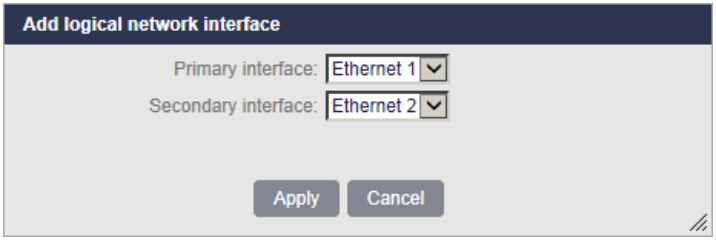
The streams are strictly identical: same source IP, same source port, same destination multicast address, same destination port and identical data. The only difference is the source MAC address (in the Ethernet II layer) of packets from primary NIC and packets from secondary NIC.

To add and configure logical Ethernet connectors, follow these steps:

1. From the **Network interface settings** section, go to the **Logical network interfaces** subsection.
2. Click the **Add** link.
3. Select the primary and secondary NIC from the drop-down list then click **Apply**.

NOTE The network interfaces must have the same role.

Figure 7. Adding logical network interfaces



4. The new logical network interfaces are added to the table.

Figure 8. Table of logical network interfaces

Network interface settings

Physical network interfaces

Interface	Address	Mask	Mode	Role	Speed & duplex
Ethernet 1	192.168.200.158	255.255.254.0	Static	Output	Auto Negotiation
Ethernet 2	192.168.200.251	255.255.254.0	DHCP	Output	Auto Negotiation
Ethernet 3	192.168.204.141	255.255.254.0	Static	Input	Auto Negotiation
Ethernet 4	192.168.205.231	255.255.254.0	DHCP	Input	Auto Negotiation
Ethernet 5	172.18.200.107	255.255.0.0	DHCP	Input & Output	Auto Negotiation
Ethernet 6	192.168.203.107	255.255.254.0	Static	Input & Output	Auto Negotiation

Logical network interfaces

Interface	Primary interface	Secondary interface	Role	Redundancy mode
Ethernet 1/Ethernet 2	Ethernet 1	Ethernet 2 (Active)	Output	Mirroring
Ethernet 3/Ethernet 4	Ethernet 3	Ethernet 4	Input	Failover
Ethernet 5/Ethernet 6	Ethernet 5	Ethernet 6	Management	Failover

5. You can configure them by clicking the pencil button, next to the interface to be configured. A new panel is displayed.

Figure 9. Logical network interfaces

Ethernet 1/Ethernet 2

Redundancy mode: ○ Failover ◉ Mirroring

Apply Cancel

You can select **Failover** or **Mirroring**.

- If you select **Failover**, either the primary interface or the secondary interface streams. It automatically switches from one to the other in case of network issues.
- If you select **Mirroring,** both primary interface and secondary interface stream.

NOTES

Mirroring is only available with IPTV DVB and ATSC Premium and Extreme export types, as well as Internet TV (DVB/ATSC), MPEG-2 TS over UDP and MPEG-2 TS over RTP outputs.

These two modes are exclusive: on a single MFVP Encoding live, it is not possible to use the same logical interface for mirroring and for backup.

Network interface switching/toggling

NOTE This feature is not available with MFVP Encoding live software version.

You can select the button to switch/toggle between primary and backup interfaces.

In case of input source failure, the encoder automatically switches to the secondary interface. If you want to go back to normal operation mode, you can manually force the active interface to be the primary one for this service.

NOTES

Network interface toggling for outputs is global to the system, i.e. it applies on every output that use the NIC.

Redundancy mode must be set to **Failover.**

Network interface toggling is possible in both directions: from primary to backup, or from backup to primary.

Network interface toggling does not disable automatic switching.

After reboot, the active network interface is always the primary one.

Configuring IGMP

You can configure the IGMP protocol version by selecting either version 2 or version 3.

To change the IGMP version, follow these steps

1. Click the **System** tab, then click the **Configuration** subtab. The configuration options are displayed in the right panel.
2. From the **IGMP settings** section, select **IGMPv2** or **IGMPv3** in the **Protocol** drop-down list, then click the **Apply** button.
3. You are asked to reboot MFVP Encoding live. Click **OK** to apply changes. If you click Cancel, the modification is not taken into account and MFVP Encoding live is not restarted.

Configuring IGMPv3 redundancy

To activate the IGMP v3 redundancy, follow these steps:

1. At the system level, set the IGMP protocol to **IGMPv3** (see *"Configuring IGMP" on page 37*).
2. At the Service level, enter the source IP address (see **"Address" on page 62**) and the multicast source IP addresses (see **"Multicast source IP addresses" on page 62**).

IGMP v3-based source redundancy

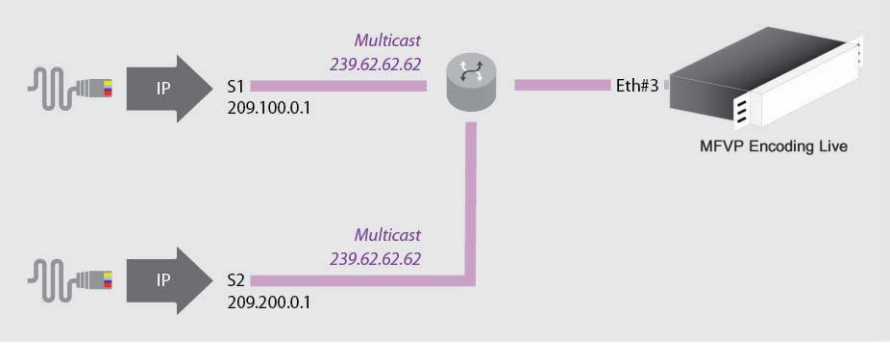
An IP capture can listen to different sources of a multicast address and choose one of them. If a signal loss is detected, the capture automatically switches to another source of the multicast address.

NOTE All the sources must reference the same signals (same content, same PID, same service name).

Redundancy examples

Source is Ethernet3

A capture is configured on **Ethernet 3** and listens to a group G (multicast address 239.62.62.62) of two sources, S1 (209.100.0.1) and S2 (209.200.0.1).

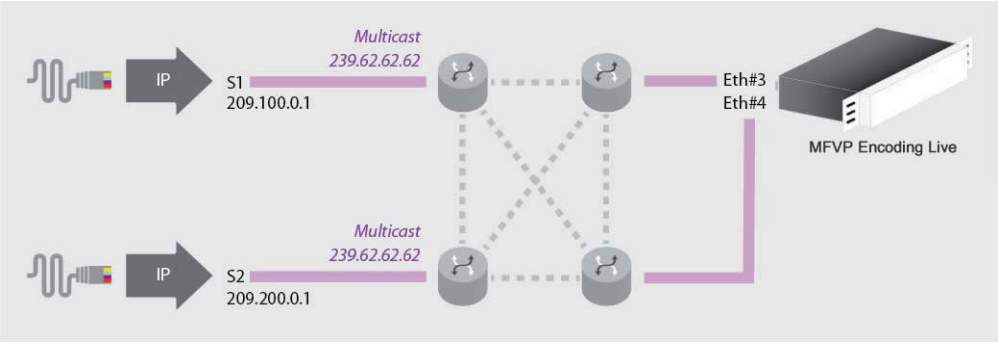


When IGMPv3 is activated:

1. Capture starts to listen to (S1, G).
2. If a signal loss is detected, capture switches to listen to (S2, G).
3. Restart at **step #1**

Source is Ethernet3 (backup: Ethernet4)

A capture is configured on **Ethernet3 (backup: Ethernet4)** and listens to a group G (multicast address 239.62.62.62) of sources S1 (209.100.0.1) and S2 (209.200.0.1).



When IGMPv3 is activated:

1. Capture starts to listen to (S1, G) on Ethernet 3.

If a signal loss is detected:

2. Capture switches to listen to (S1, G) on Ethernet 4.
3. Then capture switches to listen to (S2, G) on Ethernet 3
4. Finally capture switches to listen to (S2, G) on Ethernet 4
5. Restart at **step #1**

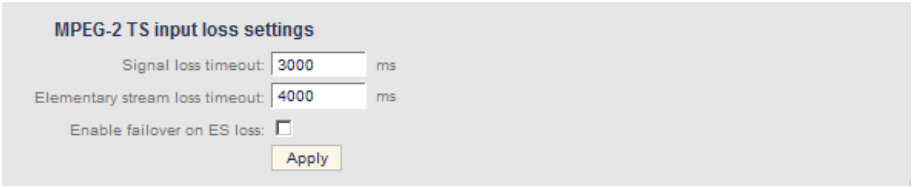
Configuring the MPEG-2 TS input loss settings

A source backup (IGMPv3 redundancy) or NIC backup (NIC redundancy) is triggered when a signal loss is detected on IP input.

A signal loss is triggered when:

- No more signal is detected during the set delay (can be triggered on Elementary Stream loss if set),
- NIC down is detected,
- or NIC toggling is requested.

Figure 10. Input loss settings



From the **MPEG-2 input loss settings** section, select the appropriate value for each parameter.

Signal loss timeout	Set the signal loss delay. Possible values: from **1 000 ms** to **10 000 ms**.
Elementary stream loss timeout	Set the elementary stream loss delay. Possible values: from **1 000 ms** to **10 000 ms**.
Enable failover on ES loss	Check this option to enable the failover on Elementary Stream loss.

Starting/Stopping encoding

The start/stop button, at each profile level, lets you respectively start and stop the current live broadcast.

- To start encoding a profile, click the button.
- To stop encoding a profile, click the button.

When you start encoding different information is displayed in the profile left panel depending on the export type (see page 203 for Mobile Streaming, page 155 for Internet TV and page 183 for IPTV).

Video quality

When encoding a profile, you can display a graphical representation of the video quality. To display the video quality monitor, follow these steps:

1. From the **Profile** menu, go to the **Video quality monitoring** section then select the **video quality monitor** link.
2. A new window is displayed showing two independent metrics:

- the video quality of the input video[1] (named Source quality)
- the encoding video quality (named Stream complexity).

These indicators display the encoder proper performance over the time.

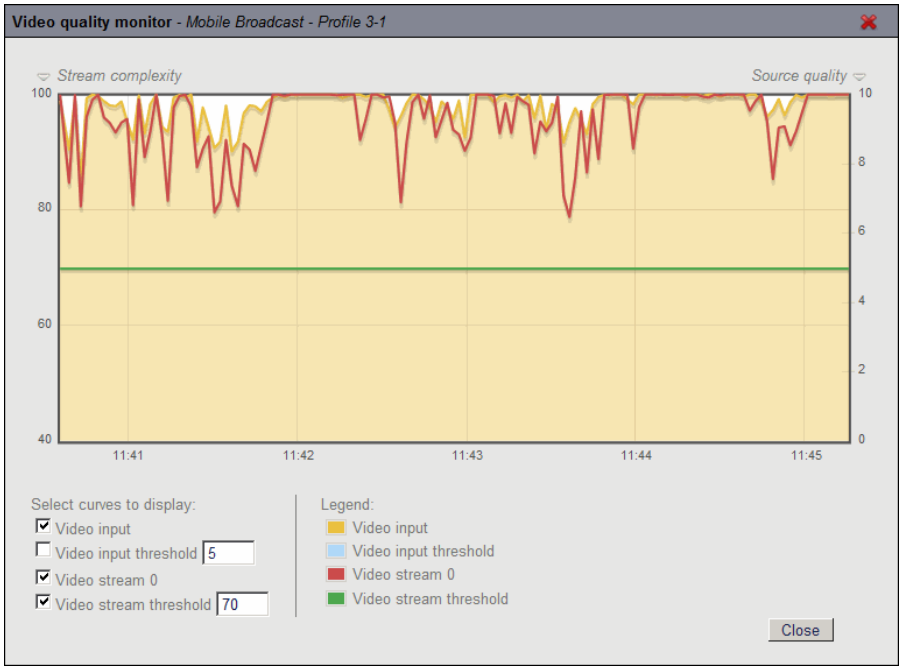
The Source quality metric indicates the video quality measured at the IP input of the encoder.
The score range is [0,10]. A score of 5 means a very bad video quality, a score of 8-10 means a very good one.

The Stream complexity metric measures the encoding performance of the video codec itself. The score range is [40,100]. A score of 70 means a very bad encoding quality, a score above 90 means a very good encoding quality.

NOTE Remember that a low encoding video quality can be due to difficult encoding settings such as low bit rate or high resolution.

1. Only available with IP input & for MPEG-2 & H264 video codecs.

Figure 11. Profile video quality monitoring



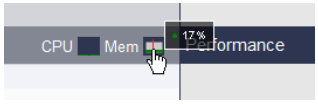
3. You can select the curves to display.

CPU and memory monitoring

The CPU and the memory usage are displayed on MFVP Encoding live main interface as icons.

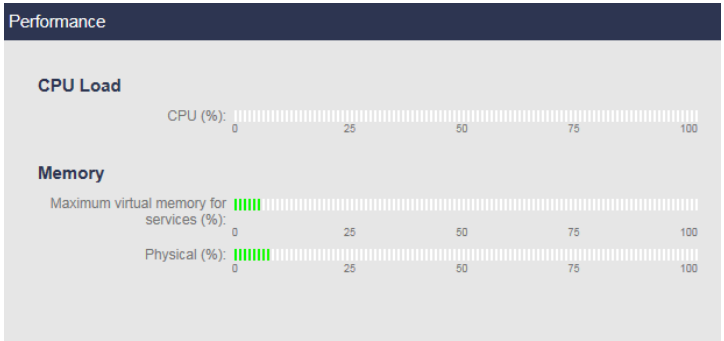
- You can move the pointer over the icons to get quick information.

Figure 12. Getting quick information on memory usage



- or you can click the icons to display additional information.

Figure 13. CPU and memory monitoring



CPU Usage

CPU usage (in percentage) exposes the average CPU usage between all the CPUs on the MFVP Encoding live platform.

Memory usage

Maximum virtual memory for services (%)	Maximum memory used by services in percentage.

Physical (%) Displays the physical Memory usage in percentage.

The physical memory consists of Random Access Memory (RAM) chips and hard drives. RAM is the amount of real storage, and is the total amount of memory installed on a device.

Chapter 4

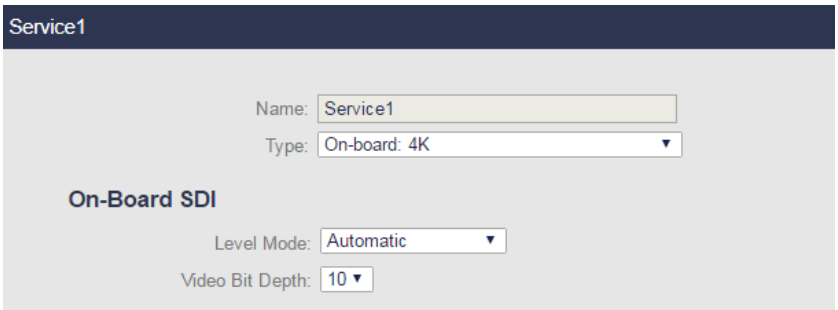
Web interface Input configuration

Configuring the input

To configure the input, follow these steps:

1. Click a **Service** tab. The service parameters are displayed in the right panel.

Figure 14. Input parameters: On board



2. Enter the name of the service
3. Select the input type used to capture the video stream, on board or Ethernet. The corresponding input video and audio parameters are available from the **Service** subtabs.

- **On board**

  Allows using the capture card (*see Installation Guide for more information*).

  Possible values (depending on license):
  On-board: UHD
  On-board: HD
  On-board: SD
  On-Board: Audio Only

- **Ethernet**

  Allows using the MPEG2-TS stream on an ethernet port: the MFVP Encoding live periodically retrieves the information on program in the transport stream (*see Installation Guide for more information*).

  See "Ethernet input" on page 59 for configuring the Ethernet parameters.

  Possible values (depending on license):
  MPEG-2 & H.264 over IP: UHD *(See "Creating an ultra-HD service" on page 48)*
  MPEG-2 & H.264 over IP: HD

MPEG-2 & H.264 over IP: **SD**
MPEG-2 over IP: HD
MPEG-2 over IP: SD
MPEG-2 & H.264 over IP: Audio only
MPEG-2 over IP: Audio only

- **ASI**

  Allows using the ASI card (*see Installation Guide for more information*).

  Possible values (depending on license):
  MPEG-2 over ASI: HD
  MPEG-2 over ASI: SD
  MPEG-2 over ASI: Audio only
  MPEG-2 & H.264 over ASI: HD
  MPEG-2 & H.264 over ASI: **SD**
  MPEG-2 & H.264 over ASI: Audio only

- **RTMP ingest**

  Allows ingesting RTMP transport standard to ingest audio/video streams.

- **VC1 ingest**

  Allows ingesting VC1 standard to ingest audio/video streams.

- **SDI over IP**

  Allows using the SDI stream on an ethernet port.

NOTE We recommend using the 10 Gbe connection.

On board input

Configuring the input

To configure the video capture input, follow these steps:

1. Click a **Service** tab. The **Service** parameters are displayed in the right panel.
2. Select the appropriate value for each parameter.

Name Enter the name of the service.

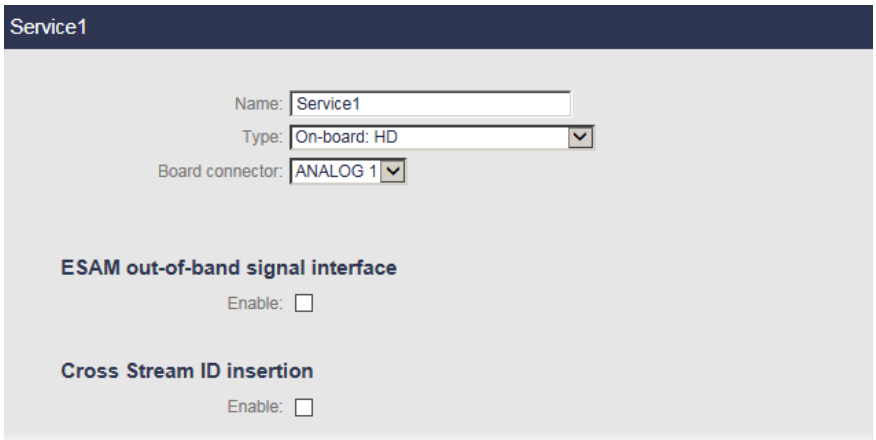
Type Possible values (depending on license): **On-board: SD**, **On-board: HD** or **On-Board: Audio Only**.

3. Select the appropriate board connector.

Board connector Possible values: **SDI 1**, **SDI 2, SDI 3**, **SDI 4, SDI 1 backup SDI 2, SDI 3 backup SDI 4** (*SDI option only*), or **ANALOG 1** to **ANALOG 8** (*Analog option only*)

NOTE: When you select **SDI 1 backup SDI 2**, you can click the button to switch between SDI connectors.

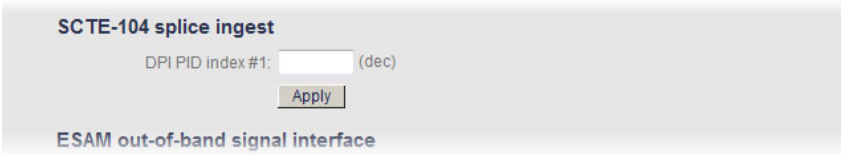
Figure 15. Video input settings



SCTE 104 splice ingest

If the SDI baseband signal contains SCTE 104, you can indicate the SCTE 104 PID.

Figure 16. SCTE 104 parameters



This newly created SCTE 104 will be visible in the **Profile**>**Private data** section (*see "Setting the private stream parameters" on page 119*).

DPI PID index #1 Enter the DPI PID index.

ESAM out-of-band signal interface

The ESAM out-of-band signal interface allows an external module (typically a POIS) to send splice information to MFVP Encoding live. The splice information is used by MFVP Encoding live to generate SCTE 35 signals and/or ad markers in the output.

Figure 17. ESAM parameters

To enable the ESAM out-of-band signal interface parameter, follow these steps:

1. Tick the **Enable** checkbox.
2. Set the appropriate value for each parameter.

   Network name Identify the broadcast TV network.

   Zone identity Identifier for the particular blackout zone of the source.

3. Click the **Rescan** button to validate.

4. MFVP Encoding live ESAM out-of-band signal interface is now accessible from **http://<encoderIP>/SMI.**

NOTES

For ESAM out-of-band, the POIS should perform an HTTP POST request to the URL **http://<encoderIP>/SMI**.

The body of the HTTP request should be an XML document containing a **SignalProcessingNotification** element as defined in the CableLabs Real-time Event Signaling and Management (ESAM) API (OC-SP-ESAM-API-I03-131025).

The **acquisitionPointIdentity** attribute of the **SignalProcessingNotification** element and/or the **ResponseSignal** element should match the **Network name** parameter (see above).

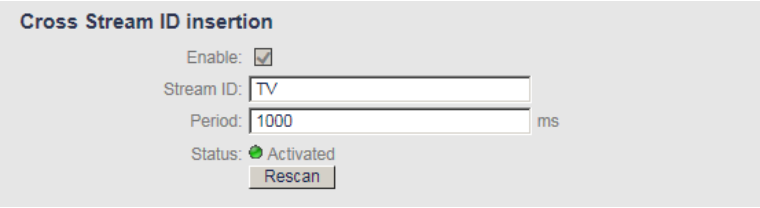
If the **ResponseSignal** contains an **AlternateContent** element to trigger alternate content switching, the **zoneIdentity** attribute of the **AlternateContent** element should match the **Zone identity** parameter (see above)

Cross stream ID insertion

You can add a stream identification in the output stream to identify each stream.

To add information in the stream, follow these steps:

1. From the **Service** tab, go to the **Cross Stream ID insertion** section then tick the **Enable** checkbox.



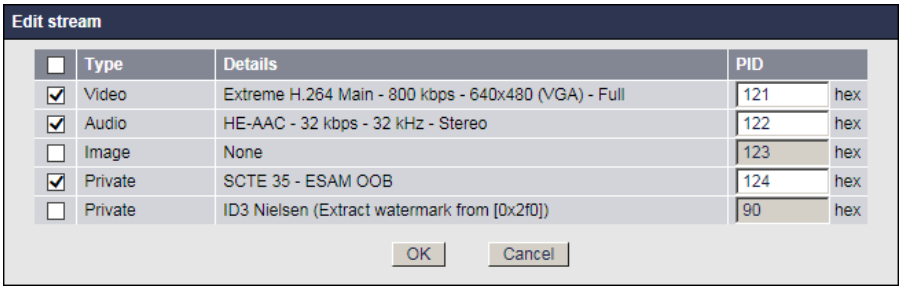
2. Select the appropriate value for each parameter.

Stream ID Enter a stream ID (value that will be inserted in the upid of an SCTE35 message).

Period Specify the time between two SCTE35 messages.
Possible values: From **500** to **3600000 ms**

3. Click the **Rescan** button.

4. You must insert this information in the output: from the **Profile**>**Output** menu, select the **Streams** tab, then tick the stream corresponding to the **SCTE35 - Cross Stream Index** stream,

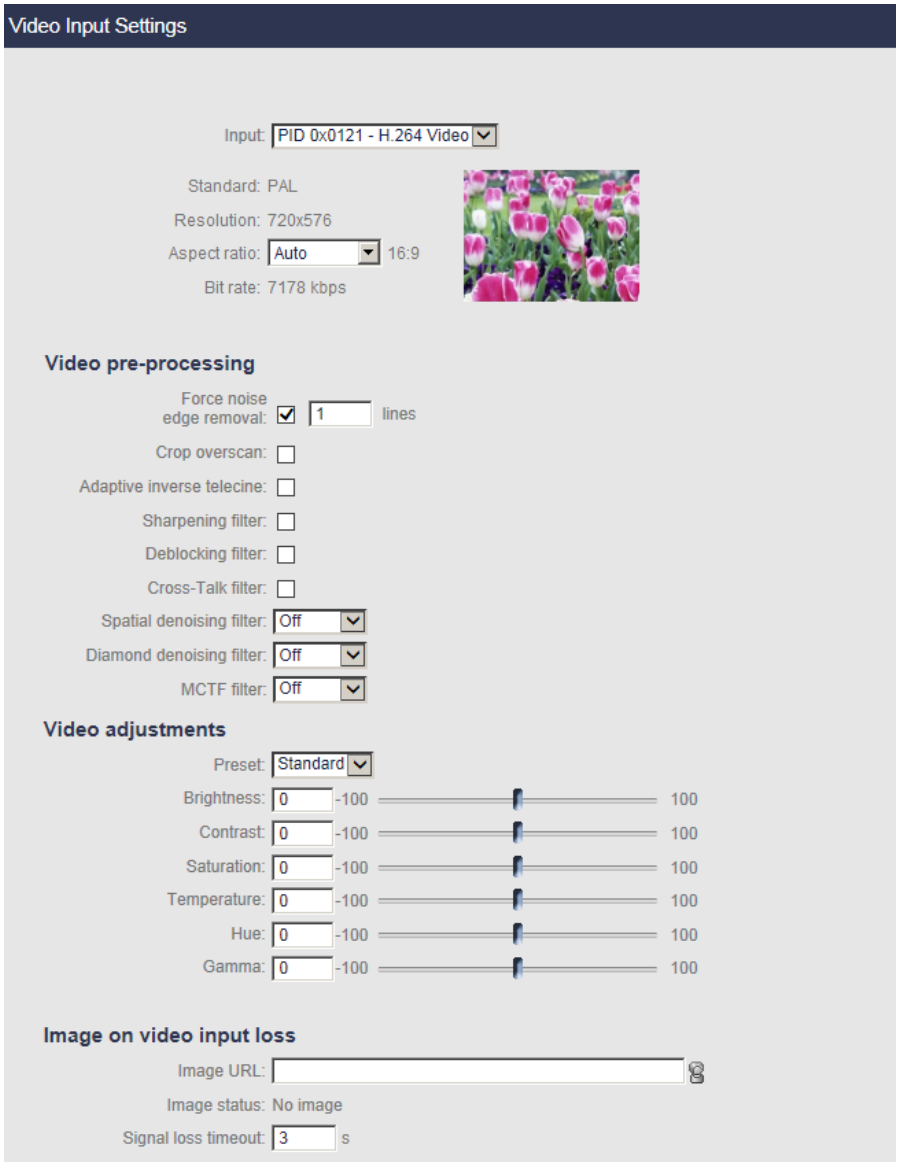


Configuring the video input settings

To configure the video capture input, follow these steps:

1. Click the **Service** tab, then click the **Video input** subtab. The video capture parameters are displayed in the right panel.

Figure 18. Video input settings



2. Select the appropriate value for each parameter.

NOTE A preview shows the pre-filtered active video. Changing the value of the parameters has a direct effect on the preview.

Input Select the input video source.

Possible values: **Video SDI** (*SDI option only*), **Composite NTSC, Composite PAL, Composite SECAM** (*Analog option only*), or **None**. *See Installation Guide for more information*.

Information on the **Standard** (PAL/NTSC), **Resolution** and **Bit rate**, is displayed.

Aspect ratio You can define the process that will be applied on the video input signal to fit in the video output format. *(See Appendix D for more details)*.

Possible values: **Auto**, **Force 4:3** or **Force 16:9**.

- If you select **Auto**, the MFVP Encoding live will take into account the WSS or the AFD information embedded in the video input signal during the encoding process and transform the input video as described in Appendix D.

NOTES: If the WSS/AFD is not present at encoding start, the video will be output in 4/3.
If the WSS was present but then is lost, the last detected WSS will be applied.

You can force it by selecting either **Force to 4:3** or **Force to 16:9**.

Video Pre-processing

These parameters are optional.

Force noise edge removal Specify the number of lines to be removed (from **0** to **6** in SD and from **0** to **14** in HD).

This filter blanks out video noise around the edges of the screen that would not be seen on a television, but can appear in a web video window. It should only be used if the source has noisy edges.

Crop overscan Check this option, to remove the overscan area from each edge (left, top, right and bottom) of the input video.

NOTE: Does not apply to **IPTV** export types.

Adaptive inverse telecine

NTSC only. Uncheck this option if you want to deactivate the adaptive inverse telecine pre-processing filter.

Deactivating this filter can be useful when a content must be encoded at 30 fps for interoperability with player device or content distribution solution (ad-insertion).

Sharpening filter

Use this filter to increase the contrast and enhance the edges of text.

When you check this option, you have to define the **Maximum resolution** for which the filter will be applied. If the profile resolution is lower or equal to the max resolution defined the sharpening filter is activated, otherwise if the profile resolution is bigger, the filter is not activated.

Deblocking filter

This filter lets you enhance the quality of an MPEG-2 encoded video by reducing the blocking and ringing effects.

Cross-talk filter

This filter lets you reduce cross-color and cross-luminance artifacts remaining in input video.

The cross-color processing is only visible on "rainbows" in PAL and NTSC sequences. If the video has been scaled beforehand, the filter is not efficient. That's why the cross-color processing is deactivated on HD sequences.

The cross-luminance artifact is a temporal effect; it is not transformed after upscale. The filter is still efficient on HD sequences.

Spatial denoising filter

Set the threshold for Spatial denoising filter

Possible values: **Off, Low, Medium, High**

Diamond denoising filter

This filter lets you remove artefacts on progressive contents.

Set the threshold for Diamond denoising filter

Possible values: **Off, Medium, High, Adaptive**

MCTF filter

This filter lets you reduce the transmission noise (mainly Gaussian noise). You can select the strength of the filter.

Possible values **Off, Low, Medium**, **High, Adaptive**

Video adjustments

Preset

You can select a preset from the drop-down list that automatically adjust the video settings depending on predefined color settings.

Possible values: **Custom** (see below)**, Standard** (no color change)**, Vivid** (intense colors)**, Movie** (suitable for watching movies in a dark room)**, Sport** (Suitable for watching sport)**, Bright** (increased luminosity)

Brightness
Contrast
Saturation
Temperature
Hue
Gamma

If you select **Custom**, you can adjust these video input parameters by dragging the sliders back and forth.

All parameters have a value range of between -100 to +100 in whole digit increments. A value of 0 has no effect.

Image on video input loss

In case of input signal loss or unsupported format, you can select a default image (PNG or JPEG file) that will be displayed in the output.

Enter the path to your custom image (http://<spath>/filename.png or file://<spath>/filename.png) then click outside the URL field.

You can modify the default **signal loss timeout**.

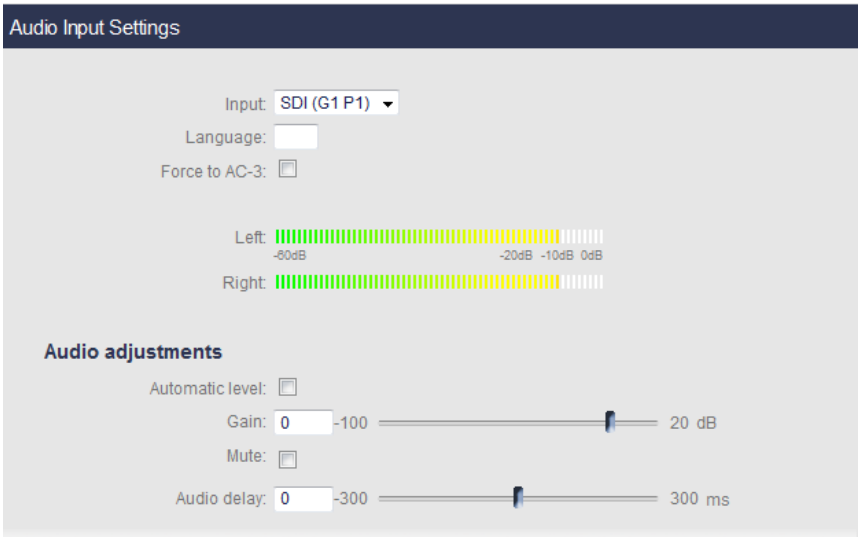
NOTE If you defined credentials to access this URL (*see* "Adding a credential" on page 235), the user icon, should appear in color.

Configuring the audio input settings

To configure the audio capture input, follow these steps:

1. Click the **Service** tab, then click the first **Audio input** subtab. The audio capture parameters are displayed in the right panel.

Figure 19. Audio input settings: unbalanced audio input



2. Select the appropriate value for each parameter.

Input Select the audio input source.

Possible values: **None** (video-only mode), **SDI** (*SDI option only*), or **Unbalanced** (*Analog option only*).

SDI can be one of these values:

- **SDI (G1 P1)**, first audio pair (2 channels) in first group
- **SDI (G1 P2)**, second audio pair in first group
- **SDI (G2 P1)**, first audio pair in second group
- **SDI (G2 P2)**, second audio pair in second group
- **SDI (G3 P1)**, first audio pair in third group
- **SDI (G3 P2)**, second audio pair in third group
- **SDI (G4 P1)**, first audio pair in fourth group
- **SDI (G4 P2)**, second audio pair in fourth group

See Installation Guide for more information.

Language You can set the language by entering an identifier corresponding to the audio channel language (ISO 639-2 three-letter code).

Examples: **eng, fra, deu, spa**. You can also use **qaa** for original language.

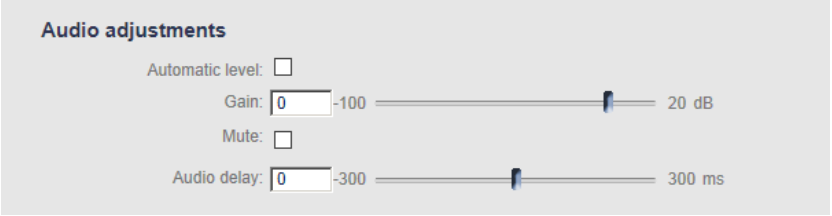
Audio mode *SDI option only*.

Possible values: **AES uncompressed audio**, **SMPTE-337 compressed audio** or **Auto**

If you select **Auto**, the audio mode will automatically adapt to the right audio capture.

Audio level adjustment

Figure 20. Audio input settings: unbalanced audio input



Automatic level Select this checkbox if you wish to maintain the level at the level defined by the **Target loudness** parameter.

We recommend -24 dB for IPTV and -16 dB for Internet TV.

Automatic level **CAUTION:** Audio level must be adjusted to a maximum of -20 dB (check volume level indicators) before activating the automatic audio level adjustment.

NOTES: This parameter has no effect if the audio stream is passed-through.

Gain / Target loudness Move the slider or enter a value to amplify or attenuate the incoming signal.

Possible values: from -100 to +20 dB where 0 is the default gain.

NOTE: This parameter has no effect if the audio stream is passed-through.

Audio delay Move the slider or enter a value to adjust the delay between the audio and the video input.

Possible values: from -300 to +300 ms.

Mute Check this option to mute the audio volume.

NOTE: This parameter has no effect if the audio stream is passed-through.

Left Right Display the left and right audio volume levels for the audio input.

NOTES: The volume indicators show the audio level after adjusting the volume with the **Gain** slider.

3. Repeat these steps to configure the other audio streams.

CAUTION With Analog option, only one audio input is available. We recommend using the first audio input. *(See Installation Guide for more information).*

Ethernet input

Configuring the input

NOTES Both IP Single Program Transport Stream (SPTS) and Multiple Program Transport Stream (MPTS) are supported.

For more information on IP stream characteristics, see "IP ingest" on page 252.

To configure the video capture input, follow these steps:

1. Click a **Service** tab. The **Service** parameters are displayed in the right panel (see graphic below).
2. Select the appropriate value for each parameter.

Figure 21. Configuring the source

Service1 Delete service | Add profile

Name: Service1
Type: MPEG-2 & H.264 over IP: HD

Sources

Input redundancy: Active / Active

Primary source

Network interface: Ethernet 3 (10.3.85.114)
Address: 239.2.3.9
Port: 1234
Enable FEC support:
Capture status: Receiving

Secondary source

Network interface: Ethernet 1 (10.3.85.224)
Address: 239.2.3.9
Port: 1234
Enable FEC support:
Capture status: Receiving

Status

Status: Receiving
Active source: primary
Rescan
Program: Default
Enable content switching:

MPEG-2 TS statistics | FEC statistics | RTP statistics

Quality indicator	Primary source
FEC status:	disabled
Recovered packets:	0
Unrecovered packets:	0
Discarded packets:	0
Discarded FEC packets:	0
Missing FEC packets:	0
FEC total bit rate:	0 kbps

Last reset time: 2015/09/17 17:35:37
Reset

Add a delay on SCTE-35 trigger

Enable:

ESAM out-of-band signal interface

Enable:

Cross Stream ID insertion

Enable:

Name Enter the name of the service.

Type Possible values (*depending on license*): **MPEG-2 & H.264 over IP: UHD, MPEG-2 & H.264 over IP: HD, MPEG-2 & H.264 over IP**: **SD, MPEG-2 over IP: HD**, **MPEG-2 over IP: SD**, **MPEG-2 & H.264 over IP: Audio only, MPEG-2 over IP: Audio only**, **RTMP ingest, SDI over IP**.

Source parameters

You can configure two multicast addresses and define the way they will switch for input redundancy. If the primary source is down then the encoder will connect to the secondary source.

To configure a second source, tick the **Display secondary source** checkbox. A secondary source section is displayed. You can configure both sources.

A manual toggle between primary and secondary source can be triggered manually by ticking the button in the **Status** section.

NOTE To delete a secondary source, delete the secondary source IP address and the Multicast source IP address(es) if you are using IGMP v3 protocol, but don't delete the port; then click the **Rescan** button.

Input redundancy You can specify the way primary and secondary sources will switch.

Possible values: **Active/Passive** or **Active/Active**

- If you select **Active/Passive**, only the primary source is active. In case of input loss (no more multicast data or NIC down) or in case of PID or ES data loss, the input is switched to the secondary source and an alarm is raised describing the new status.
- If you select **Active/Active**, both sources are active. A **Capture status** is associated with each multicast source allowing to switch from one to the other depending on the status of the current active source.

Manual switchback only Check this option to activate manual switchback.

Network interface

Select the appropriate input network interface.
Possible values: **Ethernet3, Ethernet4**, **Ethernet5**, **Ethernet6**, **Ethernet3 (backup: Ethernet4)** or **Ethernet6 (backup: Ethernet5)**
See Installation Guide for more information.

If you select **Ethernet3 (backup: Ethernet4)**, Ethernet 3 will be used as primary and backup will be done on Ethernet 4 if the audio or video PID is lost or if the audio or video input becomes encrypted.

When **Ethernet3 (backup: Ethernet4)** is selected, you can select the button to switch/toggle between Ethernet3 and Ethernet4 interfaces.

In case of input source failure, the encoder automatically switches to the secondary interface. If you want to go back to normal operation mode, you can manually force the active interface to be the primary one for this service.

NOTES: Network interface toggling is possible in both directions: from primary to backup, or from backup to primary.
Network interface toggling does not disable automatic switching.
After reboot, the active network interface is always the primary one.

Address

Enter the multicast IP address from which the source will be received.

Examples: `239.2.3.123`

Port

Enter the port number of the source; for example, `1234`.

Multicast source IP addresses

If you configured the IGMP protocol in **IGMPv3** (*see "Configuring IGMP" on page 37*), you can configure up to five multicast source IP addresses.

Enter a comma-separated list of multicast source IP addresses. Example: `IP1, IP2, IP3`.

Enable FEC support

In case of source containing embedded FEC information, this option lets you get FEC statistics (*See "FEC statistics" on page 64*) to check that there is no content alteration due to packet loss happening on your multicast IP source network.

If you tick this option you will be asked to enter the port number.

Status

Status Displays the status of the service:

- **Receiving** (green): The input capture is receiving the source signal.
- **Searching** (orange): the input capture is searching the source signal. The service can display this status during the first scan or if the input signal is lost.
- **Inactive** (grey): the capture is inactive.

NOTE: In the left panel, in the **Service** tab, a status led lets you know the MPEG-2 TS capture state (grey: capture not configured, orange: checking the capture, or green: capture OK).

Active source Indicates the active source when two sources are defined.

You can click the button to switch/toggle between primary and secondary source.

Program Program number and name.

Auto selects the first available program.

To select the program:

1. Click the **Scan** or **Rescan** button. The list of available programs associated to the selected IP address are displayed.
2. Select the appropriate program in the drop-down list.

Enable content switching service Tick this option if you want to define an alternative service that will be used as alternate video content for Alternate content. (see "Alternate content" on page 89)
Click the **Add alternate content service** link to display an alternate service in the left menu and configure it.

MPEG-2 TS statistics

In case of Ethernet input, **MPEG-2 TS statistics** are displayed in the table. If you selected the **Active/Active** input redundancy, both primary and secondary source statistics are displayed.They indicate the network and transport errors, the number of scrambled packets, the number of video/audio/private data samples, PCR jitter, buffer level and total bit rate since the last reset or scan.

NOTES You can click the **Reset** button to reset all the counters.
Default buffer level must be between 250 and 300 ms.

FEC statistics

If you activated the FEC support, the **FEC statistics** are updated. They indicate the FEC status, the number of recovered/unrecovered/discarded/Missing packets, the FEC bit rate (excluding RTP header) since the last reset or scan.

NOTE You can click the **Reset** button to reset all the counters.

RTP statistics

In case of RTP input, **RTP statistics** are displayed. They indicate the number of dropped/duplicated/reordered packets since the last reset or scan.

NOTE You can click the **Reset** button to reset all the counters.

Add a delay on SCTE-35 trigger

You can add a delay on SCTE-35 triggers to adjust the content replacement.

NOTE Delay can be positive or negative.

ESAM out-of-band signal interface

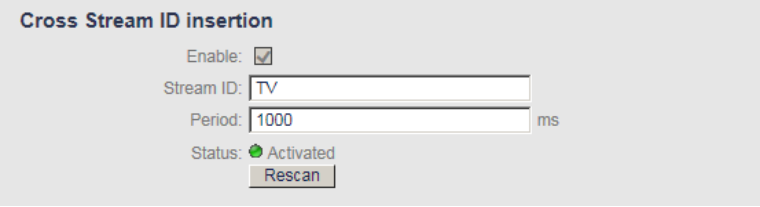
See "ESAM out-of-band signal interface" on page 49.

Cross stream ID insertion

You can add a stream identification in the output stream to identify each stream.

To add information in the stream, follow these steps:

1. From the **Service** tab, go to the **Cross Stream ID insertion** section then tick the **Enable** checkbox.



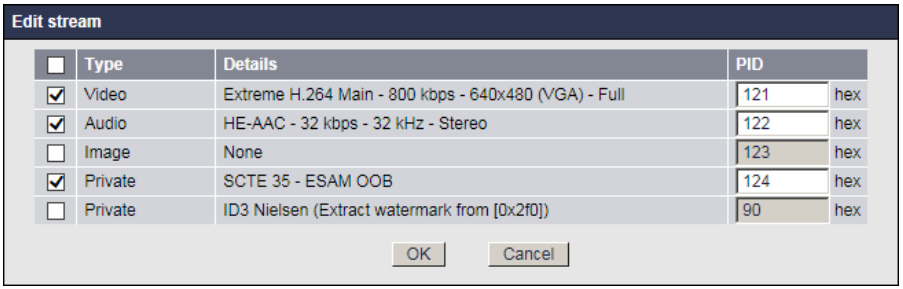
2. Select the appropriate value for each parameter.

Stream ID Enter a stream ID (value that will be inserted in the upid of an SCTE35 message).

Period Specify the time between two SCTE35 messages.
Possible values: From **500** to **3600000 ms**

3. Click the **Rescan** button.

4. You must insert this information in the output: from the **Profile**>**Output** menu, select the **Streams** tab, then tick the stream corresponding to the **SCTE35 - Cross Stream Index** stream,



Managing input events

This section explains how MFVP Encoding live handles some events in the input transport stream such as a stream PID change, a codec or a language change, a PMT or PAT change without a manual "rescan" on the service.

Use case #1: Audio PID missing

In some cases (regional switch), the second audio language is no more available.

- An alarm is triggered to inform that an audio stream is missing.
- The stream bandwidth is reserved on the output in case of stream re-appearance.
- Second audio stream on output is a silent audio.

Use case #2: Video and Audio PID change for service re-mapping

The broadcaster may change the PID list of a program without notice. It results in a PID loss during encoding. In that case, the service is re-scanned to find the corresponding content between the reference configuration and the detected content.

- PID values are updated.
- Input streams are mapped to the existing output streams.
- If an audio/private PID is missing, a data loss alarm is raised.
- The missing stream is always mapped to output with its reserved bandwidth.
- The missing audio stream is a silent audio.

Use case #3: Video and audio PID change for regional content switch

During regional contents switch, PIDs are changed, codecs are changed and some audio or private PIDs are removed.
In this case, the service is re-scanned to find the corresponding stream in the new content.

- Alarms are raised for the lost PIDs and bandwidth is reserved.
- Alarm is raised if codec is switched.
- Audio or video artifacts may occur following codec change.

Use case #4: PAT/PMT change

In the case of provider switch, PAT/PMT version and PID may change. PAT or PMT version change generates a table reprocessing. Streams are then re-mapped as described in the rules below:

- During table reprocessing, a short interruption of service may occur.
- Alarms are raised if PIDs are missing.

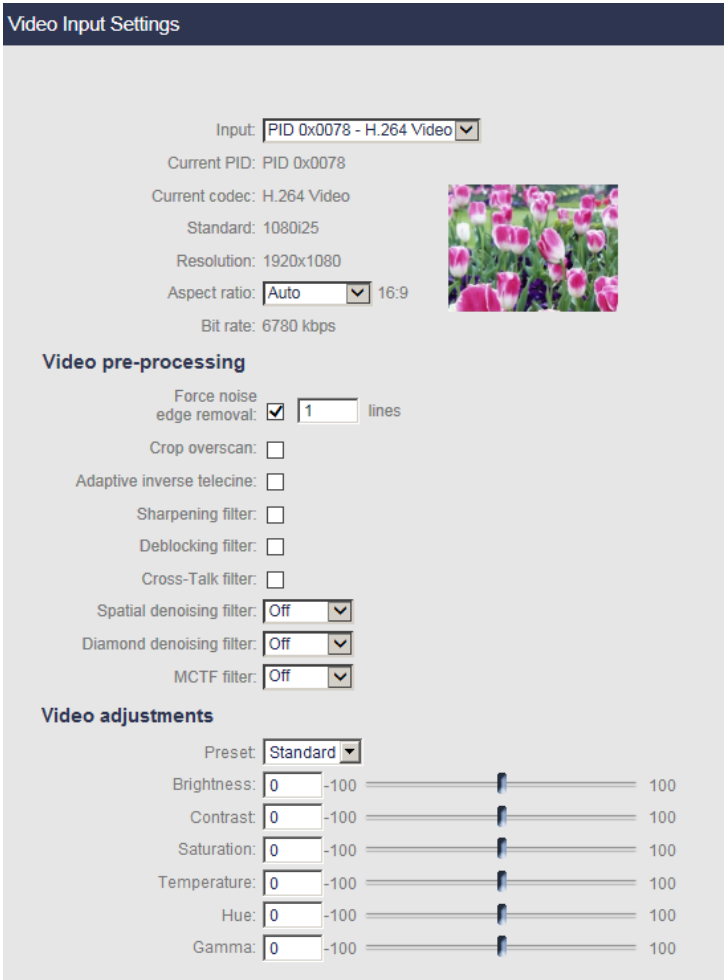
Alarms are raised if codec has changed.

Configuring the video input settings

To configure the video capture input, follow these steps:

1. Click the **Service** tab, then click the **Video input** subtab.
2. The video input parameters are displayed in the right panel.

Figure 22. Input video settings



3. Select the video PID you will use as input source. **None** means audio only input.

A preview shows the pre-filtered active video. Changing the value of the parameters has a direct effect on the preview.

NOTE Information on the input **PID**, **Video codec**, **Standard**, **Resolution, Chroma**[1]**, Bit depth**[1]**, Aspect ratio** and **Bit rate** is displayed next to the preview.

Aspect ratio You can define the process that will be applied on the video input signal to fit in the video output format. *(See Appendix D for more details)*.

Possible values: **Auto**, **Force 4:3** or **Force 16:9**.

- If you select **Auto**, MFVP Encoding live will take into account the WSS or the AFD information embedded in the video input signal during the encoding process and transform the input video as described in Appendix D.
- If you select **Force to 4:3**, no transformation is necessary. The output maintains the same aspect ratio as the video input and black borders may be added to fit in the video output format.
- If you select **Force to 16:9**, the pixel aspect ratio will be restored and black borders may be added to fit in the video output format. Use this option in case of 16:9 anamorphosed video input format.

Video pre-processing

Force noise edge removal Specify the number of lines to be removed (from **0** to **6** in SD and from **0** to **14** in HD).

This filter blanks out video noise around the edges of the screen that would not be seen on a television, but can appear in a web video window. It should only be used if the source has noisy edges.

Crop overscan Check this option, to remove the overscan area from each edge (left, top, right and bottom) of the input video.

NOTE: Does not apply to **IPTV** export types.

1. 4K parameters.

Adaptive inverse telecine

NTSC only. Uncheck this option if you want to deactivate the adaptive inverse telecine pre-processing filter.

Deactivating this filter can be useful when a content must be encoded at 30 fps for interoperability with player device or content distribution solution (ad-insertion).

Sharpening filter

Use this filter to increase the contrast and enhance the edges of text.

When you check this option, you have to define the **Maximum resolution** for which the filter will be applied. If the profile resolution is lower or equal to the max resolution defined the sharpening filter is activated, otherwise if the profile resolution is bigger, the filter is not activated.

Deblocking filter

This filter lets you enhance the quality of an MPEG-2 encoded video by reducing the blocking and ringing effects.

Cross-talk filter

This filter lets you reduce cross-color and cross-luminance artifacts remaining in input video.

The cross-color processing is only visible on "rainbows" in PAL and NTSC sequences. If the video has been scaled beforehand, the filter is not efficient. That's why the cross-color processing is deactivated on HD sequences.

The cross-luminance artifact is a temporal effect; it is not transformed after upscale. The filter is still efficient on HD sequences.

Spatial denoising filter

Set the threshold for Spatial denoising filter

Possible values: **Off, Low, Medium, High**

Diamond denoising filter

This filter lets you remove artefacts on progressive contents.

Set the threshold for Diamond denoising filter

Possible values: **Off, Medium, High, Adaptive** (adapts the filter depending on the nose level)

MCTF filter

This filter lets you reduce the transmission noise (mainly Gaussian noise). You can select the strength of the filter.

Possible values **Off, Low, Medium**, **High, Adaptive** (detects the edges to avoid applying the filter on edges)

Video adjustments

Preset

You can select a preset from the drop-down list that automatically adjust the video settings depending on predefined color settings.

Possible values: **Custom** (see below)**, Standard** (no color change)**, Vivid** (intense colors)**, Movie** (suitable for watching movies in a dark room)**, Sport** (Suitable for watching sport)**, Bright** (increased luminosity)

Brightness
Contrast
Saturation
Temperature
Hue
Gamma

If you select **Custom**, you can adjust these video input parameters by dragging the sliders back and forth.

All parameters have a value range of between -100 to +100 in whole digit increments. A value of 0 has no effect.

Image on video input loss

In case of input signal loss or unsupported format, you can select a default image (PNG or JPEG file) that will be displayed in the output.

Enter the path to your custom image (http://<spath>/filename.png or file://<spath>/filename.png) then click outside the URL field.

You can modify the default **signal loss timeout**.

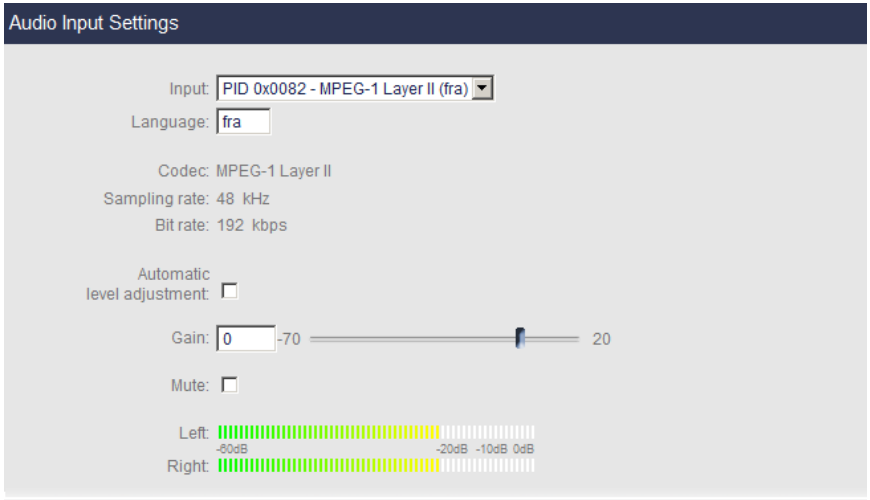
NOTE If you defined credentials to access this URL (*see* "Adding a credential" on page 235), the user icon, should appear in color.

Configuring the audio input settings

To configure the audio capture input, follow these steps:

1. Click the **Service** tab, then click the first **Audio input** subtab. The audio capture parameters are displayed in the right panel.

Figure 23. Ethernet audio settings



2. Select the appropriate value for each parameter.

Input Select the audio PID you will use as input source.

Information on the **Codec**, **Sampling rate** and **Bit rate** is displayed below.

Language You can set the language by entering an identifier corresponding to the audio channel language (ISO 639-2 three-letter code).

Examples: **eng, fra, deu, spa**. You can also use **qaa** for original language.

Dolby E (SDI-specific parameter)

The Dolby E algorithm is optimized for handling discrete multichannel audio programs and multiple audio programs. Encoded audio channels are grouped as programs that are typically mono (one channel), stereo or matrix surround (Lt/Rt) encoded (two channels), or discrete six-channel audio with five full-range channels and a bass-only Low-Frequency Effects (LFE) channel (5.1 channels). The eight available channels can be flexibly assigned in many different program configurations as follows:

- 5.1+2: Assigned to carry a 5.1-channel main audio program, plus a second stereo program in another language.
- 4×2: Assigned to carry four separate stereo programs, such as the inclusion of different language tracks or director's commentary.

- 8×1: Assigned to convey eight individual mono programs, making it highly useful for a wide variety of broadcast applications when 5.1-channel audio is not required.

Dolby E makes use of the available data space only for the number of audio channels that are present, so requiring less space if fewer than eight audio channels are used. The channel configuration is determined at the time of encoding, allowing users to choose the best mode for the specific application.

Program configuration Information field. See table below.

Program See table below.

Table 1. Dolby E Program Configurations

Program Configuration	Program Count	Channel Count	Program Sequence	Channel Sequence
0	2	8	5.1+2	0L, 0C, 0Ls, 1L, 0R, 0LFE, 0Rs, 1R
1	3	8	5.1+1+1	0L, 0C, 0Ls, 1C, 0R, 0LFE, 0Rs, 2C
2	2	8	4+4	0L, 0C, 1L, 1C, 0R, 0S, 1R, 1S
3	3	8	4+2+2	0L, 0C, 1L, 2L, 0R, 0S, 1R, 2R
4	4	8	4+2+1+1	0L, 0C, 1L, 2C, 0R, 0S, 1R, 3C
5	5	8	4+1+1+1+1	0L, 0C, 1C, 3C, 0R, 0S, 2C, 4C
6	4	8	2+2+2+2	0L, 1L, 2L, 3L, 0R, 1R, 2R, 3R
7	5	8	2+2+2+1+1	0L, 1L, 2L, 3C, 0R, 1R, 2R, 4C
8	6	8	2+2+1+1+1+1	0L, 1L, 2C, 4C, 0R, 1R, 3C, 5C
9	7	8	2+1+1+1+1+1+1	0L, 1C, 3C, 5C, 0R, 2C, 4C, 6C
10	8	8	1+1+1+1+1+1+1+1	0C, 2C, 4C, 6C, 1C, 3C, 5C, 7C
11	1	6	5.1	0L, 0C, 0Ls, 0R, 0LFE, 0Rs
12	2	6	4+2	0L, 0C, 1L, 0R, 0S, 1R
13	3	6	4+1+1	L, 0C, 1C, 0R, 0S, 2C
14	3	6	2+2+2	0L, 1L, 2L, 0R, 1R, 2R
15	4	6	2+2+1+1	0L, 1L, 2C, 0R, 1R, 3C
16	5	6	2+1+1+1+1	0L, 1C, 3C, 0R, 2C, 4C
17	6	6	1+1+1+1+1+1	0C, 2C, 4C, 1C, 3C, 5C
18	1	4	4	0L, 0C, 0R, 0S

19	2	4	2+2	0L, 1L, 0R, 1R
20	3	4	2+1+1	0L, 1C, 0R, 2C
21	4	4	1+1+1+1	0C, 2C, 1C, 3C
22	1	8	7.1	0L, 0C, 0Ls, 0Bsl, 0R, 0LFE, 0Rs, 0Bsr
23	1	8	7.1 screen	0L, 0C, 0Ls, 0Lc, 0R, 0LFE, 0Rs, 0Rc
24–63	Reserved	Reserved	Reserved	Reserved

NOTE Information on the **Audio codec**, **Sampling rate** and **Bit rate** is displayed.

Audio adjustment

Automatic level Select this checkbox if you wish to maintain the level at the level defined by the **Gain** parameter.

CAUTION: Audio level must be adjusted to a maximum of -20 dB (check volume level indicators) before activating the automatic audio level adjustment.

NOTE: This parameter has no effect if the audio stream is passed-through.

Gain Move the slider to amplify or attenuate the incoming signal.

Possible values: from -100 to +20 dB where 0 is the default gain.

NOTE: This parameter has no effect if the audio stream is passed-through.

Mute Check this option to mute the audio volume.

NOTE: This parameter has no effect if the audio stream is passed-through.

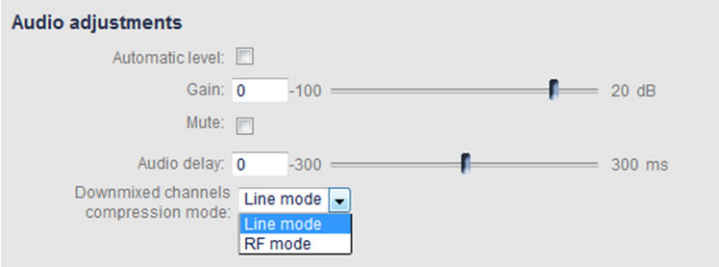
Audio delay Move the slider or enter a value to adjust the delay between the audio and the video input.

Possible values: from -300 to +300 ms.

Downmixed channels compression mode Specify how the downmix from 5.1 to stereo is done. Both 5.1 and stereo outputs can be managed separately in terms of audio levels.

Possible values: **Line mode** or **RF mode**

Figure 24. Audio adjustments: downmix



3. Repeat these steps to configure the second audio stream.

Audio levels

Left
Right

Display the left and right audio volume levels for the audio input.

NOTES: The volume indicators show the audio level after adjusting the volume with the **Gain** slider.

ASI input

Configuring the input

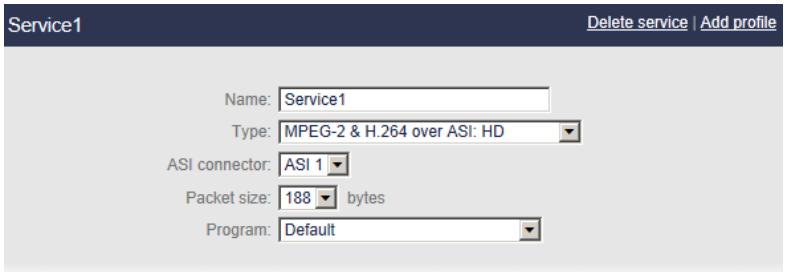
NOTES Both ASI Single Program Transport Stream (SPTS) and Multiple Program Transport Stream (MPTS) are supported.

For more information on ASI characteristics, see "ASI ingest" on page 252.

To configure the ASI input, follow these steps:

1. Click a **Service** tab. The **Service** parameters are displayed in the right panel.

Figure 25. Configuring the source



2. Select the appropriate value for each parameter.

Type Possible values (depending on license): **MPEG-2 over ASI: SD, MPEG-2 over ASI: HD, MPEG-2 over ASI: Audio only, MPEG-2 & H.264 over ASI: SD, MPEG-2 & H.264 over ASI: HD, MPEG-2 & H.264 over ASI: Audio only**.

ASI connector Select the appropriate input ASI connector.
Possible values: **ASI 1, ASI 2, ASI 3** or **ASI 4.**
See Installation Guide for more information.

Packet size Size of the TS streams packets to be ingested.
Possible values: **188** or **204** bytes.

Program Program number and name (MPTS parameter).

Auto selects the first available program.

To select the program, select the appropriate program in the drop-down list.

Configuring the video input settings

See "Configuring the video input settings" on page 52.

Configuring the audio input settings

See "Configuring the audio input settings" on page 55.

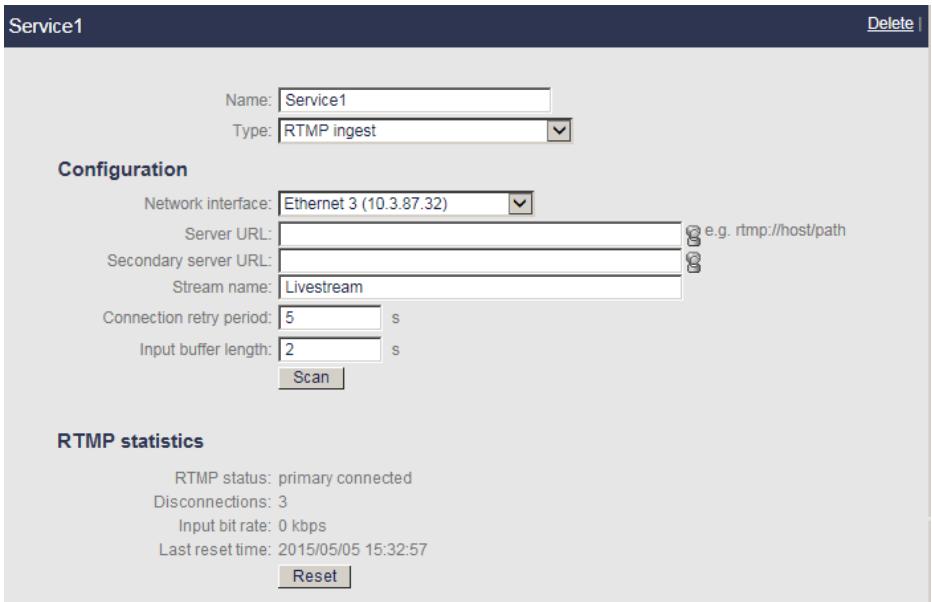
RTMP ingest

Configuring the input

To configure the RTMP ingest, follow these steps:

3. Click a **Service** tab. The **Service** parameters are displayed in the right panel.

Figure 26. Configuring the source



4. Select the appropriate value for each parameter.

Network interface

Select the appropriate input network interface.
Possible values: **Ethernet3, Ethernet4**, **Ethernet5**, **Ethernet6**, **Ethernet3 (backup: Ethernet4)** or **Ethernet6 (backup: Ethernet5)**
See Installation Guide for more information.

If you select **Ethernet3 (backup: Ethernet4)**, Ethernet 3 will be used as primary and backup will be done on Ethernet 4 if the audio or video PID is lost or if the audio or video input becomes encrypted.

When **Ethernet3 (backup: Ethernet4)** is selected, you can select the button to switch/toggle between Ethernet3 and Ethernet4 interfaces.

In case of input source failure, the encoder automatically switches to the secondary interface. If you want to go back to normal operation mode, you can manually force the active interface to be the primary one for this service.

NOTE: Network interface toggling is possible in both directions: from primary to backup, or from backup to primary.
Network interface toggling does not disable automatic switching.
After reboot, the active network interface is always the primary one.

Server URL

Enter the RTMP server primary URL.
Format: `rtmp://host/path`

Secondary server URL

Enter the RTMP server secondary URL

Stream name

Name of the RTMP stream.

Connection retry period

Specify the retry period in case of signal loss.
Possible values: from **1** to **10** seconds

Input buffer length

Define the data input buffer.
Possible values: from **1** to **10** seconds

RTMP statistics

In case of RTMP ingest, an **RTMP statistics** section is displayed. It indicates the network and transport errors since the last reset or scan.

NOTE You can click the **Reset** button to reset all the counters.

Configuring the video input settings

See "Configuring the video input settings" on page 52.

Configuring the audio input settings

See "Configuring the audio input settings" on page 55.

VC1 ingest

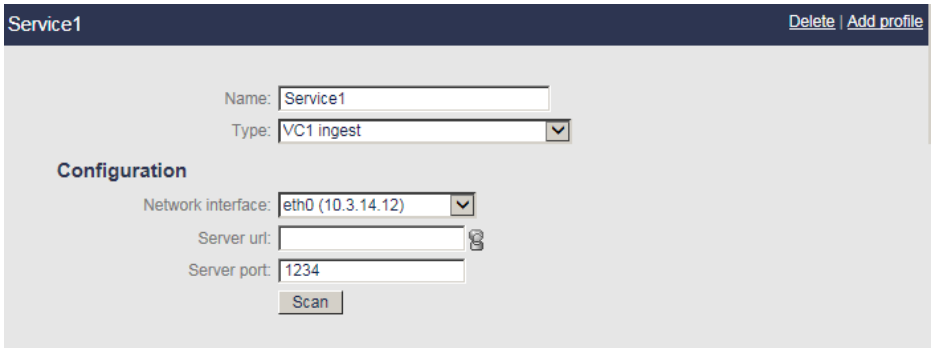
NOTE This input type is only available with Linux version.

Configuring the input

To configure the VC1 ingest, follow these steps:

5. Click a **Service** tab. The **Service** parameters are displayed in the right panel.

Figure 27. Configuring the source



6. Select the appropriate value for each parameter.

Network interface Select the appropriate input network interface. Possible values: **Ethernet3, Ethernet4**, **Ethernet5**, **Ethernet6**, **Ethernet3 (backup: Ethernet4)** or **Ethernet6 (backup: Ethernet5)**
See Installation Guide for more information.

If you select **Ethernet3 (backup: Ethernet4)**, Ethernet 3 will be used as primary and backup will be done on Ethernet 4 if the audio or video PID is lost or if the audio or video input becomes encrypted.

When **Ethernet3 (backup: Ethernet4)** is selected, you can select the button to switch/toggle between Ethernet3 and Ethernet4 interfaces.

Network interface In case of input source failure, the encoder automatically switches to the secondary interface. If you want to go back to normal operation mode, you can manually force the active interface to be the primary one for this service.

NOTE: Network interface toggling is possible in both directions: from primary to backup, or from backup to primary.
Network interface toggling does not disable automatic switching.
After reboot, the active network interface is always the primary one.

Server URL Enter the server URL and port.

Server Port Format: `IP_address` or `http://IP_address`

Configuring the video input settings

See "Configuring the video input settings" on page 52.

Configuring the audio input settings

See "Configuring the audio input settings" on page 55.

Chapter 5

Configuring an Internet TV profile

Basic workflow

Below is the basic workflow for generating an Internet TV stream.

Step #1: Set the service parameters

- Enter a service name,
- Select the input type used to capture the video stream:
  - If **On Board** input is selected, select the input type and the standard
  - If **Ethernet** input is selected, select the Ethernet interface, then enter the source IP address and port where your input source is broadcasting, and select the audio and video PIDs detected in the stream

Step #2: Set the profile parameters

- Enter a profile name
- Select the export type and the bit rate

Step #3: Set the profile video and audio parameters

- Select the audio and video input
- Select the audio and video codecs

Step #4: Set the profile output parameters

- Select the transport protocol, then set the output parameters

Step #5: Start the live broadcast session

Step #6: Display alarms (optional)

CAUTION When you modify a parameter and you press ENTER, the modification is automatically saved in the current configuration.

Setting up the encoding profiles

Creating a profile

By default only one profile by service is already activated and configured. To create a new profile, you can either add a new profile or duplicate an existing one.

Adding a new profile

To add a profile, click the **Service** tab, then select the **Add profile** button in the right panel.

MFVP Encoding live creates a new profile using default parameters.

Duplicating a profile

To duplicate a profile, follow these steps:

1. Click the **Profile** tab of the profile you want to duplicate, then select the **Duplicate** button in the right panel.

MFVP Encoding live creates a new profile using the selected profile parameters.

Deleting a profile

To delete a profile, click the **Profile** tab, then click the **Delete** button in the right panel. Note that all the profile parameters will be lost.

Setting the profile general parameters

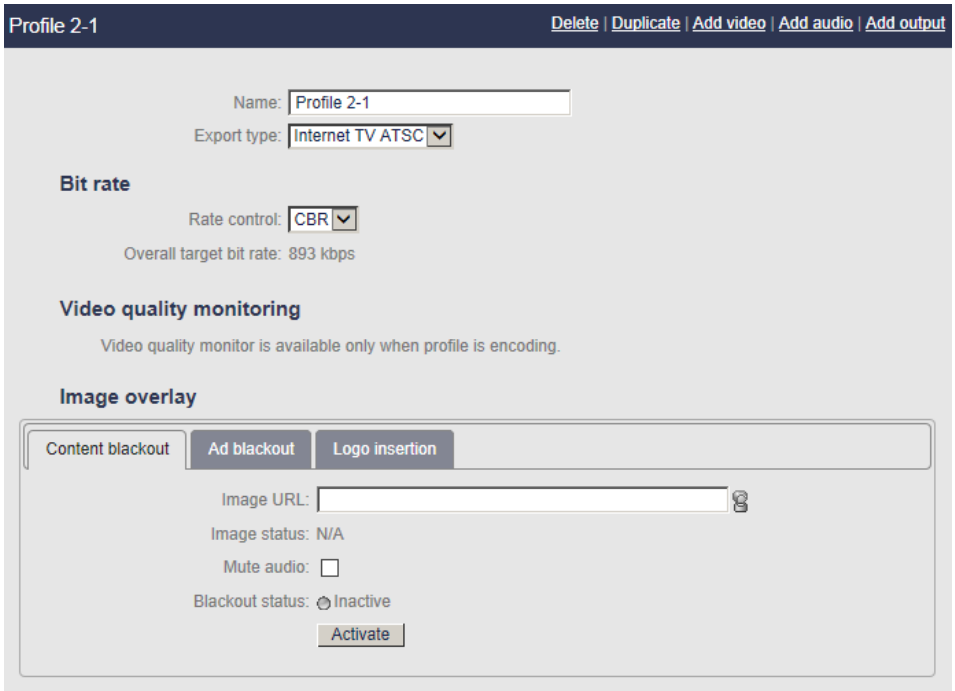
NOTES You can refer to *"Encoding recommendations" on page 245* for typical encoding settings.

During encoding, all the parameters are greyed and cannot be modified.

To set the profile general parameters, follow these steps:

1. Click a **Profile** tab. The profile general parameters are displayed in the right panel.

Figure 28. Profile general parameters: Internet TV ATSC export type



2. Select the appropriate value for each parameter.

Name Enter a profile name. We recommend using a name that clearly identifies the channel.

Export type Select an export type.

Possible values: **Internet TV ATSC, Internet TV DVB** or **ISMA**

NOTE: If a codec-linked restriction is not respected, an error message is displayed when you start encoding.

Bit rate control

For more information, *see Chapter 8 on page 205*.

Rate control Possible values: **CBR**, **VBR**, or **ABR**
In case of H.264, the stream remains compliant with the normative HRD.

- With **CBR** (Constant bit rate), you can stream content over a limited bit rate channel such as a network. The output stream fits in one bit rate, which you specify as a parameter. Constant bit rate means that the bit rate is constant according to the leaky bucket concept *(see "Appendix G" on page 266).*
- If you select **VBR** (Variable bit rate), MFVP Encoding live will use non constant bitrate inside a segment, but will keep average network bitrate constant (*See "Optimized Available Bit Rate mode (based on VBR)" on page 208*.

NOTES: Note that default GOP duration is automatically set to 10 000 ms (see *See "GOP duration" on page 102*)

- With **ABR** (Available Bit Rate), MFVP Encoding live will never exceed the maximum bit rate, but may go lower than the **Overall target bit rate** if no more bit rate is needed for encoding.

Overall target bit rate Informative field.

The overall target bit rate is automatically computed.

- For RTP, the **Overall target bit rate** is the resulting IP bit rate: it includes all overhead down to and including IP headers.
- For MPEG-2 TS output, the **Overall target bit rate** includes the resulting transport stream bit rate.

Overall max. bit rate *VBR mode only*.

Informative field.

SCTE 35 reference stream *Statmux mode only*.

You can automatically switch from CBR to VBR (and vice-versa) on SCTE-35 trigger by selecting the reference stream PID in the drop-down list; or manually by ticking the button (*See "Adding one or several profiles to the group" on page 214*).

Synchronization

You can synchronize two encoders geographically distant so that you can create a disaster recovery site with seamless switching. Both encoders output exactly the same streams.

If you check this option, you must complete the following information on both encoders:

Network interface Select the appropriate input network interface that will be used by both encoders to communicate.
Possible values: **Ethernet3, Ethernet4**, **Ethernet5**, **Ethernet6**, **Ethernet3 (backup: Ethernet4)** or **Ethernet6 (backup: Ethernet5)**
See Installation Guide for more information.

If you select **Ethernet3 (backup: Ethernet4)**, Ethernet 3 will be used as primary and backup will be done on Ethernet 4 if the audio or video PID is lost or if the audio or video input becomes encrypted.

When **Ethernet3 (backup: Ethernet4)** is selected, you can select the button to switch/toggle between Ethernet3 and Ethernet4 interfaces.

In case of input source failure, the encoder automatically switches to the secondary interface. If you want to go back to normal operation mode, you can manually force the active interface to be the primary one for this service.

NOTES: Network interface toggling is possible in both directions: from primary to backup, or from backup to primary.
Network interface toggling does not disable automatic switching.
After reboot, the active network interface is always the primary one.

Address Enter the multicast IP address from which the source will be received.

Examples: `239.2.3.123`

Port Enter the port number of the source; for example, `1234`.

Multicast source IP addresses

If you configured the IGMP protocol in **IGMPv3** (*see "Configuring IGMP" on page 37*), you can configure up to five multicast source IP addresses.

Enter a comma-separated list of multicast source IP addresses. Example: **`IP1, IP2, IP3`**.

Video quality monitoring

When encoding a profile, you can display a graphical representation of the video quality (see *"Video quality" on page 41*) by clicking the **video quality monitor** link.

Image overlay

Several types of image overlay management are provided by MFVP Encoding live:

- Content blackout
- Alternate content
- Logo insertion
- Automation

Content blackout

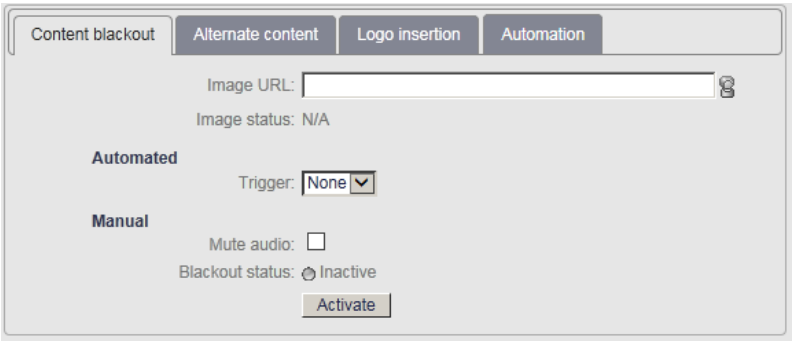
At any moment you can manually occult a program and replace the video by a default still image (PNG or JPEG file) that will be displayed instead of the output and mute the audio. You can also automatically replace the video by a still image on SCTE-35 or SCTE-104 flag by specifying the trigger PID.

NOTE Automated blackout on SCTE-35 or SCTE-104 flag is not applied on webdelivery triggers but only on SCTE-35/104 triggers from splices (webdelivery comes from a time signal).

The still image is stored on an external device and it can be retrieved by the encoder via HTTP (network file sharing).

NOTE If no default image is specified, a black frame will be displayed instead.

Figure 29. Content blackout parameters



To define the default image, follow these steps:

1. Enter the path to your custom image:
   http://<spath>/filename.png or file://<spath>/filename.png
2. Click outside the URL field.

NOTE If you defined credentials to access this URL (*see "Adding a credential" on page 235*), the user icon, should appear in color.

3. To mute the audio, check the **Mute audio** option.
4. For automated blackout on SCTE-35 or SCTE-104 flag, select the trigger PID in the drop-down list (Note that you will have to add the SCTE-35 or SCTE-104 stream in the output).
5. For manual blackout, click the **Activate** button to occult the current program or **Deactivate** to display the current video.

Alternate content

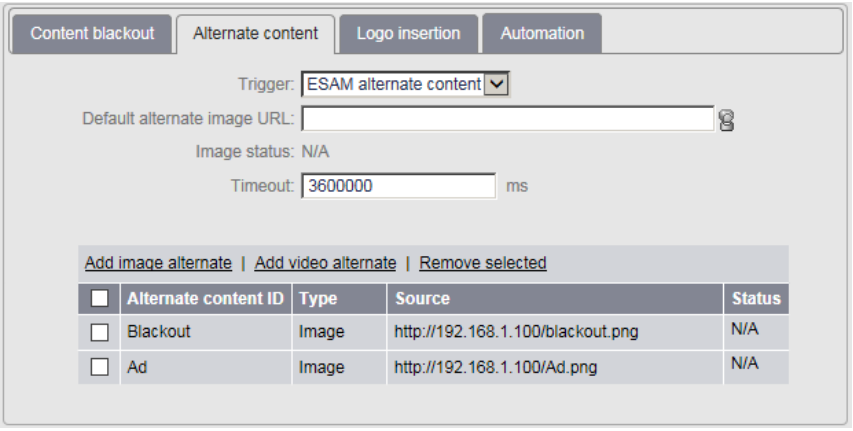
You can occult an advertisement and replace it by a default still image (PNG or JPEG file) that will be displayed instead of the output. The audio is automatically muted or replaced by the audio of the alternate video.

This alternate content is triggered by an out of band ESAM request that contains the ID of the alternate content (image or video).

The default still image is stored on an external device and it can be retrieved by the encoder via HTTP or CIFS (network file sharing).

NOTE If no default image is specified, a black frame will be displayed instead.

Figure 30. Alternate content parameters



Adding a default still image

To define the default image, follow these steps:

1. Select the trigger mode: **ESAM alternate content.**

   ESAM alternate content can be triggered by:

   – Out-of-band request containing alternate information. In this case, there is no need to interrogate a POIS server.

   – Out-of-band request, without alternate information. In this case, POIS server must be interrogated to get alternate information.

   – In-band signal. SCTE-35 does not contain alternate information. Thus a POIS must be interrogated to trigger an alternate switch.

NOTES Potential alternate sources are available in the profile private stream.

To setup the out of band interface, see *"ESAM out-of-band signal interface" on page 49*.

2. Enter the path to your custom image:

   http://<path>/filename.png
   file://<path>/filename.png

3. Click outside the URL field.

NOTE If you defined credentials to access this URL (*see "Adding a credential" on page 246*), the user icon, should appear in color.

4. Specify the **Timeout** (in milliseconds) that removes the occultation if no splice-in point comes. Possible values: from **1 000** to **99 999 999** ms.

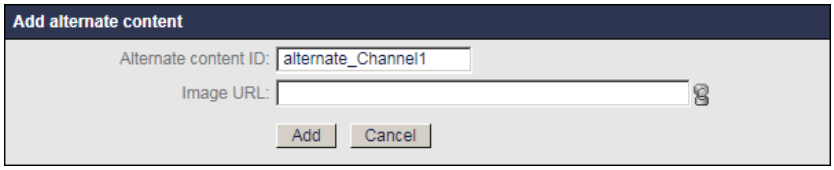
Adding an alternate image

The alternate image is stored on an external device and it can be retrieved by the encoder via HTTP or CIFS (network file sharing).

To add an alternate image, follow these steps:

1. Click the **Add image alternate** link to display the **Add alternate content** window:

Figure 31. Adding an alternate still image



2. Select the appropriate value for each parameter:

Alternate content ID Enter the ESAM ID.

Image URL Enter the path to your custom image:
http://<path>/filename.png
File://<path>/filename.png

NOTE: If you defined credentials to access this URL (*see "Adding a credential" on page 246*), the user icon, should appear in color.

3. Click the **Add** button to validate.

   The alternate video is added to the alternate contents table.

4. Repeat these steps for every alternate image you want to add.

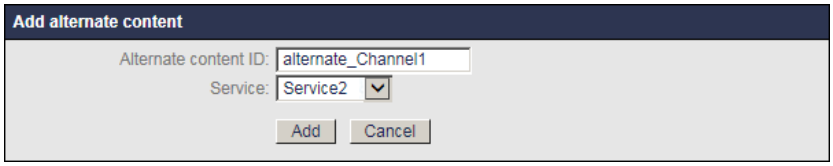
Adding an alternate video

To add an alternate video content, you prior have to create one or several alternate content service(s) (*see "Enable content switching service" on page 63*). These services will provide video source for alternate video content.

To add an alternate video follow these steps:

1. Click the **Add video alternate** link to display the **Add alternate content** window.

Figure 32. Adding an alternate video



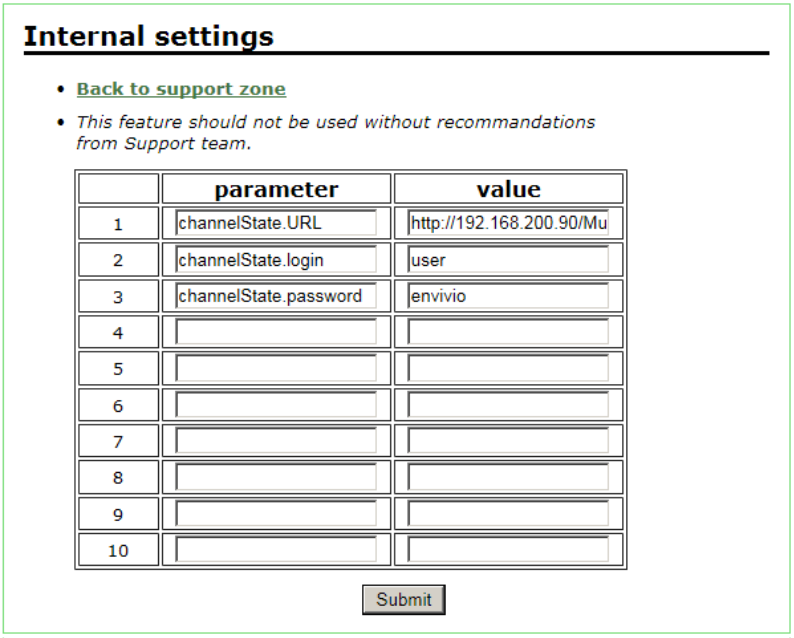
2. Enter the Alternate content ID
3. Select the alternate service that will be used as alternate video content.
4. Click the **Add** button to validate.
   The alternate video is added to the alternate contents table.
5. Repeat these steps for every alternate video content you want to add.

Activating blackout redundancy

To activate blackout redundancy, you need to configure specific parameters using advanced configuration settings.

To activate blackout redundancy, follow these steps:

1. Open a browser and enter the following address:
   `http://<MFVP-EncodingLive_IP_Address>/Support.html`
2. Click the **Advanced configurations** green link at the bottom of the page.
   The following page is displayed:



3. Enter the following parameters:

   channelState.URL=http://IP_address_of_the_Webdav_Server

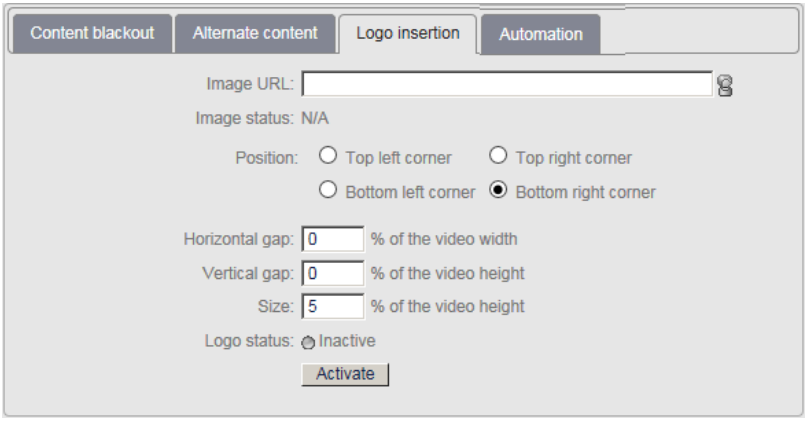
   channelState.login=login

   channelState.password=password

Logo insertion

Use this tab if you want to insert a picture (PNG or JPEG file) in the video output stream.

Figure 33. Logo insertion parameters



To insert a logo, follow these steps:

1. Enter the path to the image (`file://` or `http://Webserver/directory/filename.png` or `filename.jpg`), then click outside the URL field.
2. Select the position of the logo to be inserted (Top left corner, Top right corner, Bottom left corner, or Bottom right corner), the **horizontal/vertical gap** (percentage of the image width and height), and the **Size** (percentage of the image height)
3. Click the **Activate/Deactivate** button to respectively display/remove the logo in the output stream

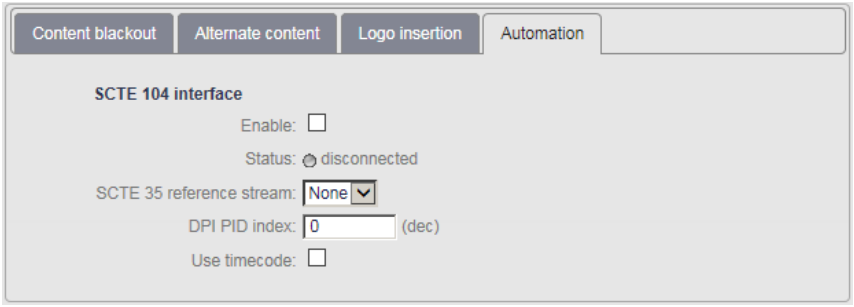
NOTE To remove the logo, delete the URL, then click outside the URL field. Note that you can only add or remove a logo when the profile is not encoding.
If you defined credentials to access this URL (*see "Adding a credential" on page 235*), the user icon, should appear in color.

Automation

NOTE Only available in SDI.

Use this tab if you want to insert a picture (PNG or JPEG file) in the video output stream on SCTE-104 or SCTE-35 trigger.

Figure 34. Logo insertion parameters



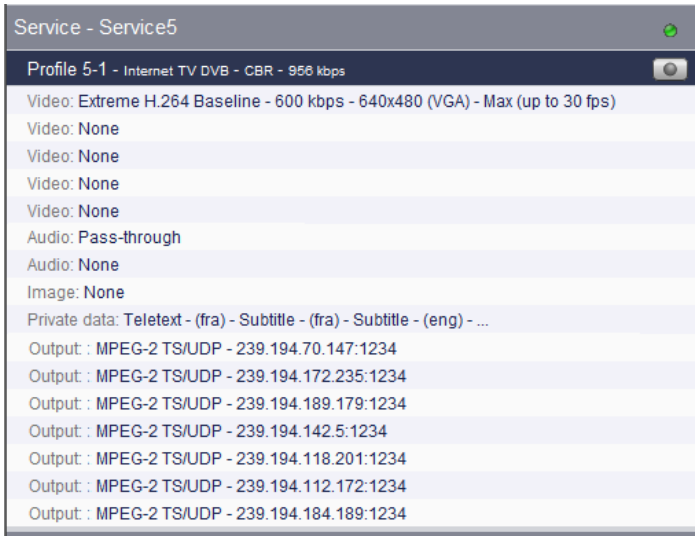
To activate automation, follow these steps:

1. Tick the **Enable** checkbox.
2. Select the SCTE-35 Reference stream.
3. Enter the DPI PID index to identify the service.
4. Tick the **Use timecode** checkbox to use the timecode (VITC) present in the SDI stream instead of UTC time to get more accurate timekeeping.

Displaying the audio, video, and output parameters

To display the video, audio, and output parameters of a profile, click the **Profile #** tab. The selected profile subtabs are displayed below.

Figure 35. Profile subtabs: Internet TV DVB export type



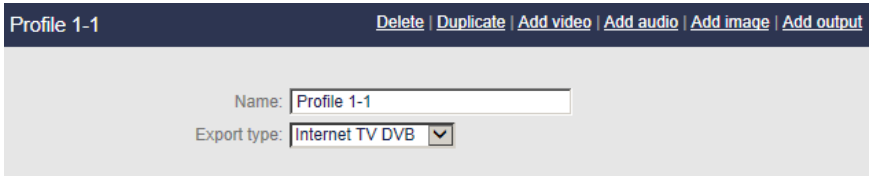
Setting the video stream parameters

NOTE Depending on license and export type, up to fifteen video streams are allowed per profile.

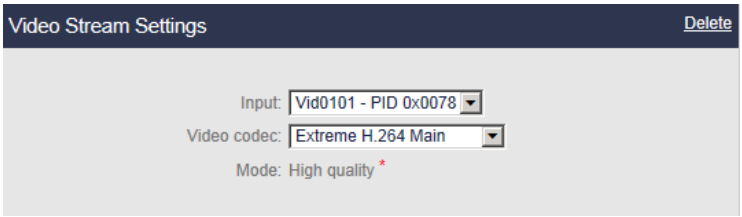
Adding or removing a video stream

With **Internet TV DVB** and **Internet TV ATSC** export types, you can add and remove the video streams.

To add a video stream, click the **Profile #** tab in the left panel, then click the **Add video** link in the top right corner. A video stream is added.



To delete a video stream, select the video stream in the left panel then click the **Delete** link in the top right corner. The video stream is deleted.

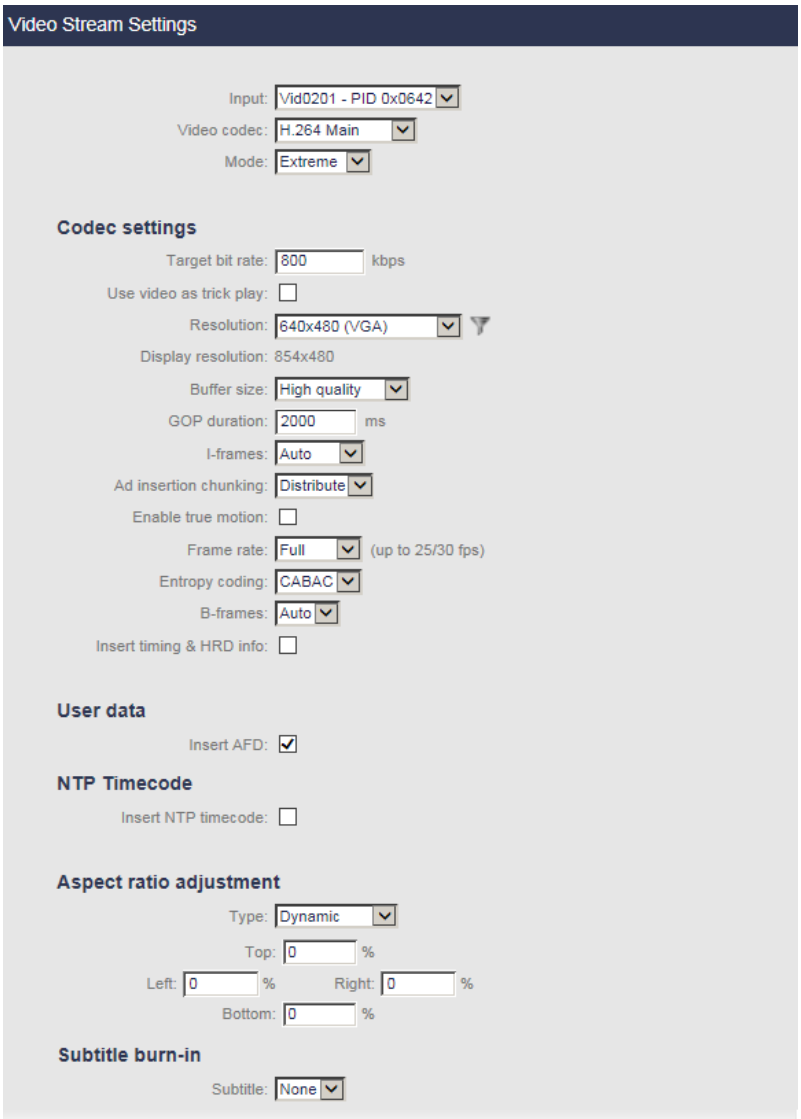


Configuring the video stream parameters

To display the video parameters, follow these steps:

1. Click the **Video** stream subtab, the video parameters are displayed in the right panel.

Figure 36. Profile video parameters: H.264 Main video codec



2. Select the appropriate value for each parameter.

NOTE With **InternetTV DVB** and **InternetTV ATSC** export types, **Mode**, and **Key frame period**, parameters are selected in the first video stream and displayed for information in the other video streams.

Input Select the appropriate video input channel

None means audio-only mode.

Video codec Select the video codec.

Possible values: depends on the export type. *See below.*

Table 2. Available video codecs per export type

Export type	Video codec
Internet TV ATSC **Internet TV DVB**	H.264 Baseline, H.264 Main, H.264 High, VC-1 Advanced; HEVC Main, HEVC Main 10
ISMA	H.264 Baseline, H.264 Main, H.264 High

Mode Possible values: **Standard, Extreme**, or **UP!**

You can adjust the encoding quality of the service (from standard to high quality)

Codec settings

Target bit rate Set the overall bandwidth target (in kbps) used by the stream.

Possible values: *see table below*.

NOTE: Video bit rate corresponds to "raw" video bit rate, and does not include bit rate overhead due to encapsulation (such as PES headers, MPEG-2 TS headers).

Max. bit rate *VBR mode only*.

Set the maximum overall bit rate.

Possible Values: from **800** to **10 000** kbps

Table 3. Target bit rate per export type

Export type	Video bit rate
Internet TV ATSC **Internet TV DVB**	From 20 to 10 000 kbps
ISMA	From 20 to 5 000 kbps

Use video as trick play

Only available with H.264 video codec.

Check this option if you want to create a low frame rate, IDR-only stream that can be used for trick play.

If you check this option, only few video parameters are displayed: **Resolution**, **Display resolution** and **Frame rate** (you can specify a custom value).

To be used as trick play, encoding is IDR-only and EBP signaling is added on all frames (if checked in the MPEG-2 TS configuration).

Coding mode

Only available with H.264 High/Main. Not available with 720p input.

Possible values: **Progressive** or **Interlace**

Resolutions available for **Interlace** mode:

1920x1080i25 or PAL input:

- 1920x1080; 1440x1080; 1280x1080; 960x1080; 720x576; 704x576; 544x576; 528x576; 480x576; 352x576

1920x1080i30 or NTSC input:

- 1920x1080; 1440x1080; 1280x1080; 960x1080 720x480; 704x480; 640x480; 544x480; 528x480; 480x480; 352x480

Resolution

Select a resolution value.

NOTE: Resolutions are sorted by the total number of pixels per frame.

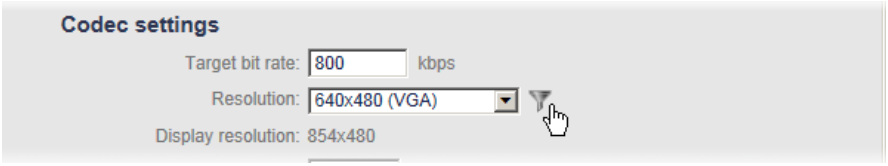
Filtering the resolutions

You can use a resolution filter based on specific parameters (category, aspect ratio) to shorten the resolution list.

To open the filtering options, follow these steps:

1. Click the filter icon, next to the resolution list.

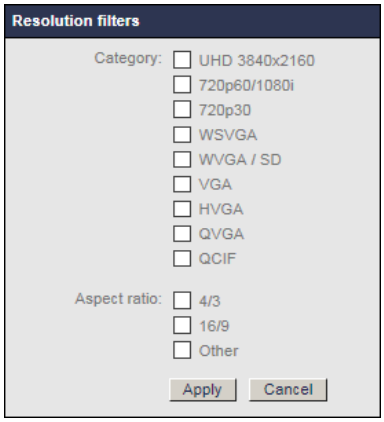
Figure 37. Filtering resolutions



2. Select category or/and aspect ratio filters to be applied on the list.

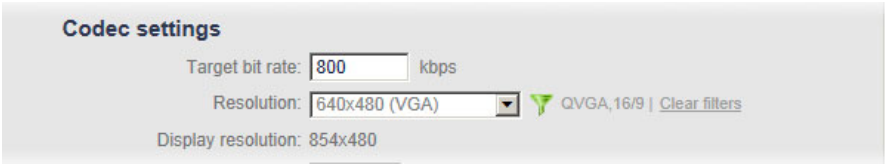
- Filter by category: **QCIF, QVGA, HVGA, VGA, WVGA/SD, WSVGA, 720p30, 720p60/1080i, UHD 3840x2160**
- Filter by aspect ratio: **4/3, 16/9, Other**

Figure 38. Filters



3. Click the **Apply** button to display the filtered resolutions.

Figure 39. Filtering resolutions



When one or several filters are applied:

- The filter icon turns to green, : you can still click the filter icon to open the filtering window.
- The available filter(s) is(are) displayed.

NOTE If the current resolution does not belong to the filtered list, the value will still be in the list, but appears in grey.

4. You can click the **Clear filters** link to delete the filter.

Display resolution Information field. Depends on the aspect ratio management and on the encoding resolution.

Buffer size Possible values: **Ultra low delay**(1)**, Low delay, Standard, High quality** or **Custom**.

- **Ultra low delay:** allows reducing the end-to-end delay inside the Silverlight smooth streaming ecosystem to
  1-2 second(s)
- **Low delay:** VBV size = 1 000 ms
- **Standard**: VBV size = 2 000 ms
- **High quality**: VBV size = 5 000 ms
- **Custom**: enter a value between 500 and 10 000 ms
  Defines the size of the VBV (Video Buffering Verifier). If the set-top box is not performing correctly (desynchronization), you can adjust the delay settings.

GOP duration Possible values: from **1 000** to **10 000** ms

Enter the maximum time (in ms) between two key frames. You can specify how often you want key frames to be inserted into the video stream. Key frames are inserted into the stream periodically to synchronize the decoder and enable it to recover from errors.

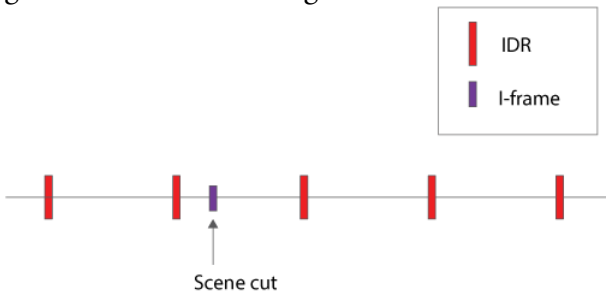
Refreshing the image more often (by setting a shorter key frame period) reduces the recovery time but requires a higher bit rate to maintain encoding quality.

NOTE: The Key frame period value defines respectively the fragment or the segment size in **Smooth Streaming** and **HTTP Live Streaming**.

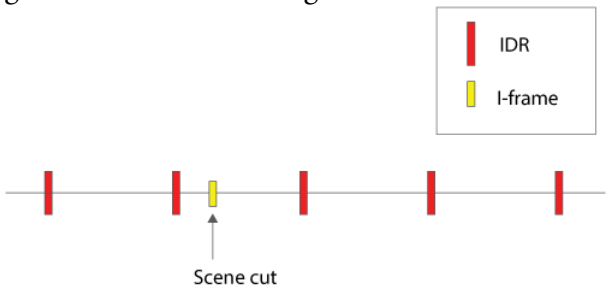
I-frames I-frames may be inserted to ensure short zapping time.

Possible values: **Auto**, **Off** or **Periodic**

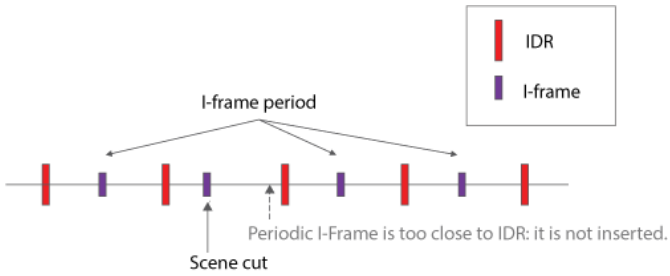
Auto: GOP boundaries are encoded as IDR frames. I-frames are generated on scene changes.



Off: GOP boundaries are encoded as IDR frames. P-frames are generated on scene changes.



Periodic: An I-frame is inserted every **I-frame period** inside a GOP. I-frame periods are reset either on scene changes or on IDR. If the next I-Frame is too close to the IDR or to the scene change I-frame, it will not be inserted.



I-frame period *Only available with Periodic I-frames.*

Enter the maximum time (in ms) between two I-frames.

Possible values: from **300** ms to the value of key frame period.

Ad insertion chunking

You can select the coding strategies in case of ad insertion trigger. This will impact the GOP structure.

Possible values: **Distribute** or **Merge**.

If you select **Distribute**, the current segmentation is reset in case of splice point, preventing too short segments by creating two segments of the same size.

If you select **Merge**, the segmentation is not reset on splice point. This strategy prevents from small chunks by merging them with their neighbor if they are less than 50% of the target duration.

Enable true motion

Only available with Progressive Coding mode (see page 100)

Select this checkbox if you want to increase the frame rate (up to 50/60 fps). See **Frame rate** parameter below.

This option enhances the visual quality, especially with fast-moving sport contents or with scrolling text. It increases the sharpness of the images and the smoothness of the motion.

NOTES: For optimal quality with telecine contents, we recommend not selecting the **Adaptive Inverse Telecine** option (see *"**Adaptive inverse telecine**" on page 54*). If the **Adaptive Inverse Telecine** is selected, frame rate may vary from 59.94 fps to 23.98 fps.

CAUTION: **True motion** is implemented at the service level to reduce processing cost. When it is enabled it applies to all videos belonging to the related service.
If you want to disable the true motion for one specific video we recommend setting the **Frame rate** to **Half**, **Quarter** or **Custom** to disable true Motion.

CAUTION: If you activate true motion, do not set the first video input stream to **None**.

Frame rate

Possible values: **Full** (up to 25/30 fps; or up to 50/60 fps if you ticked the **True motion** checkbox), **Half**, **Quarter, Eighth** or **Custom**

If you select **Custom,** enter a specific value from **5** to **25** fps (PAL) or from **5** to **29.97** fps (NTSC).

NOTE: If you select **Custom**, the GOPs are not synchronized and the IDR is not aligned between streams.

Entropy coding *Not available with Extreme H.264 Baseline video codecs.*

Possible values: **CABAC or CAVLC**.

CABAC means Context Adaptative Binary Arithmetic Coding. This is an entropy lossless compression algorithm, but very time-consuming, especially at the decoder side.

CAVLC means Context Adaptative Variable Length Coding. This is the other entropy lossless compression algorithm used in the H.264 format. It is less efficient than CABAC, but faster (especially at the decoder side).

B-frames *Not available with Premium and Extreme H.264 Baseline video codecs.*

Possible values: **Off**, **Auto**, **2** or **3.**

B-frames are used to increase quality. It indicates whether bidirectional encoded pictures (B-frames) are used.

A B-frame can be considered as an interpolated frame between reference frames (I or P), interpolation is done using forward and backward motion vectors used to select pieces of pictures in reference frames.

Insert timing & HRD info Timing info is an optional parameter in the Video Usability Information layer (VUI) that indicates the frame rate. The hypothetical reference decoder (HRD) enables to verify whether coded bitstreams conform to the requirements of H.264/AVC.

1. Only available with Internet TV ATSC and Internet TV DVB

Specific Internet ISMA, TV DVB and Internet TV ATSC parameters

Use hierarchical P *Only available with Premium or Extreme H.264 Baseline video codecs.*

This feature improves the video quality of terminal devices especially for half to full frame rate and from medium to high bit rate. Outside these settings quality remains unchanged.

This uses a new GOP structure with several temporal levels which has a better coding efficiency than a classical P structure.

NOTE: This option may not be supported by the device player and must be deactivated in case of non fluent video.

NTP Timecode

Insert NTP timecode Check this option if you want to add timecodes in the H.264 output stream. These timecodes are added in the SEI information of the stream. They are following this format: hh:mm:ss:ff. The timecodes are based on the NTP time at the start of the first profile. Values of the timecodes are incremented on all the following images.

User data

NOTE Only available with **Internet TV DVB** and **Internet TV ATSC** export type.

Use closed captions *Only available if Export type is ATSC.*
Check this option if you want to extract and embed the closed caption or SCTE20 information in the output stream.

NOTE: Compliant with the EIA-608 standard.

Insert AFD Check this option if you want to take into account the display aspect ratio of the source and provide this information to the output.

Aspect ratio management

See Appendix D for more details.

Type Select the aspect ratio adaptation type.

Possible values: **Dynamic**[1], **Letter Boxing** or **Stretching**.

NOTE: We recommend launching the encoding, and modify the **Aspect ratio adjustment** value to adjust it with accuracy.

NOTE: For better compatibility with HLS and HSS players, we recommend setting the aspect ratio adaptation type to **Letter Boxing** when upscaling from SD to 720p.

1. Internet TV DVB and Internet TV ATSC export types only.

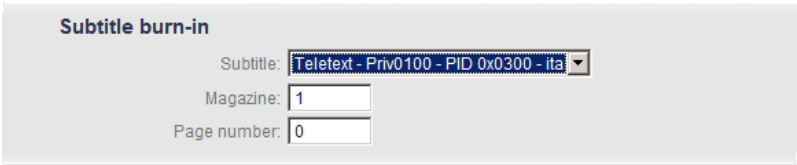
Output aspect ratio

- If you select **Dynamic**, you can define the Top, Bottom, Right and Left values (in%).
- If you select **Letter boxing**, you can define **Output aspect ratio** (square pixel, 4/3 or 16/9) and the **Zoom level** (in%).
  Possible values for **Zoom level**: 0 (no zoom, black stripes on the top and bottom), 10, 20, 30, 40, 50, 60, 70, 80, 90, 100 (the video is zoomed, and black stripes are removed).
- If you select **Stretching**, you can define the **Output aspect ratio** (square pixel, 4/3 or 16/9) and the stretching values (in%) for Top, Bottom, Right and Left.

Subtitle burn-in

The subtitle burn-in feature enables burning DVB teletext subtitle, DVB subtitle or Closed Caption streams in the video.

Figure 40. Subtitle burn-in: Teletext



1. Select the stream you want to burn.

NOTE By default the **Subtitle** parameter is set to **None**.

2. In case of **Teletext**, specify the **Magazine** and the **Page number**.

Setting the audio stream parameters

NOTES Depending on license and export type, up to six audio streams are allowed per profile.

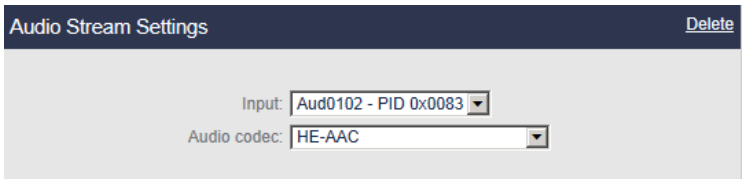
With **Internet TV ATSC Internet TV DVB** and **HTTP Live Streaming** output, the third audio is reserved for audio-only HTTP live streaming output.

Adding or removing an audio stream

With **Internet TV DVB** and **Internet TV ATSC** export types, you can add and remove the audio streams.

To add an audio stream, click the **Profile #** tab in the left panel, then click the **Add audio** link in the top right corner. An audio stream is added.

To delete an audio stream, select the video stream in the left panel then click the **Delete** link in the top right corner. The audio stream is deleted.

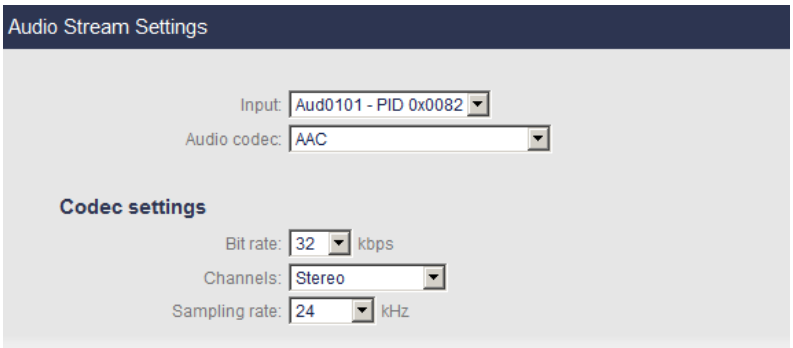


Configuring the audio stream parameters

To display the audio parameters, follow these steps:

1. Click the **Audio** stream subtab, the audio parameters are displayed in the right panel.

Figure 41. Audio parameters: AAC audio codec



2. Select the appropriate value for each parameter.

Input Select the appropriate audio input channel.
None means video-only mode.

Audio codec Specify the audio codec.
Possible values: *see Table 4 below*.

Table 4. Available audio codecs per export type

Export type	Audio codecs
Internet TV ATSC **Internet TV DVB**	MPEG-1 Layer II, AAC, HE-AAC, HE-AAC v2, Pass-through[1], or Windows Media Audio 9.2, Windows Media Audio 10 Prof. (*only available with Smooth streaming output*), Dolby Digital Plus, Dolby Digital
ISMA	AAC, HE-AAC, HE-AAC v2

1. Pass-through is not available when audio input is AAC.

Codec settings

NOTE These parameters are not available with Pass-through.
For Dolby Digital Plus parameters, see *"Specific Dolby parameters" on page 113*.

Bit rate Specify the audio bit rate value in kilobits per second.
See Table 5 below.

NOTE: Audio bit rate corresponds to "raw" audio bit rate, and does not include bit rate overhead due to encapsulation (such LATM or ADTS headers, PES headers, MPEG-2 TS headers).

Channels Possible values: *see table below*.

Sampling rate Specify the sampling rate in kilohertz (kHz).
See table below.

NOTE Depending on the selected codec, the channel mode or the sampling rate can be disabled because they are not applicable. The following tables summarize this behavior.

Table 5. Bit rates depending on codec

Codec type	Bit rate
MPEG-1 Layer II	32, 48, 56, 64, 80, 96, 112, 128, 160, 192, 224, 256, 320, 384
AAC	8, 10, 12, 16, 20, 24, 28, 32, 40, 48, 56, 64, 80, 96, 112, 128, 160, 192, 224, 256, 320
HE-AAC	8, 10, 12, 16, 20, 24, 28, 32, 40, 48, 56, 64, 80, 96, 112, 128
HE-AAC v2	16, 20, 24, 28, 32, 40, 48
WMA Audio 9.2	32, 40, 48, 64, 80, 96, 128, 160, 192, 256, 320
WMA Audio 10 Prof.	32, 48, 64, 80, 96, 128, 160, 192, 256

Table 6. Channel modes depending on bit rate and codec

Codec type	Bit rate	Channel mode
AAC	32 to 160	Mono downmix/left/right, Dual-mono, Stereo
	192 and over	Dual-mono, Stereo
HE-AAC	32 to 64	Mono downmix/left/right, Dual-mono, Stereo
	80 and over	Dual-mono, Stereo
HE-AAC v2	All bit rates	Stereo

Table 7. Channel modes depending on bit rate and codec

Codec type	Bit rate	Channel mode
WMA Audio 9.2	32	Mono downmix, Stereo
	40 and over	Stereo
WMA Audio 10 Prof.	All bit rates	Stereo

Table 8. Sampling rate depending on codec and bit rate

Codec	Bitrate (kbps)	Channel mode	Sampling rates (kHz)
AAC	**32**	**Mono**	11.025, 12, 16, 22.05, 24, 32, 44.1, 48
		Dual-mono, Stereo	11.025, 12, 16, 22.05, 24
	40	**Mono**	16, 22.05, 24, 32, 44.1, 48
		Dual-mono, Stereo	16, 22.05, 24, 32
	48	**Mono**	22.05, 24, 32, 44.1, 48

Codec	Bitrate (kbps)	Channel mode	Sampling rates (kHz)
		Dual-mono, Stereo	22.05, 24, 32
	56	All channel modes	22.05, 24, 32, 44.1, 48
	64 and over	All channel modes	32, 44.1, 48
HE-AAC	All bit rates	All channel modes	32, 44.1, 48
HE-AAC v2	All bit rates	Stereo	32, 44.1, 48
WMA 9.2	32	Stereo	22.05, 32, 44.1
		Mono	44.1
	40	Mono, Stereo	32
	48	Stereo	32, 44.1
		Mono	44.1
	64	Mono, Stereo	44.1, 48
	80	Mono, Stereo	44.1
	96 to 192	Mono, Stereo	44.1, 48
	Over 192	Mono, Stereo	44.1
WMA 10 Prof.	32	Stereo	32
	Over 48	Stereo	44.1, 48

Specific Dolby parameters

Bit rate Specify the audio bit rate value in kilobits per second.
Possible values: from **96** to **1024 kbps**

NOTE: Minimum bitrate for surround is **160 kbps**

Channels Select the number of output channels.

Possible values: **Fixed Output 2/0** (stereo) or **Follow input** (maintains the number of channels).

NOTE: **Follow input** is only available if bit rate is higher than 128 kbps.

Sampling rate Specify the sampling rate in kilohertz (kHz).

NOTE: Only 48kHz is supported.

LFE Enable *Only available with Follow input.*

Check this option to activate the LFE (Low Frequency Effects) filter (*see "LFE Lowpass Filter" on page 114*).

Dialog Normalization Possible values: **Auto** or, from **-1 dB** to **-31 dB**

Auto means that no processing will be done on the input.

Informational metadata for Fixed Output option

Dolby Surround EX Mode Possible values: **Auto**, **Not Indicated**, **NOT Dolby Surround EX Encoded**, or **Dolby Surround EX Encoded**

Preprocessing parameters for Fixed Output option

NOTE **Auto** means that no processing will be done on the input.

DC Highpass Filter Check this option to activate a DC highpass filter.

DRC Line Mode Profile Possible values: **None**, **Film: Standard, Film: Light, Music: Standard, Music: Light, Speech**

Enables Dynamic Range Control that restricts the absolute peak level for a signal.

DRC RF Mode Profile Possible values: **None**, **Film: Standard, Film: Light, Music: Standard, Music: Light, Speech**

Enables heavy Dynamic Range Control, ensures that the instantaneous signal peaks do not exceed specified levels.

Informational metadata for Follow input option

Dolby Surround EX Mode Possible values: **Auto**, **Not Indicated**, **NOT Dolby Surround EX Encoded**, or **Dolby Surround EX Encoded**

Preprocessing parameters for Follow input option

DC Highpass Filter Check this option to activate a DC highpass filter.

LFE Lowpass Filter Applies a 120Hz lowpass filter to the LFE (Low Frequency Effects) channel prior to encoding.

90 Degree Phase Shift Applies a 90-degree phase shift to the surround channels.

3 dB Attenuation Applies a 3 dB attenuation to the surround channels.

DRC Line Mode Profile Possible values: **Film: Standard, Film: Light, Music: Standard, Music: Light, Speech**

Enables Dynamic Range Compression that restricts the absolute peak level for a signal.

DRC RF Mode Profile Possible values: **Film: Standard, Film: Light, Music: Standard, Music: Light, Speech**

Enables heavy Dynamic Range Compression, ensures that the instantaneous signal peaks do not exceed specified levels.

Downmix Metadata for Follow input option

NOTE **Auto** means that no processing will be done on the input.

Lo/Ro Center Mix Level This parameter indicates the level shift applied to the center channel as a result of downmixing to an Lo/Ro output.

Lo/Ro Surround Mix Level
This parameter indicates the level shift applied to the surround channels when downmixing to an Lo/Ro output.

Lt/Rt Center Mix Level
This parameter indicates the level shift applied to the center channel as a result of downmixing to an Lt/Rt output.

Lt/Rt Surround Mix Level
This parameter indicates the level shift applied to the surround channels when downmixing to an Lt/Rt output.

Stereo Downmix Preference
Select the stereo downmix preference.

Possible values: **Not Indicated**, **Lt/Rt Downmix preferred**, **Lo/Ro Downmix preferred** or **Dolby Prologic II downmix preferred**

Setting the image stream parameters

You can add an image stream needed by HLS (audio-only with still-image).

NOTE This option is only available with MPEG-2 TS or HLS output.

Adding or removing an image stream

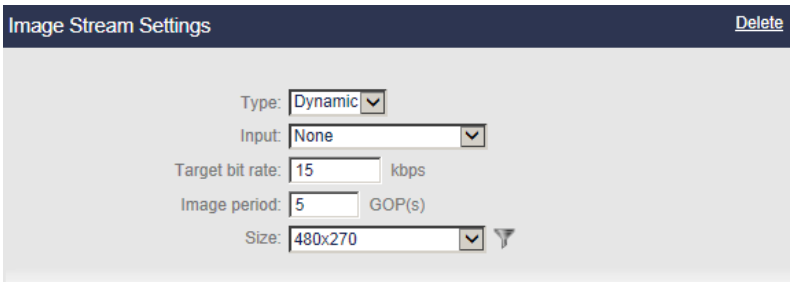
You can add up to four image streams per profile.

To add an image stream, click the **Profile #** tab in the left panel, then click the **Add image** link in the top right corner. An image stream is added.

Figure 42. Adding an image

To delete an image stream, select the image stream in the left panel then click the **Delete** link in the top right corner. The image stream is deleted.

Figure 43. Deleting an image

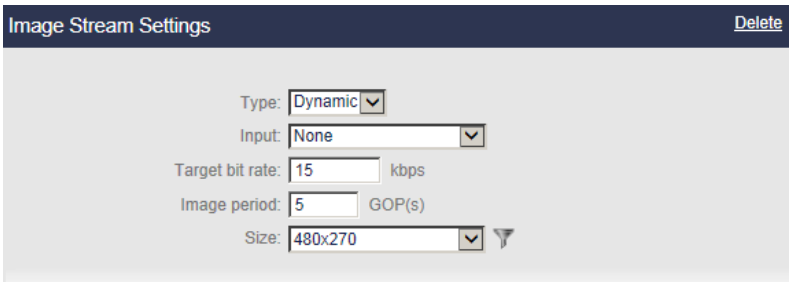


Configuring the image stream parameters

To display the image parameters, follow these steps:

1. From the **Profile #** tab in the left panel click the **Image** stream subtab. The image parameters are displayed in the right panel.

Figure 44. Image stream parameters



2. Select the appropriate value for each parameter.

Type This image can be either **None**, **static** or **dynamic**.

- If you select **static**, enter the path (**URL**) to the image then click outside the URL field. Path can be:
  - `file://<spath>/filename.png`
  - `ftp://<spath>/filename.png`
  - `sftp://<spath>/filename.png`
  - `http://<spath>/filename.png`
  - `https://<spath>/filename.png`

NOTE: If you defined credentials to access this URL (*see "Adding a credential" on page 235*), the user icon, should appear in color.

- If you select **dynamic**, select the appropriate video input channel (**Input**). An image will be regularly extracted from the video input.

Target bit rate Specify the target bit rate in kbps.

Value must be less than or equal to 2000.

Image period Specify the number of GOPs between two images.

Values must be less than or equal to 10.

Size Select the target resolution.

Possible values: from 80x64 to1920x1080.

3. Once you have defined the image, you can add it to the output (*see "Specific MPEG-2 TS/RTP and MPEG-2 TS/UDP parameters" on page 130* and *"Specific HTTP Live Streaming parameters" on page 134*.)

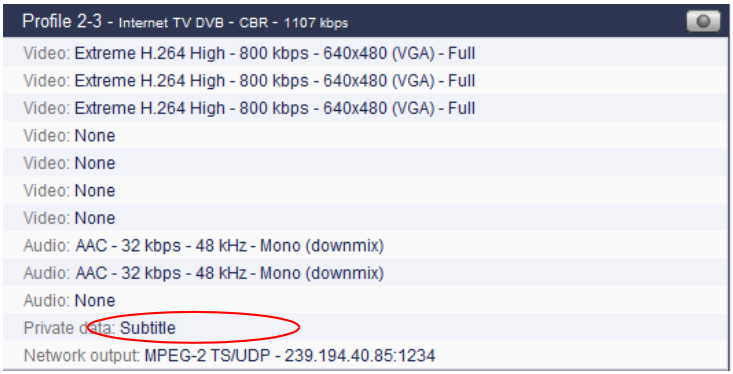
Setting the private stream parameters

NOTE Private data is only available if **Export type** is **Internet TV DVB** or **Internet TV ATSC**.

To display the private stream parameters, follow these steps:

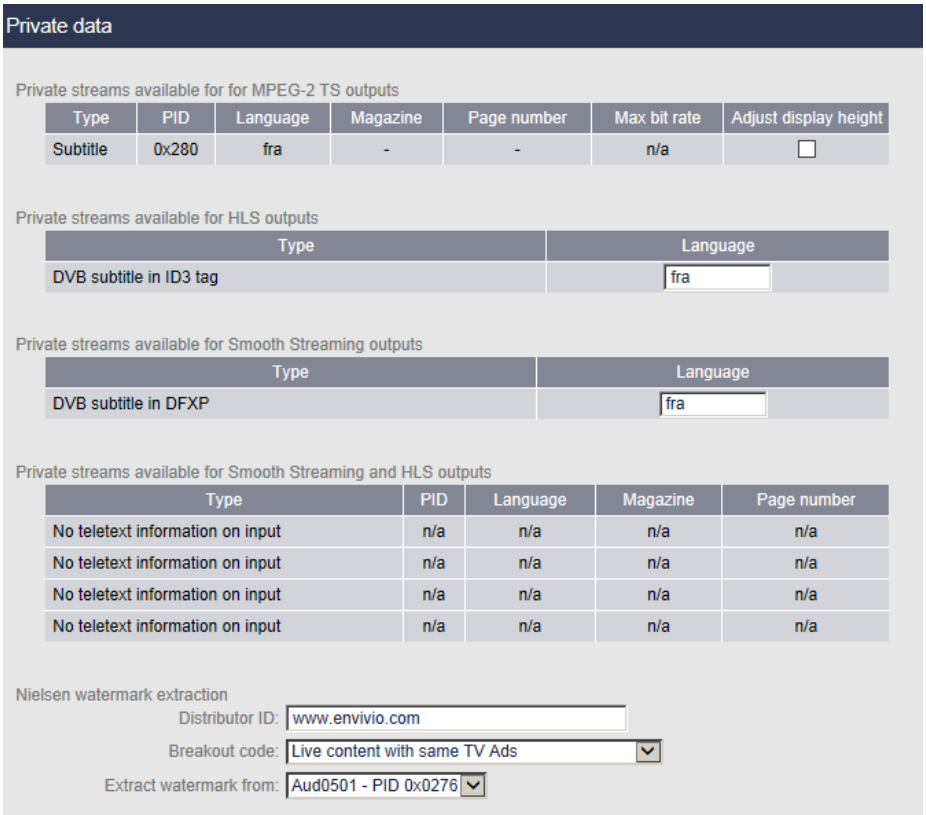
1. Set the **Export type** to **Internet TV DVB**. The **Private data** subtab is added to the profile subtabs.

Figure 45. Private data subtab: Internet TV DVB



2. Click the **Private data** subtab, the private stream parameters are displayed in the right panel.

Figure 46. Private data: Internet TV DVB



3. The following information is displayed for each private stream:

Type Possible values for Ethernet: **Teletext**[1], **Subtitle**, **VBI teletext, SCTE 35, SCTE 35 Passthrough, SCTE 27, Closed captions**[2], **EBIF, EIS, AIT**
Possible values for On Board: **Teletext**, **Subtitle**

1. See *"Subtitle burn-in" on page 107* for more information on subtitle burn-in feature.
2. Uses DFXP, an XML format for subtitling. Used for Smooth Streaming output.

PID PID of the input private stream. The management depends on the selected mode (**Ethernet**/**On board**).

NOTE: If you change the VBI teletext PID value, go to the **Output>Streams** submenu and click the **Edit** link to edit the private stream PID value.

Language In **Ethernet** and in **ASI** modes, the input language is displayed.
In **On Board** mode, the default is eng.

Magazine
Page number Magazine/page number.
For example, for DVB teletext 577, magazine number is 5 and page number is 77.

Max bit rate *Only available with teletext.*
You can define the bandwidth reserved for the teletext DVB data.
Possible values: from **30** to **600** kbps

Adjust display height *Only displayed with Subtitle.*
Check this option to adjust the DVB subtitle display height in case of SCTE 27.

NOTES In **Ethernet** and in **ASI** mode, the input PID is displayed but you can change its value. Teletext and Subtitle information is extracted from the input.

In **On board** mode, only one Teletext and two additional Subtitles are displayed. You cannot add additional information.

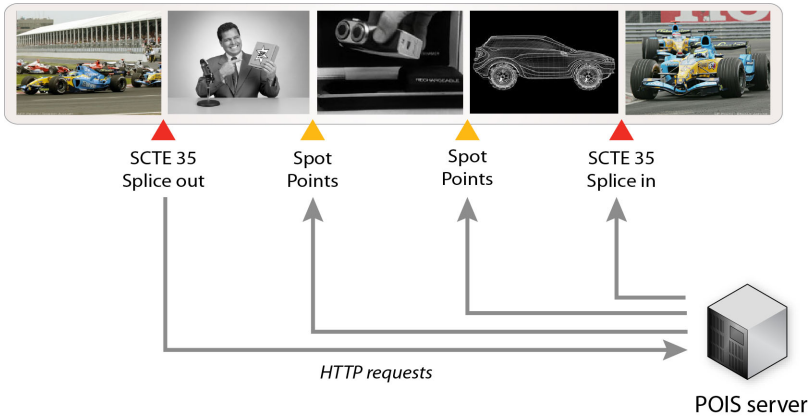
SCTE 35 private streams

For a given source (service), you can associate a POIS server to an SCTE 35 stream for splicing information validation. Validated information is then used by the encoder to perform segmentation for the different export types and update the SCTE 35 information.

How does SCTE 35 validation work?

When receiving an SCTE 35 splice out trigger, MFVP Encoding live addresses an HTTP request to the POIS server.

Figure 47. SCTE 35 validation process

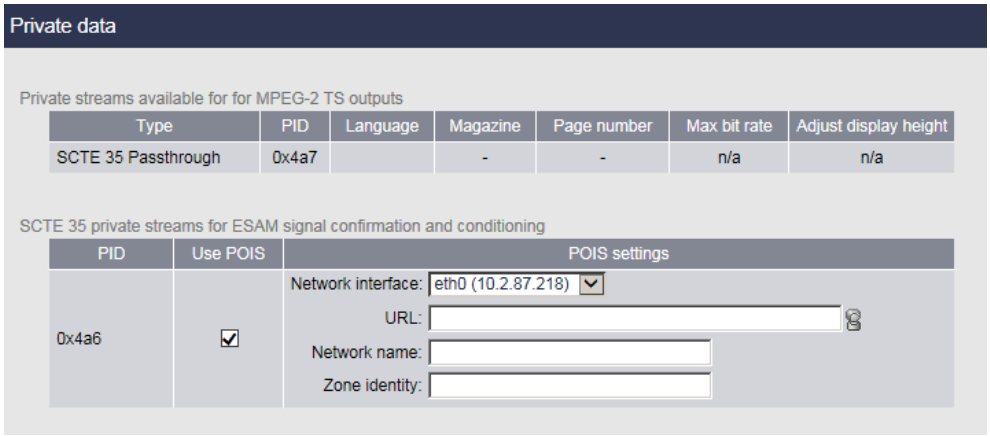


In its answer, the POIS server indicates the global ad break duration. It may also define one or more spot point(s) which correspond(s) to individual advertisement duration.

As the splice duration is overridden by the POIS server, every incoming splice-in is discarded and replaced by a splice-in that corresponds to the new duration. Moreover, in case of auto-return splice-out, it is converted into a classical splice-out and a splice-in added at returned point.

NOTE You can have both the Passthrough and the signal modified by POIS in the same output choosing two different PIDs.

Figure 48. Private data: SCTE 35



To activate this feature, follow these steps:

1. Go to the **SCTE 35 private streams for ESAM signal confirmation and conditioning** section.
2. In the table, tick the **Use POIS** checkbox to enable the POIS server, then select the **Network interface**, and specify the POIS server **URL**, the **Network name** and the **Zone identity**.

NOTE If you defined credentials to access this URL (*see "Adding a credential" on page 235*), the user icon, should appear in color.

SCTE 27 private streams

This feature lets MFVP Encoding live convert the SCTE-27 subtitles to DVB-Bitmap subtitle or/and burn them on the video.

SCTE-27 subtitles can be output in Open caption (burning) or DVB subtitles, and can also be passed-through.

Nielsen private streams

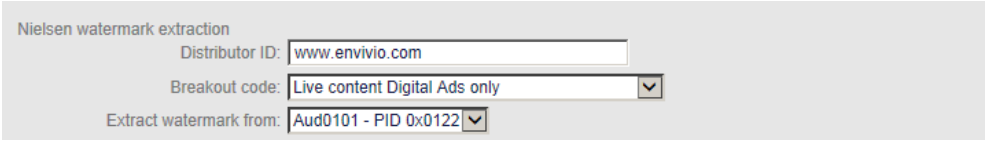
NOTE Nielsen ID3 ratings for the audience measurement systems is only available if **Export type** is **Internet TV DVB, Internet TV ATSC** and if **Output type** is **MPEG-2 TS/UDP.**

MFVP Encoding live uses audio watermarking and converts it in ID3 tags to be integrated in HLS stream.

To activate Nielsen watermark, follow these steps:

1. From **Private data** subtab, a specific section is displayed.

Figure 49. Private data: Nielsen ID3



2. In the **Nielsen watermark extraction** section, specify the following information:

Distributor ID DNS domain name of the company or entity operating the device.
Default value is **www.envivio.com**
Max. number of characters: 40

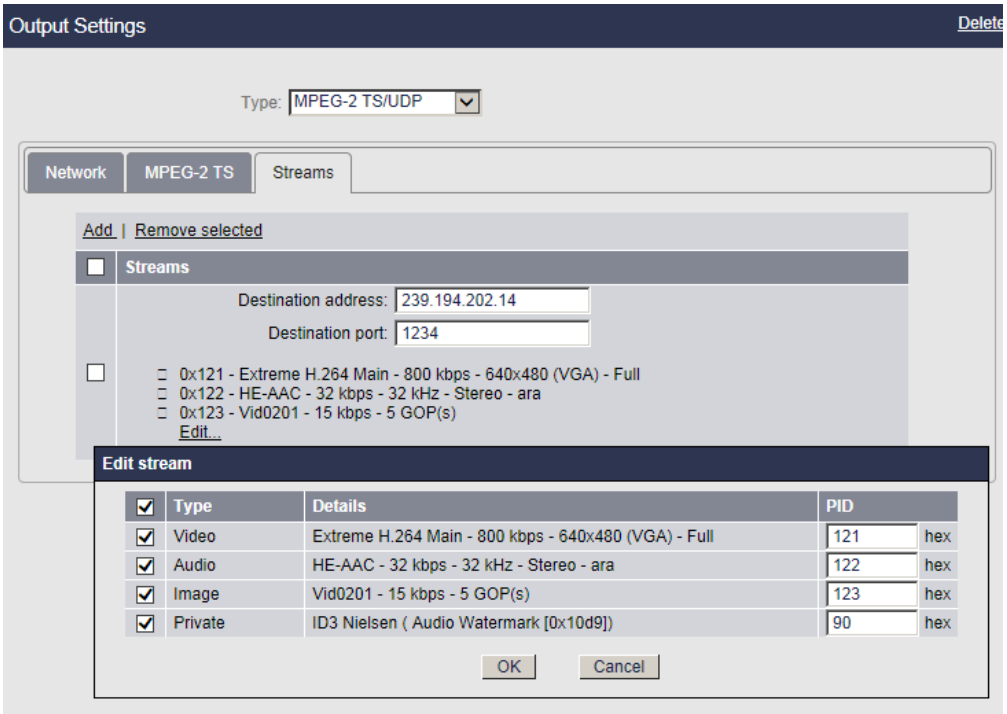
Breakout code Select the breakout code.

Possible values: **Live content with same TV Ads (**Original simulcast content with same Ad-load), **Live content without same TV Ads (**Original simulcast content with no or altered Ad-load), **Live content with same TV Ads and Digital Ads (**Original Simulcast content with same Ad-load with additional Digital Ads), **Live content Digital Ads only (**Simulcast content with altered Ads and additional Digital Ads)

Extract watermark from Select the audio PID.

3. Once you have defined the Nielsen stream, you can add it to the output (see *"Specific MPEG-2 TS/RTP and MPEG-2 TS/UDP parameters" on page 130*).

Figure 50. Example of MPEG-2 TS/UDP output settings with Nielsen stream



Metadata Management

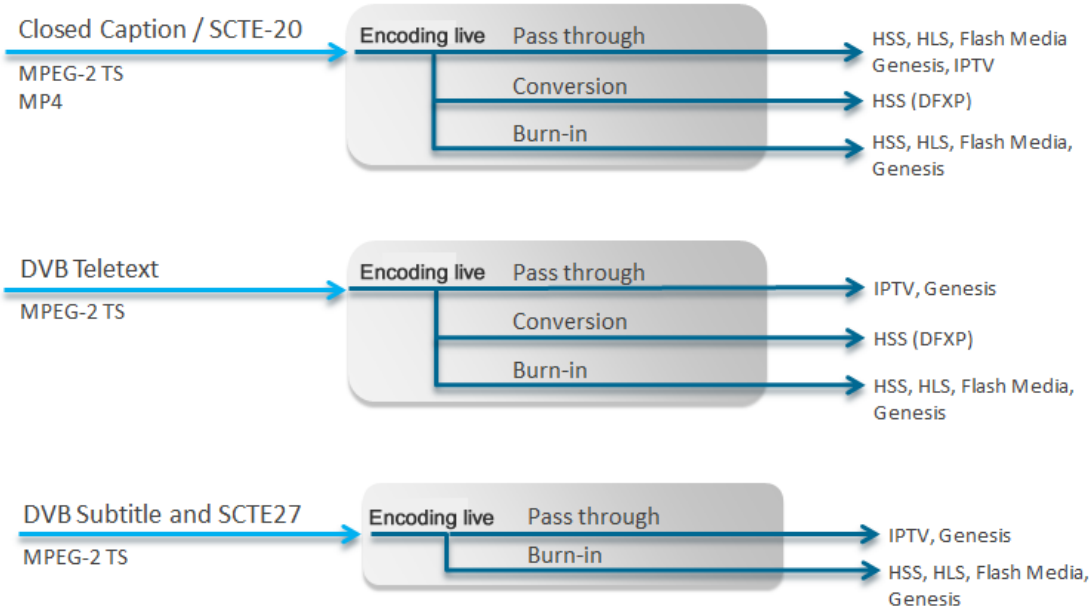
Subtitles

Depending of the ingest type, the following subtitle types are supported by MFVP Encoding live:

Input type	Closed Caption	DVB Teletext/ SMPTE 2031	DVB Subtitle	SCTE 27(1)
Analog	[EIA608B]	[ETSI EN 300 706]	-	-
SD-SDI	[EIA608B]	[ETSI EN 300 706]	Through IP	-
HD-SDI	[EIA608B] [EIA708B]	[OP-47/ SMPTE2031]	-	-
MPEG-2 TS	[ATSC A/53] [SCTE-20]	Private stream [ETSI EN 300 472]	Private stream [ETSI EN 300 743]	Private stream [SCTE 27]

1. SCTE 27 is translated into DVB subtitle format or passed-through

Depending of the network output type, metadata are managed by MFVP Encoding live as described in the table below:



Subtitles for Internet TV outputs

Due to legal requirements in many countries, subtitles are mandatory on all broadcasted channels. Each Internet TV output has its own standard to manage these subtitles (WebVTT for HLS, DFXP for Smooth Streaming). Moreover some subtitle formats (DVB bitmaps) need to be translated into specific formats within the Internet TV standards.

Time code management

MFVP Encoding live can pass-through source LTC/VITC time code to provide the foundation and time basis for adaptive streaming. (EDL in compressed format (for example, rough cut editor))

Time code information is stored in H264 SEI message.

Ad signaling

Depending of the ingest type, the following cue tones types are supported by MFVP Encoding live:

Input type	SCTE 35	EBIF	SCTE 104
Analog	-	-	-
SD-SDI	-	-	Translation to SCTE 35
HD-SDI	-	-	Translation to SCTE 35
MPEG-2 TS	[SCTE 35]	[EBIF]	-

SCTE 104 to SCTE 35

SDI baseband signal contains SCTE 104 signal. MFVP Encoding live can translate it into an SCTE 35 stream. This SCTE 35 stream can then be delivered in an MPEG-2 TS stream or translated to HTTP Live Streaming or Smooth Streaming as described in the table below.

Depending of the network output type, metadata are managed by MFVP Encoding live as described in the table below:

Network output	SCTE 35	EBIF
IPTV (MPEG-2 TS)		[EBIF]
Smooth Streaming	Sparse track	Sparse track
HTTP Live Streaming	[SCTE 35] Specific segmentation	[EBIF]
Flash	*No Standard*	*No Standard*
Genesis	[SCTE 35]	[EBIF]

Smooth Streaming: Text tracks are defined to deliver metadata to Silverlight player. Sparse track is a text track type for delivering sporadic metadata such as EBIF and SCTE 35

HTTP Live Streaming: SCTE 35 metadata can impact the segmentation mechanism by generating a new segment at each splice point. The encoder also inserts splicing information tags in the playlist based on SCTE 35 data.

SCTE 35 can be optionally passed through in a specific PID in all MPEG-2 TS outputs.

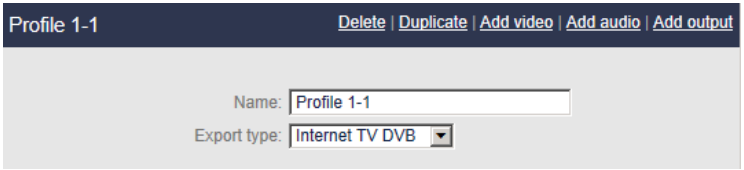
Setting the output parameters

NOTE Depending on license and export type, up to fifteen outputs are allowed per profile.

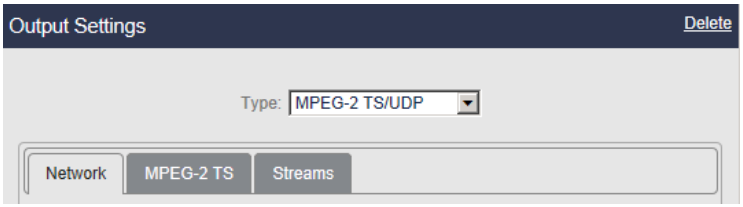
Adding or removing an output

With **Internet TV DVB** and **Internet TV ATSC** export types, you can add and remove the outputs.

To add an output, click the **Profile #** tab in the left panel, then click the **Add output** link in the top right corner. An output is added.



To delete an output, select the output stream in the left panel then click the **Delete** link in the top right corner. The output stream is deleted.



Configuring the output parameters

To display the output parameters, follow these steps:

1. Click the **Output** subtab, the output parameters are displayed in the right panel.
2. Select the appropriate value for each parameter.

NOTES Depending on the export type, you can configure one or two outputs independently.

Output parameters are displayed in several tabs that you have to fill one after the other.

Type Select the broadcast type.

Possible values: *see table below*.

Depending on the selected output type, specific parameters are displayed.

3. Available outputs per export type

Export type	Output type
Internet TV ATSC **Internet TV DVB**	None, MPEG-2 TS/UDP[1], MPEG-2 TS/RTP, HTTP Live Streaming, Smooth Streaming, RTMP
ISMA	None, RTP

1. MPEG-2 TS is compliant with ISO/IEC IS 13818-1.

Network interface Select the appropriate network interface.

Possible values: **Ethernet 1, Ethernet 2, ethernet 5** or **Ethernet 6, Ethernet 1 (backup: Ethernet 2),** or **Ethernet 6 (backup: Ethernet 5).**

Select **Ethernet 1 (backup: Ethernet 2)** –or Ethernet 6 (backup: Ethernet 5)– if you want redundant Ethernet interfaces. In this case, Ethernet 1 will be used to stream by default, but if Ethernet 1 fails, then Ethernet 2 will be used instead.

The **Ethernet 1 (backup: Ethernet 2)** is a logical interface. The same physical Ethernet interface will be used by all profiles which have selected this interface. For example, if a profile 1 uses Ethernet 2 after a backup, all profiles that have selected **Ethernet 1 (backup: Ethernet 2)** will use the same Ethernet interface after stop/start.

If you want to force MFVP Encoding live to switch back to Ethernet 1, you will have to either stop/start all the profiles that have selected **Ethernet 1 (backup: Ethernet 2)** or to unplug Ethernet 2.

Specific MPEG-2 TS/RTP and MPEG-2 TS/UDP parameters

Network

Source address You can either select the Ethernet address configured via the web interface *(see "Configuring the Ethernet connectors" on page 31)* or enter a virtual IP address.

NOTE: The virtual IP address can be the same as another output on MFVP Encoding live but it must be linked to the same Ethernet interface.

The virtual IP address MUST be on the same subnet as the primary IP it virtualizes. Otherwise, an error is displayed. A unique virtual IP address should be applied to other profiles of the same service.

When the virtual IP applies to **Ethernet 1 (backup: Ethernet 2)**–or Ethernet 6 (backup: Ethernet 5)–, both network interfaces 1 and 2 should be on the same subnet, otherwise a warning is displayed.

Using a virtual interface is a convenient way to decorrelate the streaming from the hardware, that is the Ethernet connectors.

Virtual IP addresses are useful:

- When two network interfaces are bonded, for preserving the stream when a network interface fails (no need to replace the source IP with the backup network interface)
- In general, for preserving the stream when using backed up encoders

Source port Source port for streams.

Possible values: from **1024** to **65534** (even number)

TTL TTL stands for Time To Live.

A value in the range 0 through 255 defines the scope within which multicast packets should be sent over a network using Internet Protocol (IP).
Each router decrements the TTL by one. When the value reaches a predefined lower limit, the router throws the packet away. By default the TTL is set to 128.

ToS **ToS** stands for Type Of Service byte (for QoS purpose).

Possible value: numerical value (3 hexadecimal bytes), from **0** to **0x3F**.

A flag is added to the IP packet headers to show which kind of information is embedded in the IP stream so that switches can identify the traffic type without having to understand the traffic.

GOP signaling Specify the method used to signal the boundary of the fragments (GOPs). Following methods are available:

- **IDR-based**: use this mode to detect GOP boundaries, using IDR frames.
- **RAP-based**: GOP is signaled using Random Access Point of the MPEG-TS standard.
- **ALD-based**: GOP is signaled using Adaptation Data field layer from the MPEG-TS source (deprecated CableLabs specification)
- **EBP-based**: GOP is signaled using Adaptation Data field layer from the MPEG-TS source (up-to-date with CableLabs specification)

MPEG-2 TS settings

PMT PID PID (Packet IDentifier) of the PMT (Program Map Table) stream.

Possible values: from **0x0010** and **0x1ffe** (hexadecimal) or from 16 to 8190 (decimal).

PCR PID PID of the PCR, Program Clock Reference used to synchronize the video and audio packets.

Possible values: from **0x0010** and **0x1ffe** (hexadecimal) or from 16 to 8190 (decimal).

Target PCR Period PCR period (in milliseconds)

Possible values: from **20** to **1000** ms.

Target PSI Period *Internet TV DVB / ATSC export types only.*

The PSI period size is the same as the segment size.

Insert TOT & TDT Check this option to generate a TOT/TDT table. The TOT/TDT tables are used to insert time information into MPEG-2 TS streams.

If you check this option, you must define the **Target TOT/TDT period** (from **1** to **30** s).

Program number
Program identifier.
Possible values: from **0x0001** to **0x1fff**.

Service name
Internet TV DVB / ATSC export types type only.
Service name in SDT.

Service provider
Provider of the program stream.

Discard stream lost packets
Select MFVP Encoding live behavior if the input audio or video input is lost.
Possible value: **Stuffing**, **Drop PID** or **None**

In CBR mode:

- If you select **Stuffing**, stuffing will be sent on PID 0x1fff to maintain the profile target bit rate.
- If you select **Drop PID**, packets will be dropped and the output PMT will be updated to indicate that there is nor more audio PID.
- If you select **None**, silence or test pattern will be sent at the same bit rate.

In Statmux mode:

- If you select **Stuffing**, stuffing will be sent at statmux group level to maintain the statmux group target bit rate.
- If you select **Drop PID**, packets will be dropped and the corresponding free bandwidth will be reallocated to the statmux group for audio and video.
- If you select **None**, silence or test pattern will be sent at the same bit rate.

In VBR mode:

- If you select **Stuffing**, stuffing will be sent on input loss.
- If you select **Drop PID**, packets will be dropped on input loss but the output PMT table will not be updated.
- If you select **None**, silence or test pattern will be sent on input loss.

Use LATM encapsulation for AAC
By default, AAC encapsulation is ADTS.
Check this option if you want it to be LATM.

Align video frames to PES packets Check this option to align video frames to PES packets.

NOTE: This option guarantees the interoperability with set-top boxes but increases the bandwidth.

Include max bitrate in PMT Check this option if you want the max bitrate to be included in the PMT.

PCR always on video Check this option to align video stream and PCR PIDs for all output streams.

Transport streams

You can specify the composition of the output stream by selecting the streams that will be sent to the selected output.

NOTE The number of audio streams depends on the license.

Destination address The first time, a random multicast IP address is initially set, you can modify this address by entering a unicast or multicast address.

NOTES: A multicast address is between **224.0.1.0** and **239.255.255.255**.

Destination port Destination port for streams.
Note that if you are using RTP, the RTCP stream will be incremented from this port number, for example:

RTP stream: **5020**
RTCP stream: **5021**
Possible values: from **1024** to **65535**

Audio PID MPEG-2 TS parameter. PID of the audio stream.

Possible values: from **0x0010** and **0x1ffe** (hexadecimal)

NOTES: The audio PID defined is for the first audio. The second is increased by +1 and so on.

Video PID *MPEG-2 TS parameter*. PID of the video stream.

Possible values: from **0x0010** and **0x1ffe** (hexadecimal).

Image PID *MPEG-2 TS parameter*. PID of the image stream.

Possible values: from **0x0010** and **0x1ffe** (hexadecimal)

Private PID *MPEG-2 TS parameter*. PID of the private stream.
Possible values: from **0x0010** and **0x1ffe** (hexadecimal)

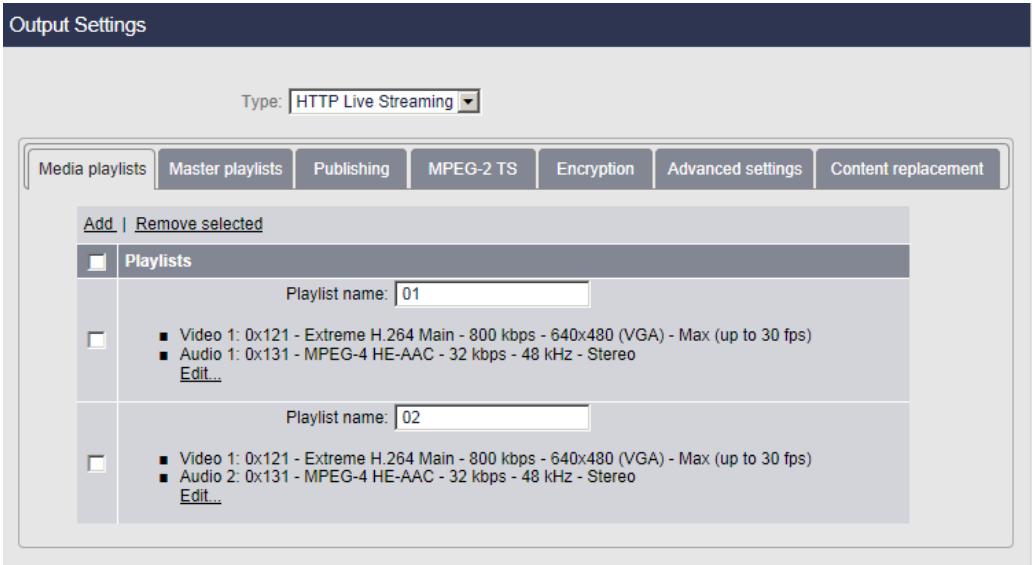
NOTE: If you change the VBI teletext PID value, go to the **Output>Streams** submenu and click the Edit link to edit the private stream PID value

PIDs consistency rules

- PID must be between **0x0010** and **0x1ffe**.
- PMT PID must be different from video and audio PIDs.
- Video PID must be different from audio PID.
- PCR PID can be equal to PMT PID or to Video/Audio PIDs.

Specific HTTP Live Streaming parameters

Figure 51. Output parameters: HTTP Live Streaming



Media playlists

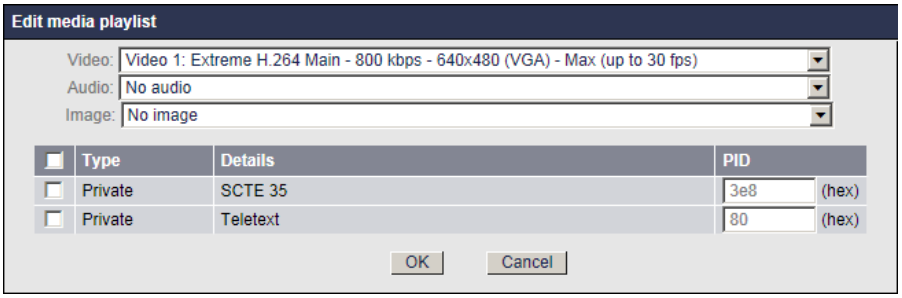
You can add or remove media playlists and define the composition of the playlist by selecting the video stream, the audio stream, the image and the private data that will be sent to the selected output. You can define up to 16 media playlists.

To define a media playlist, follow these steps:

1. Click the **Add** button, a default media playlist is created.
2. Click the **Edit** button.

NOTE We recommend changing the media playlist name.

Figure 52. Media playlist definition



3. In the drop-down lists, select the video, audio and image streams to be added then select the private data by ticking the corresponding checkbox.
4. Click the **OK** button.
5. The media playlist is updated.

NOTES Once selected, private data streams (teletext, subtitle, SCTE-35) are passed through.

You can delete an existing media playlist by checking the media playlist to be removed and clicking the **Remove selected** button.

Master playlists

Once you have defined media playlists, you can define master playlists.

You can define up to 5 master playlists.

To define a master playlist, follow these steps:

1. In the master playlist table, click the **Add** link to create a new master playlist, then click the **Edit** link.
2. Tick the checkbox next to the playlists to be added then click the **OK** button.
3. The master playlist is added to the list of master playlists.

NOTE You can delete an existing master playlist by checking the master playlist to be removed and clicking the **Remove selected** button.

iOS 5 related features

Alternative groups

This feature lets you package each media stream (audio or video stream) and subtitles in a flexible manner.

An audio group (*rendering group* in Apple specifications) aggregates several audio playlists to propose an alternative rendition of the audio for a given video stream. For instance, an English audio playlist can be replaced by a French or a Spanish audio playlist. Inside a group, a default playlist can be set.

To use this feature, you must first create audio-only, video-only and subtitle-only playlists, then you will associate the subtitle, audio and video streams into one or more master playlist(s).

Creating an alternative group

If you want to define an alternative group, follow these steps:

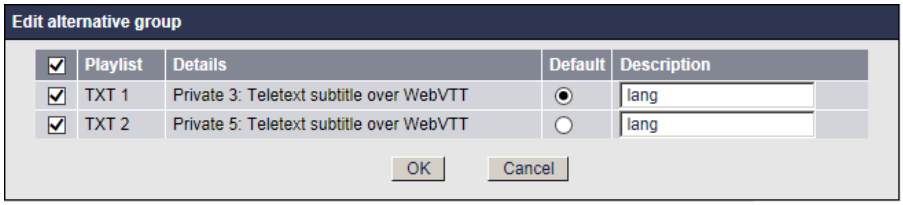
1. Create audio-only, video-only and subtitle-only playlists (*see "Media playlists" on page 135*)
2. From the **Master playlist** tab, tick the **Use alternative groups** checkbox. A new table is displayed.
3. In the alternative group table, you can either:

- click the **Auto-configure** link to automatically create alternative audio and alternative subtitle groups based on the audio-only and subtitle-only playlists you created,

- click the **Add subtitle Groups...** link to create a new group then click the **Edit** link to define the composition of your alternative subtitle group. Select the default audio stream and add a description.

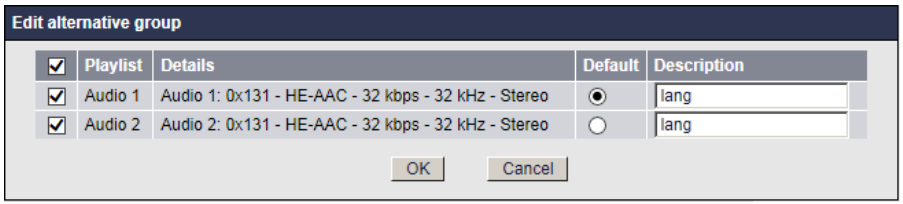
NOTE Alternative subtitle group only works with teletext subtitles (Closed-caption and webVTT).

Figure 53. Editing alternative subtitle groups



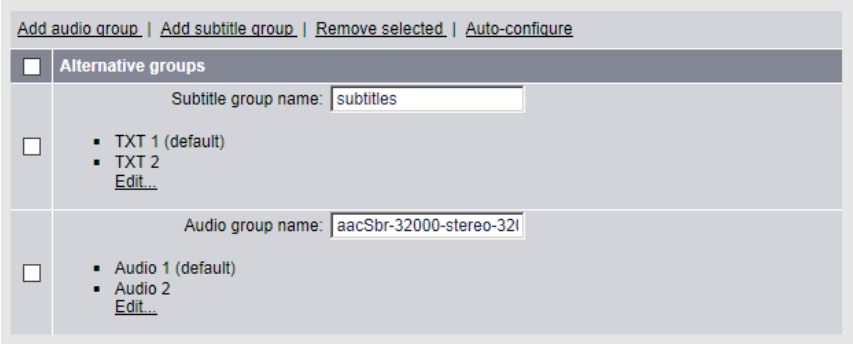
- click the **Add audio group** link to create a new audio group then click the **Edit** link to define the composition of your alternative audio group. Select the default audio stream and add a description.

Figure 54. Editing alternative audio groups



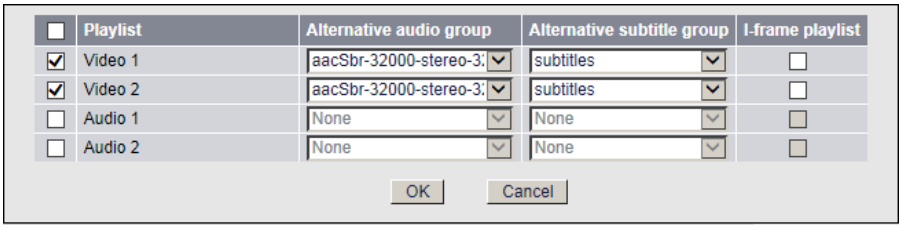
4. The alternative subtitle or audio group is added to the table:

Figure 55. Alternative groups table



5. In the master playlist table, click the **Add** link to create a new master playlist, then click the **Edit** link.
6. Select the alternative audio group and alternate subtitle group you want to associate to the video stream then click the **OK** button.

Figure 56. Editing master playlists



7. You can check the **I-frame playlist** option to reference I-frames in the stream.

NOTE To optimize rapid forward and reverse playback, Apple introduced the notion of I-Frame playlist in iOS5.

8. The master playlist is added to the master playlist table.

Figure 57. Master playlists table

HLS Protocol version depending on MFVP Encoding live settings

Below is a table showing the value of the EXT-X-VERSION tag that is inserted by the encoder for each type of playlist (columns in the table) based on different encoder settings (rows in the table):

Feature	Master playlist	Media Playlist	I-frame playlist	Encrypted media playlist
Default	-	3	4	3
I-frame playlist enabled	-	3	4	3
Use alternative audio groups	4	3	4	3

Publishing

Publishing point Specify the location where fragments and playlist/index files will be published.

http://<IP Address>/path

http://<DNS host name >/path

NOTE: If you defined credentials to access this URL (*see "Adding a credential" on page 235*), the user icon, should appear in color.

Distribution point Enter the URL where fragments and playlist files can be retrieved.

Secondary publishing point Enter the URL of the secondary publishing point for redundancy purpose.

Secondary distribution point

Enter the URL of the secondary distribution point where fragments and playlist files can be retrieved.

Separate playlists

Check this option if you want to upload/download playlists to/from a different location.

If you check this option, new parameters are displayed. Specify the main/backup publishing and distribution points for the playlists.

Separate segments

Check this option if you want to upload/download segments to/from a different location.

If you check this option, new parameters are displayed. Specify the main/backup publishing and distribution points for the segments.

Use thumbnails

If you check this option, JPEG thumbnail images will be generated every 3 seconds. These JPEG files will be uploaded in the same location as fragments and playlist/index files. Thumbnails can be used to create a dynamic EPG for instance.

Separate thumbnails

Click this checkbox if you want the thumbnails to be published in another location than the master playlist file, then fill in the different parameters.

Network interface: select which network interface is used to publish thumbnails.

Publishing point: specify the location where thumbnails will be published.

Secondary publishing point: for redundancy aspects, a secondary publishing point can be configured.

NOTE: If you defined credentials to access this URL (*see "Adding a credential" on page 235*), the user icon, should appear in color.

Master playlist publishing period

You can define the master list republishing.

Possible values: from **0** to **86400** seconds

If you set this parameter to 0, no republishing will occur.

Delete expired content

If you check this option, contents generated by the HLS output (segments, encryption keys, thumbnails and directories) are removed from the publishing server when they are no more referenced in the available playlist.

It deletes one segment every time it uploads a segment. The number of segments on the server at any point in time is equal to 2 x N + 3 (where N is **Segment(s) per playlist** setting).

The sequence numbers on the segments DO NOT rollover. However, perhaps related to that, if you enable the **Use subdirectories** option, the encoder will create a new subdirectory for putting segments inside once every N segments (where N is the **Segments per subdirectory** setting).

NOTE: Media playlist and master playlist are not removed.

Use subdirectories

If you check this option, you can define the maximum number of files per upload directory.

NOTE: Deleted fragments are part of the overall count.

MPEG-2 TS settings

Video stream PID

MPEG-2 TS parameter. PID of the video stream.

Possible values: from **0x0010** to **0x1FFE** *(see "PIDs consistency rules" on page 134).*

CAUTION: If you set the **Export type** to **HTTP Live Streaming**, all the video streams will have the same PID as the first video stream.

Audio stream PID

MPEG-2 TS parameter. PID of the audio stream.

Possible values: from **0x0010** to **0x1FFE**

See "PIDs consistency rules" on page 134.

Image stream PID

MPEG-2 TS parameter. PID of the image stream.

Possible values: from **0x0010** to **0x1FFE**

See "PIDs consistency rules" on page 134.

PMT PID PID (Packet IDentifier) of the PMT (Program Map Table) stream.

Possible values: from **0x0010** and **0x1ffe** (hexadecimal) or from 16 to 8190 (decimal).

PCR PID PID of the PCR, Program Clock Reference used to synchronize the video and audio packets.

Possible values: from **0x0010** and **0x1ffe** (hexadecimal) or from 16 to 8190 (decimal).

Target PCR Period PCR period (in milliseconds)

Possible values: from **20** to **1000** ms.

Target PSI Period PSI period is aligned with segments

Insert TOT & TDT Check this option to generate a TOT/TDT table. The TOT/TDT tables are used to insert time information into MPEG-2 TS streams.

If you check this option, you must define the **Target TOT/TDT period** (from **1** to **30** s).

Program number Program identifier.

Possible values: from **0x0001** to **0xffff** (hexadecimal) or from 1 to 65 535 (decimal).

Use LATM encapsulation for AAC By default, AAC encapsulation is ADTS.
Check this option if you want it to be LATM.

Align video frames to PES packets Check this option to align video frames to PES packets.

NOTE: This option guarantees the interoperability with set-top boxes but increases the bandwidth.

Include max bitrate in PMT Check this option to XXX

.

Encryption

Encryption type Possible values: **None**, **Segment based** or **Authentec**

By default, the protection is not activated.

If you enable the encryption, specific parameters are displayed *(see "Encryption for HTTP Live Streaming" on page 222)*.

Key source If you selected **Segment based** as **Encryption type,** possible values are **Internal key generation** or **External key generation**

If you selected **Authentec** as **Encryption type,** possible values are **External key generation** or **Fixed key**

Advanced settings

GOP(s) per segment You can define the number of GOP(s) per segment.

Possible values: from 1 to 40

Recommended value: from 1 to 10

The estimated segment size is automatically updated.

NOTE: The segment duration is computed from the Number of GOP(s) and the Key frame period.
It cannot exceed 20 seconds.

Segment(s) per playlist You can set the number of GOP(s) per segment.

Possible values: from 3 to 3600

The estimated playlist size is automatically updated.

Program date time If you check this option, a program date information is inserted in the playlist. This tag associates the first sample of a media segment with an absolute date and time.

Cross reference The cross reference enables redundancy at player level. It is used when playlists are published on both primary and backup publishing points. If you check this option, master playlists reference both the primary and the backup publishing points.

Example of a master playlist using cross reference:

```
#EXTM3U
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=500832
http://myPrimaryServer/france5/01.m3u8
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=500832
http://myBackupServer/france5/01.m3u8
```

End of stream Tag value: `#EXT-X-ENDLIST`

If you check this option an EOS (End Of Stream) tag is inserted at the end of the playlist when the associated profile is stopped.

Custom naming

Use custom naming

CAUTION: Reserved for advanced users.

Check this option if you want to customize the playlist name, the master playlist name, the segment name and the thumbnail name.

Five parameters are used to customize the output file names: **Segment pattern**, **Master playlist pattern**, **Media playlist pattern**, **I-frame playlist pattern** and **Thumbnail pattern**

These five patterns support the following keys (these keys will be replaced at encoding time with the appropriate value):

Key	Description
${starttime}	Time at session start
${bitrate}	Bitrate (in kbps) associated to the file
${id}	Session identifier
${seq}	Sequence number associated to the file
${curtime}	Time at file creation
${lang}	Language associated to the file
${variantid}	Name of the master playlist.

NOTE **${id}** must be before **${seq}**, and **${seq}** must be before **${curtime}**.

Below are the rules to follow for an Apple HTTP output:

Table 9. File naming rules

Parameter name	Default value	Supported keys	Required keys	Specific rules
Segment	${starttime}-${id}-${seq}.ts	${starttime} ${bitrate} ${id} ${seq} ${curtime} ${lang}	${id} ${seq}	If subdirectories are used: - ${id} must be before ${seq} - ${seq} must be before ${curtime}
Master playlist	${variantid}.m3u8	${lang}		
Media playlist	${id}.m3u8	${id} ${bitrate} ${lang}		
iFrame playlist	${id}-iframe.m3u8	${id} ${bitrate} ${lang}		
Thumbnail	thumb.jpg	${starttime} ${seq} ${curtime}		

Example for file naming rules:

- pattern=${id}_${seq}_${bitrate}_${curtime}.ts

The created files will have these names:

- 01_341_750_20100101T010101.ts
- 01_342_750_20100101T010103.ts
- 01_343_750_20100101T010105.ts

64 kbps stream for iPhone and iPad

Apple requires an alternate stream at 64 kbps or less for slow data connections. Recommended audio settings:

- **Audio codec**: HE-AAC (MPEG-2)
- **Bit rate**: 40 kbps
- **Channels**: Stereo

- **Sampling rate**: 48 kHz

Content replacement

Linear ad insertion in HLS consists in inserting tags or replacing the URLs referencing the ad chunks by new ones (local/regional ad) within the playlist. Two strategies are possible:

- Splice Marker playlist insertion (**Playlist Marker**)
- URL substitution (**Playlist Marker/URL substitution from POIS**)

NOTE By default, the replacement type is set to **None** and therefore the content replacement is not activated.

Replacement type	Possible values: **None**, **Playlist marker** or **Playlist marker/URL substitution from POIS**
Reference SCTE 35 stream	Select the SCTE 35 PID from which splice-in and splice-out information will be derived.

Specific parameters for URL substitution from POIS

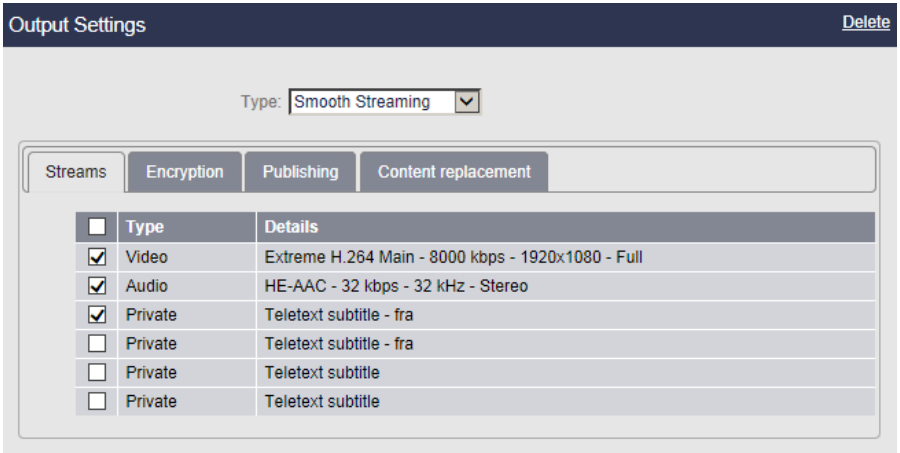
Network interface	Select the network interface to be used.
POIS server URL	Enter the POIS (Placement Opportunity Information Service) server URL.
Network name	Enter the name of the Network.

Specific Smooth Streaming parameters

Streams

You can specify the composition of the output stream by selecting the streams that will be sent to the selected output.

Figure 58. Smooth Streaming output parameters: Streams tab



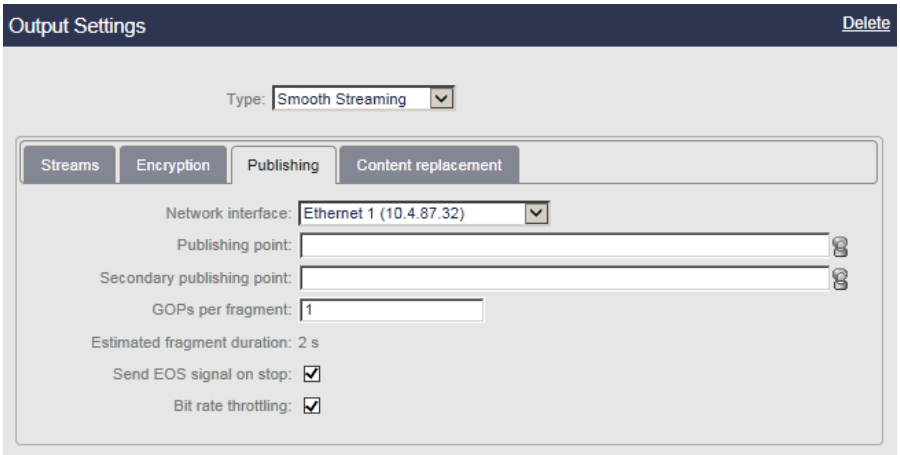
NOTE The number of audio streams depends on the license.

Encryption

By default, the protection is not activated. If you set the **Encryption type** to **Fixed key, Fixed key seed, Fixed protected key seed** or to **External key generation**, specific parameters are displayed *(see "Encryption for Smooth Streaming profile" on page 219)*.

Publishing

Figure 59. Smooth Streaming output parameters: Publishing tab



Publishing point

Enter the IP address of the publishing point you previously created.

URL: http://<server address>/Publishing point

NOTE: If you defined credentials to access this URL (*see "Adding a credential" on page 235*), the user icon, should appear in color.

Secondary publishing point

Enter the IP address of the secondary publishing point.

URL: http://<backup server address>/Publishing point

NOTE: If you defined credentials to access this URL (*see "Adding a credential" on page 235*), the user icon, should appear in color.

GOPs per fragment

You can define the number of GOP(s) per fragment.

Possible values: from 1 to 20, recommended values: from 1 to 10

The estimated fragment size is automatically updated.

NOTE: The fragment size is computed from the Number of GOP(s) and the Key frame period.
The maximum fragment duration is 20 seconds.

Estimated fragment duration Informative field.

NOTE: This value is computed by multiplying the number of GOPs per fragment by the video frame rate. It cannot exceed 20 seconds.

Send EOS signal on stop If you check this option an EOS (End Of Stream) is automatically sent to the IIS server when the associated profile is stopped.

Bit rate throttling To ensure that the bandwidth usage of the smooth streaming output is kept close to the stream bitrate, MFVP Encoding live implements a "leaky bucket" mechanism within the smooth streaming output.

To activate this mechanism, tick this checkbox.

Content replacement

Linear ad insertion in Smooth streaming consists in inserting markers referencing the ad boundaries within the dedicated sparse track.

NOTE By default, Content replacement mode is set to **None** and therefore not activated.

Replacement type Possible values: **None** or **Sparse track marker from POIS**

Reference SCTE 35 stream Select the SCTE 35 PID from which splice-in and splice-out information will be derived.

Network interface Select the network interface to be used.

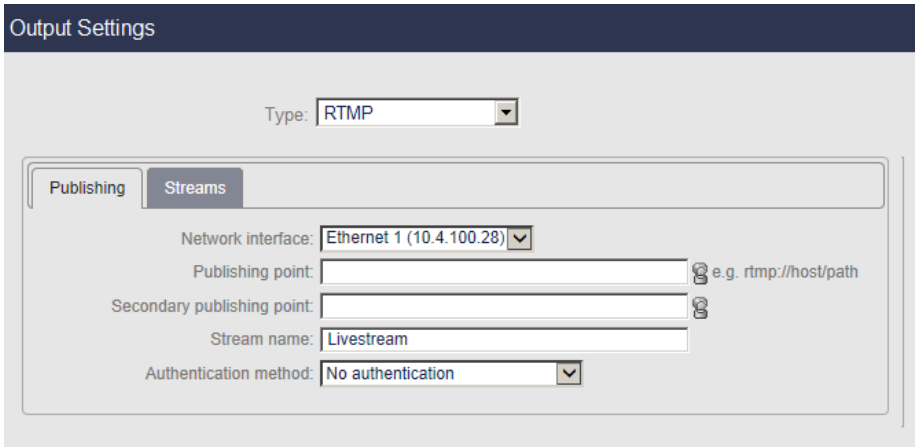
POIS server URL Enter the POIS (Placement Opportunity Information Service) server URL.

Network name Enter the name of the Network.

Specific RTMP parameters

CAUTION Akamai Compliance: to ensure a single TCP connection per stream, configure a Flash RTMP output per video stream

Figure 60. Output parameters: RTMP



Publishing

Publishing point

Enter the URL of the primary Flash Media Server publishing point. For live streaming, configuration shall be done the following way:

RTMP://<Primary FMS IP Address>/live

or RTMP://<Primary FMS host name >/live

NOTE: If you defined credentials to access this URL (*see "Adding a credential" on page 235*), the user icon, should appear in color.

Secondary publishing point

Enter the URL of the secondary Flash Media Server publishing point.

NOTE: If you defined credentials to access this URL (*see "Adding a credential" on page 235*), the user icon, should appear in color.

Stream name

Name of the live stream.

See "Using parameters in filenames and stream names" on page 151.

Authentication method

Select the authentication type.

Possible values: **No authentication**, **Authentication with Flash Media Server**, **Authentication with Akamai, Authentication with Limelight,** or **Authentication with Level 3**

When publishing from the broadcasters facility to CDNs ingest such as Akamai, Limelight or Level 3 (Flash), a level of authentication is required. MFVP Encoding live implements Akamai proprietary authentication, as well as Flash (Adobe Level 3 type).

Using parameters in filenames and stream names

When saving encoded streams to file, you can use parameters in filenames and stream names. During multi-bit rate encoding, it is especially helpful to include the bit rate or other information in the name to distinguish the content.

NOTE For stream names, you can use either parameters or separate unique stream names with a semicolon, but not both together. If you use both parameters and semicolons, Flash Media Live Encoder treats the characters used to represent parameters as literal characters in the stream name.

To use parameters, enter an optional identifier (name) of your choice, followed by one or more parameters.

Parameter	Description
%i	The index of the bit rate. This value is meaningful when encoding streams at multiple bit rates. The first index value is 1. Flash Media Live Encoder supports up to three streams at a time, so the index values are 1, 2, and 3. Adobe recommends that you include the index parameter in the stream name. The index parameter can differentiate streams/files that otherwise have identical values for bit rate, sample rate, and so on.
%v	The video bit rate.
%f	The output frame size.
%a	The audio bit rate.
%s	The audio sample rate.
%b	The total bit rate (that is **%v+%a**).

You can use multiple parameters in a single name. When using multiple parameters, if desired, you can separate parameters with underscores to make the resulting names easier to read.
For example, if you want to create an FLV file whose name reflects the video bit rate of 156 Kbps and the output frame rate of 24 fps, enter **myFLVFile_%v_%f**. The resulting filename is **myFLVFile_156_24**.

NOTE While it is not required to enter an identifier, such as the identifier ***myFLVFile*** used in the example, an identifier of your choosing can help differentiate files and streams.

Transport Streams

You can specify the composition of the output stream by selecting the streams that will be sent to the selected output.

By default, one transport stream is defined, composed of the first video stream and the first audio stream.

You can define up to 31 transport streams.

To add a transport stream, follow these steps:

1. Click the **Add** link. The transport stream is added to the list of transport streams.
2. Click the **Edit** link and select the video stream, audio stream and private data to be added then click the **Ok** button.

NOTE You can delete an existing transport stream by checking the transport stream to be removed and clicking the **Remove selected** link.

Specific RTP parameters (ISMA)

Destination address The first time, a random multicast IP address is initially set, you can modify this address by entering a unicast or multicast address.

NOTES: A multicast address is between **224.0.1.0** and **239.255.255.255**.

Destination port Destination port for streams.
Note that if you are using RTP, the RTCP stream will be incremented from this port number, for example:

RTP stream: **5020**
RTCP stream: **5021**
Possible values: from **1024** to **65535**

TTL TTL stands for Time To Live.

A value in the range 0 through 255 defines the scope within which multicast packets should be sent over a network using Internet Protocol (ιπ).
Each router decrements the TTL by one. When the value reaches a predefined lower limit, the router throws the packet away. By default the TTL is set to 128.

Max packet size Defines the maximum size of the RTP packet (in bytes). It includes payload and RTP header.

Possible values: from **300** to **65495**

See paragraph below to calculate the maximum packet size.

ToS video, audio & SAP **ToS** stands for Type Of Service byte (for QOS purpose).

Possible value: numerical value (3 hexadecimal bytes), from **0** to **0x3f**. One separate flag can be set for video/audio/network packets.

A flag is added to the IP packet headers to show which kind of information is embedded in the IP stream so that switches can identify the traffic type without having to understand the traffic.

Session name Session name used to access the live broadcast when using SAP. A default SAP session name is initially set.

SAP announce enabled Check this option to activate the SAP announce.

Content protection mode *This parameter is reserved for future use.*

Defining the maximum packet size

To define the maximum packet size, follow these steps:

1. Identify the MTU size of your network.
2. Identify the packet structure used on your network.
3. Following your streaming configuration, subtract the size of the packet headers from the MTU to get the size available for the RTP packet.

Example 1. RTP streaming over UDP over Ethernet:

The MTU size is 1500, and the packet structure is as follows:

20 bytes IP header	8 bytes UDP header	12 bytes RTP header	N bytes RTP payload

The value to set for Max. packet size when streaming over UDP is:
1500 - 20 - 8 = 1472

Starting/Stopping encoding

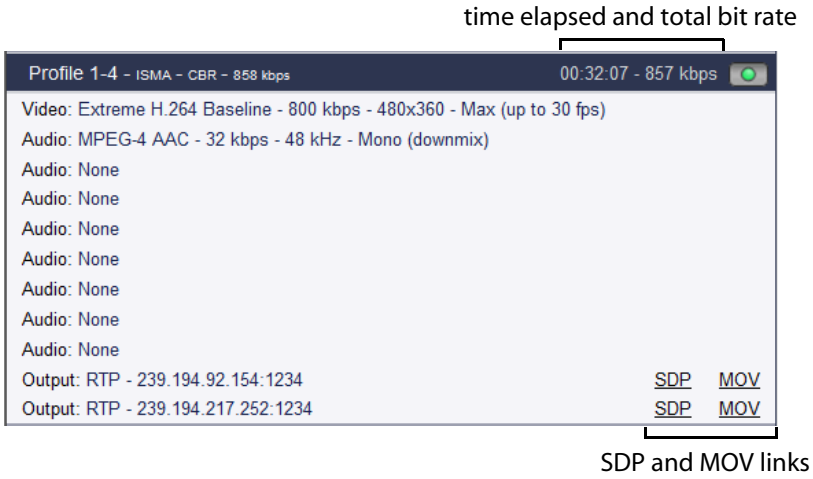
The start/stop button, at each profile level, lets you respectively start and stop the current live broadcast.

- To start encoding a profile, click the button.
- To stop encoding a profile, click the button.

When you start encoding the following information is displayed in the profile left panel. These parameters are regularly updated:

- time elapsed since the beginning of the encoding
- total bit rate (in kbps)
- SDP and MOV links. *ISMA export type with RTP output only. See "RTP streaming" on page 156.*
- SMIL link. *RTMP output only. See "Flash Media" on page 156.*

Figure 61. Encoding information: ISMA export type



Previewing the live broadcast

RTP streaming

Each RTP streaming output has its own announce: SDP file and SAP announce.

When you use RTP, two links, **SDP** and **MOV**, are created when you start encoding *(see "Starting/Stopping encoding" on page 155)*. These links let you access the SDP file or the MOV file.

Flash Media

With RTMP output (Flash Media), a link to a **SMIL** file is created.

The file describes all the video streams available with the associated configuration.

Chapter 6

Configuring a TV profile

Basic workflow

Below is the basic workflow for generating an IPTV stream.

Step #1: Set the service parameters

- Enter a service name,
- Select the input type used to capture the video stream:
  - If **On Board** input is selected, select the input type and the standard
  - If **Ethernet** input is selected, select the Ethernet interface, then enter the source IP address and port where your input source is broadcasting, and select the audio and video PIDs detected in the stream
- Set the video and audio inputs

Step #2: Set the profile parameters

- Enter a profile name
- Select the export type and the bit rate

Step #3: Set the profile video and audio parameters

- Select the audio and video input
- Select the audio and video codecs

Step #4: Set the private data

Step #5: Set the profile output parameters

- Select the transport protocol, then set the output parameters

Step #6: Start the live broadcast session

Step #7: Display alarms (optional)

CAUTION When you modify a parameter and you press ENTER, the modification is automatically saved in the current configuration.

Setting up the encoding profiles

Depending on license, up to four services are displayed, and each service may include one or two profiles that can be configured and started independently from the other profiles.

Creating a profile

By default only one profile by service is already activated and configured (except for PiP option where two profiles are activated). To create a new profile, you can duplicate an existing one.

Adding a new profile

To add a profile, click the **Service** tab, then select the **Add profile** button in the right panel.

MFVP Encoding live creates a new profile using default parameters.

Duplicating a profile

To duplicate a profile, follow these steps:

1. Click the **Profile** tab of the profile you want to duplicate, then select the **Duplicate** button in the right panel.

MFVP Encoding live creates a new profile using the selected profile parameters.

For PiP option, MFVP Encoding live duplicates both the main and PiP profiles.

Deleting a profile

To delete a profile, click the **Profile** tab, then click the **Delete** button in the right panel. Note that all the profile parameters will be lost.

NOTE With PiP option, the two profiles are deleted.

Setting the profile general parameters

NOTES

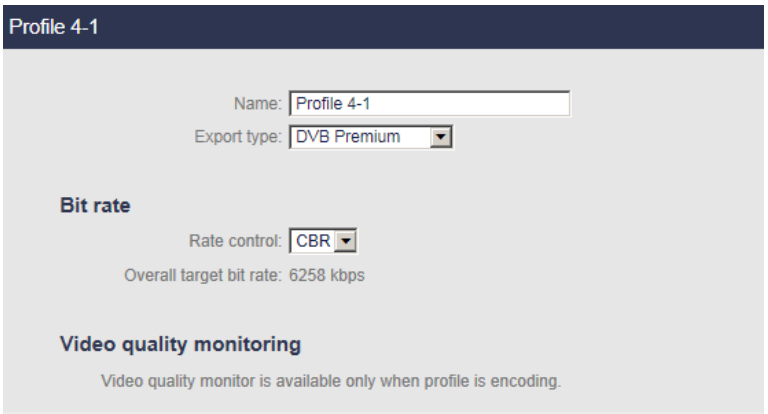
You can refer to "Encoding recommendations" on page 245 for typical encoding settings.

During encoding, all the parameters are greyed and cannot be modified, except the **Overall target bit rate**.

To set the profile general parameters, follow these steps:

1. Click a **Profile** tab. The profile general parameters are displayed in the right panel.

Figure 62. Profile general parameters: DVB Premium export type



2. Select the appropriate value for each parameter.

Name Enter a profile name. We recommend using a name that clearly identifies the channel.

Export type Select one of these values: **DVB Premium**, **DVB Extreme**, **ATSC Premium**, **ATSC Extreme**

If a codec-linked restriction is not respected, an error message is displayed when you start encoding.

Bit rate control

NOTE For more information on bit rate control, see Section 8 on page 205.

Rate control Possible values: **CBR**, **VBR**, **ABR**, or **Statmux**

NOTE: In case of H.264, the stream remains compliant with the normative HRD.

- With **CBR** (Constant bit rate), you can stream content over a limited bit rate channel such as a network. The output stream fits in one bit rate, which you specify as a parameter. Constant bit rate means that the bit rate is constant according to the leaky bucket concept (*see "Appendix G" on page 266.*).
- With **VBR** (Variable bit rate), the output stream will never exceed the Overall max bit rate, and will try to reach the Target network bit rate. This is usually used to save bandwidth. This model is also called "Capped VBR".
- With **ABR** (Available Bit Rate), MFVP Encoding live will never exceed the maximum bit rate, but may go lower than the Target network bit rate if no more bit rate is needed for encoding.
- For **Statmux**, see *"StatMux, statistical rate control" on page 211*

Overall target bit rate Informative field.

The overall target bit rate is automatically computed.

- For RTP, the **Overall target bit rate** is the resulting IP bit rate (i.e. it includes all overhead down to and including IP headers).
- For MPEG-2 TS output, the **Overall target bit rate** includes the resulting transport stream bit rate.

PiP profile

The PiP profile (Picture in Picture) is a specific option.

When PiP is enabled, a new profile is available per activated profile.

- Its export type is the same as the main SD profile:
  DVB Premium PiP, DVB Extreme PiP, ATSC Premium PiP or
  ATSC Extreme PiP

- The resolution value can be:
  96x96, 128x96, 176x144 or 192x192[1]
- Only one audio stream is available for PiP profile.

Video quality monitoring

When encoding a profile, you can display a graphical representation of the video quality (see "Video quality" on page 41) by clicking the **video quality monitor** link.

Image Overlay

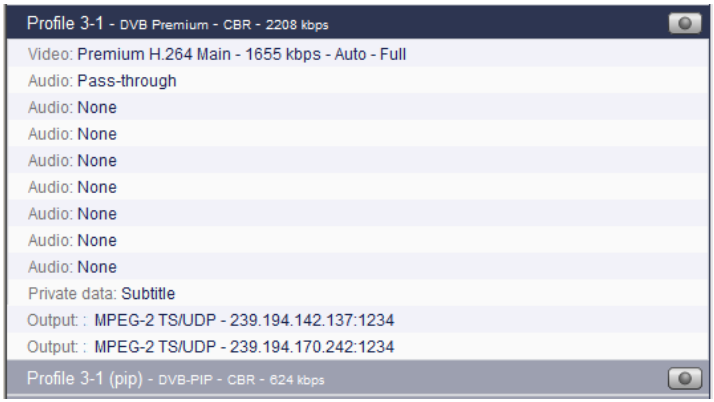
See "Image overlay" on page 88.

1. Extreme video codec only

Displaying the video, audio, and output parameters

To display the video, audio, and output parameters of a profile, click the **Profile #** tab. The selected profile subtabs are displayed below.

Figure 63. Profile subtabs: DVB export type

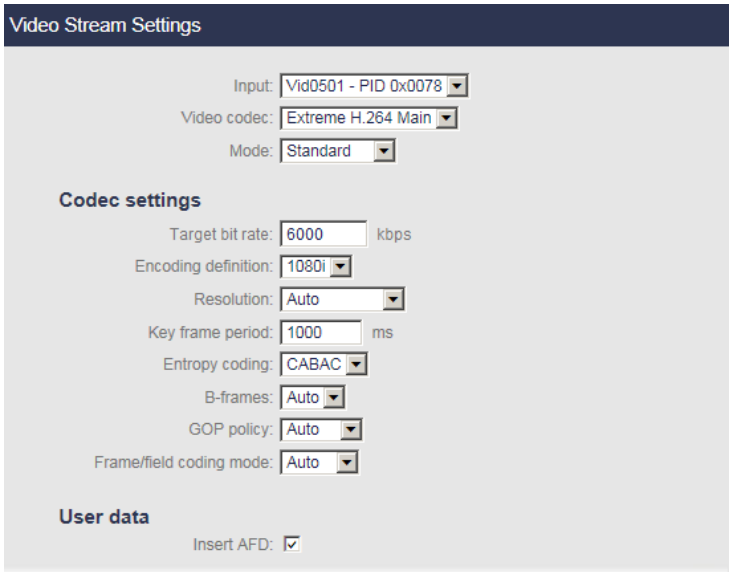


Setting the video stream parameters

To display the video parameters, follow these steps:

1. Click the **Video** stream subtab, the video parameters are displayed in the right panel.

Figure 64. Profile video parameters: Extreme H.264 Main video codec



2. Select the appropriate value for each parameter.

Input Select the appropriate video input channel.
None means audio-only mode.

Video codec Select the video codec.

Possible values: **Premium H.264 main**, **Premium H.264 High**, **Extreme H.264 Main, Extreme H.264 High, MPEG-2 main**, **HEVC Main 10** depending on your license and the export type (*see Table 10)*.

Table 10. Available video codecs per export type

Export type	Video codec
DVB Premium	H.264 main, H.264 high
DVB Extreme	H.264 main, H.264 High, MPEG-2, HEVC main, HEVC Main 10
ATSC Premium	H.264 main, H.264 high
ATSC Extreme	H.264 main, H.264 High, MPEG-2, HEVC main, HEVC Main 10

Buffer size Possible values: **Low delay**, **Standard** or **High quality**.

- **Ultra low delay:** allows reducing the end-to-end delay inside the Silverlight smooth streaming ecosystem to 1-2 second(s)
- **Low delay**: VBV size = 0.500 seconds
- **Standard**: VBV size = 1.000 seconds
- **High quality**: VBV size = 2.000 seconds
- **Custom**: Enter a specific value from 500 to 2000 ms)

The latency modifies the video buffer size. The video buffer is the standard MPEG-4 VBV (Video Buffering Verifier). It is the instantaneous bit rate smoother (transmission buffer) that guarantees that the instantaneous bit rate never exceeds the target bit rate.

Codec settings

Target bit rate Set the video target bit rate (in kbps).

Possible values: **SD**: from 128 kbps to 4 Mbps;
HD: from 2 Mbps to 20 Mbps

NOTE: Video bit rate corresponds to "raw" video bit rate, and does not include bit rate overhead due to encapsulation (such as PES headers, MPEG-2 TS headers).

Foresight encoding *Only available with Extreme video codecs with H.264 over TS sources.*

If you check this option, MFVP Encoding live will use the video source GOP structure to encode the video stream. Foresight encoding allows improving video quality.

NOTE: In this case some parameters are not available.

Encoding definition

Possible values: **UHD**, **1080i, 720p, SD**

Depending on the selected Encoding definition, a list of resolutions is displayed.

Resolution

Select a resolution value. Values depend on the selected **Encoding definition**.

Possible values: *see Table 11 below*.

Auto automatically calculates the resolution depending on video bit rate.

Dynamic changes resolution depending on complexity.

NOTES: Note that resolutions are sorted by the total number of pixels per frame.
For low bit rates, it is recommended to use a low resolution to improve encoding quality.

Table 11. Possible resolutions per encoding definition

Resolution Width x Height		UHD	1080i	720p	SD(NTSC source)	SD (PAL source)	PiP(1)
3840	2160	x					
3D optimized			x				
1920	1080						
1440	1080		x				
1280	1080		x				
960	1080		x				
1280	720			x			
960	720			x			
854	720			x			
848	720			x			
640	720			x			
720	576					x	
704	576					x	
720	480				x		
704	480				x		
544	576					x	
640	480				x		
528	576					x	

Resolution Width	Resolution Height	UHD	1080i	720p	SD(NTSC source)	SD (PAL source)	PiP[1]
480	576					x	
544	480				x		
528	480				x		
480	480				x		
352	576					x	
352	480				x		
352	288					x	
352	240						
320	240				x		
192	192						x
176	144					x	x
176	120				x		
128	96						x
96	96						x

1. Only available with PiP (Picture in Picture) option.

Resolution

With **Premium H.264 Main** and **Premium H.264 High** video codecs, the following resolution is applied when you select **Auto**:

Video bit rate (kbit/s)	Resolution
1500 to 4000	3/4
1100 to 1499	2/3
800 to 1099	1/2

With **Extreme H.264 Main,** and **Extreme H.264 High** video codecs, the following resolution is applied when you select **Auto**:

1080i

Video bit rate (kbit/s)	Resolution
7000 and above	1/1
5000 to 6999	3/4
3500 to 4999	2/3
2000 to 3499	1/2

720p	5500 and above	1/1
	4000 to 5499	3/4
	2000 to 3999	1/2
SD	2000 to 4000	1/1
	1500 to 1999	3/4
	1100 to 1499	2/3
	800 to 1099	1/2
	Below 800	1/4

Key frame period

Possible values: from **500** to **3000** ms

Enter the maximum time between two key frames. You can specify how often you want key frames to be inserted into the video stream. Key frames are inserted into the stream periodically to synchronize the decoder and enable it to recover from errors.

Refreshing the image more often (by setting a shorter key frame period) reduces the recovery time but requires a higher bit rate to maintain encoding quality.

Entropy coding

Only available with Extreme video codecs.

Possible values: **CABAC or CAVLC**.

CABAC means Context Adaptative Binary Arithmetic Coding. This is an entropy lossless compression algorithm, but very time-consuming, especially at the decoder side.

CAVLC means Context Adaptative Variable Length Coding. This is the other entropy lossless compression algorithm used in the H.264 format. It is less efficient than CABAC, but faster (especially at the decoder side).

B-Frames

Possible values: **Off**, **Auto** (the number of B-frames is calculated by the encoder to be compliant with most set-top boxes), **2B**, or **3 Hierarchical B** (*only available with Extreme export types*)

B-frames are used to increase quality. It indicates whether bidirectional encoded pictures (B-frames) are used. A B-frame can be considered as an interpolated frame between reference frames (I or P), interpolation is done using forward and backward motion vectors used to select pieces of pictures in reference frames.

GOP policy

The GOP policy allows to support any STB for interoperability. Open GOP is the best solution to optimize VQ but may be not supported by STB.

Possible Values: **Auto** (open gop), **Open** (frames may have reference frames from outside the GOP), or **Closed** (each frame of the GOP is independent from the frames outside of the GOP).

Frame/field coding mode

Possible values for H.264 and MPEG-2 video codecs: **Auto** (PAFF), **Frame** (frame coding) or **PAFF** (picture adaptive frame field)

Possible values for HEVC video codec: **Auto** (SAFF), **Frame** (frame coding) or **Field** (field coding)

A picture to be encoded may have interlaced structure (2 fields) or progressive structure. Even though a picture may be interlaced, video quality may be improved with frame coding when the correlation between odd and even fields is very strong.

- In H.264 and MPEG-2, **PAFF** coding is the best option as it dynamically chooses either frame or field coding.
- In HEVC, **SAFF** (Sequence Adaptive Frame Field) coding is the best option as it dynamically chooses either frame or field coding.

Use MBAFF

Only available with Extreme video codecs.

When **Frame** or **PAFF** modes are selected, you can check the **MBAFF** option (macroblock-adaptive frame-field) to encode the macroblock pairs of a single frame either in field mode or in frame mode.

User data

Use closed captions

Only available if Export type is ATSC.

Check this option if you want to extract and embed the closed caption information in the output stream.

NOTE: Compliant with the EIA-608 standard.

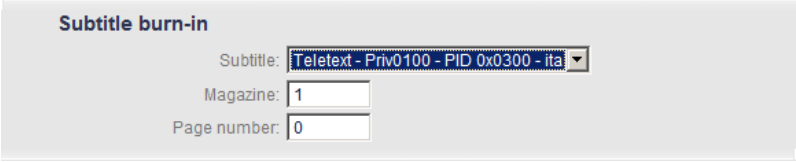
Insert AFD

Check this option if you want to take into account the display aspect ratio of the source and provide this information to the output.

Subtitle burn-in

The subtitle burn-in feature enables burning DVB teletext subtitle, DVB subtitle or Closed Caption streams in the video.

Figure 65. Subtitle burn-in: Teletext



1. Select the stream you want to burn.

NOTE By default the **Subtitle** parameter is set to **None**.

2. In case of Teletext, specify the Magazine and the page number.

Encoding Distribution

Encoded on: If you activated distributed encoding (*See "Configuring a TV profile" on page 157*), select the encoder role.

Possible values: **Master** or **Slave**

Always set this parameter to **Master**.

The value to choose depends on the configuration and the platform:

- When encoding UHD set this parameter to **Master**.
- When transcoding UHD set this to **Slave1**.

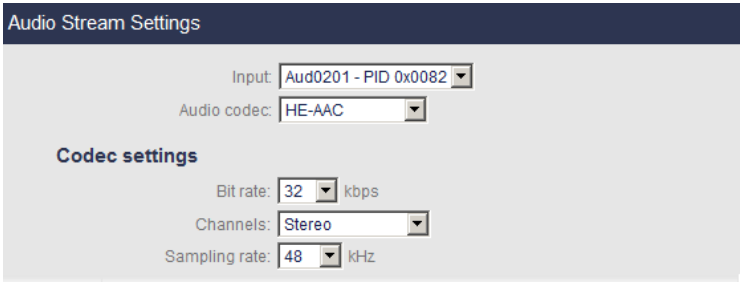
Distributed encoding for MBR is yet supported. In this case, the Encoding Distribution for each stream will have to be set in such way that the overall load is well distributed. Please contact Ericsson Support team for assistance.

Setting the audio stream parameters

NOTE Depending on license and export type, up to eight audio streams are allowed per profile.

To display the audio parameters, follow these steps:

1. Click the **Audio** stream subtab, the audio parameters are displayed in the right panel.



2. Select the appropriate value for each parameter.

Input Select the appropriate audio input channel.

None means video-only mode.

Audio codec Specify the audio codec.

Possible values: **AAC, HE-AAC, HE-AAC v2, MPEG 1 Layer II, Pass-through**, **Dolby Digital**, or **Dolby Digital Plus**.

Pass-through lets you receive an already encoded Dolby (DD/DD+) or AC-3/Enhanced AC-3 stream and forward it to the output. Supported format for **Pass-through** are:

- MPEG-1 Layer II (Ethernet only)
- AC-3, Enhanced AC-3 Audio (SDI AC-3 mode or Ethernet)
- Pass-through is not available when audio input is AAC

Codec settings

NOTES These parameters are not available with **Pass-through**.

For Dolby Digital and Dolby Digital Plus parameters, see *"Specific Dolby parameters" on page 173*.

Bit rate Specify the audio bit rate value in kilobits per second.
The specified value does not include the packet overhead. Possible values: See table below.

NOTE: Audio bit rate corresponds to "raw" audio bit rate, and does not include bit rate overhead due to encapsulation (such as LATM or ADTS headers, PES headers, MPEG-2 TS headers).

Channels Possible values: See table below.

Sampling rate Specify the sampling rate in kilohertz (kHz).
See table below.

Depending on the selected codec, the channel mode or the sampling rate can be disabled because they are not applicable. The following tables summarize this behavior.

Table 12. Channel modes depending on bit rate and codec

Codec type	Bitrate (kbps)	Channel
AAC	32, 40, 48, 56, 64, 80, 96, 112, 128, 160	Mono downmix/left/right, Dual-mono, Stereo
	192, 224, 256, 320	Dual-mono, Stereo
HE-AAC	32, 40, 48, 56, 64	Mono downmix/left/right, Dual-mono, Stereo
	80, 96, 112, 128	Dual-mono, Stereo
HE-AAC v2	16, 20, 24, 28, 32, 40, 48	Stereo
MPEG 1 Layer II	32, 48, 56, 64, 80, 96, 112, 128, 160, 192, 224, 256, 320, 384	Mono downmix, Dual-mono, Stereo, Joint stereo

Table 13. Sampling rate depending on codec, on bit rate and on channel

Codec type	Bitrate (kbps)	Channel	Sampling rates (kHz)
AAC	32, 40, 48	Mono downmix/left/right	32, 44.1, 48
	40,48	Dual-mono, Stereo	32
	56 and over	Mono downmix/left/right	32, 44.1, 48
HE-AAC	All	All	32, 44.1, 48
HE-AAC v2	All	All	32, 44.1, 48
MPEG-1 Layer II	All	All	32, 44.1, 48

Specific Dolby parameters

Bit rate
Specify the audio bit rate value in kilobits per second.
Possible values: from **96** to **640 kbps** for Dolby Digital, and from **64** to **1024 kbps** for Dolby Digital Plus

NOTE: Minimum bitrate for surround is **160 kbps**

Channels
Select the number of output channels.
Possible values: **Fixed Output 2/0** (stereo) or **Follow input** (maintains the number of channels).

NOTE: **Follow input** is only available if bit rate is higher than 128 kbps.

Sampling rate
Specify the sampling rate in kilohertz (kHz).

NOTE: Only 48kHz is supported.

LFE Enable
Only available with Follow input.
Check this option to activate the LFE (Low Frequency Effects) filter (*see* "LFE Lowpass Filter" on page 174).

Dialog Normalization
Possible values: **Auto** or, from **-1 dB** to **-31 dB**
Auto means that no processing will be done on the input.

Informational metadata for Fixed Output option

Dolby Surround EX Mode Possible values: **Auto**, **Not Indicated**, **NOT Dolby Surround EX Encoded**, or **Dolby Surround EX Encoded**

Preprocessing parameters for Fixed Output option

NOTE **Auto** means that no processing will be done on the input.

DC Highpass Filter Check this option to activate a DC highpass filter.

DRC Line Mode Profile Possible values: **None**, **Film: Standard, Film: Light, Music: Standard, Music: Light, Speech**

Enables Dynamic Range Control that restricts the absolute peak level for a signal.

DRC RF Mode Profile Possible values: **None**, **Film: Standard, Film: Light, Music: Standard, Music: Light, Speech**

Enables heavy Dynamic Range Control, ensures that the instantaneous signal peaks do not exceed specified levels.

Informational metadata for Follow input option

Dolby Surround EX Mode Possible values: **Auto**, **Not Indicated**, **NOT Dolby Surround EX Encoded**, or **Dolby Surround EX Encoded**

Preprocessing parameters for Follow input option

DC Highpass Filter Check this option to activate a DC highpass filter.

LFE Lowpass Filter Applies a 120Hz lowpass filter to the LFE (Low Frequency Effects) channel prior to encoding.

90 Degree Phase Shift Applies a 90-degree phase shift to the surround channels.

3 dB Attenuation Applies a 3 dB attenuation to the surround channels.

DRC Line Mode Profile Possible values: **Film: Standard, Film: Light, Music: Standard, Music: Light, Speech**

Enables Dynamic Range Compression that restricts the absolute peak level for a signal.

DRC RF Mode Profile Possible values: **Film: Standard, Film: Light, Music: Standard, Music: Light, Speech**

Enables heavy Dynamic Range Compression, ensures that the instantaneous signal peaks do not exceed specified levels.

Downmix Metadata for Follow input option

NOTE **Auto** means that no processing will be done on the input.

Lo/Ro Center Mix Level This parameter indicates the level shift applied to the center channel as a result of downmixing to an Lo/Ro output.

Lo/Ro Surround Mix Level This parameter indicates the level shift applied to the surround channels when downmixing to an Lo/Ro output.

Lt/Rt Center Mix Level This parameter indicates the level shift applied to the center channel as a result of downmixing to an Lt/Rt output.

Lt/Rt Surround Mix Level This parameter indicates the level shift applied to the surround channels when downmixing to an Lt/Rt output.

Stereo Downmix Preference Select the stereo downmix preference.

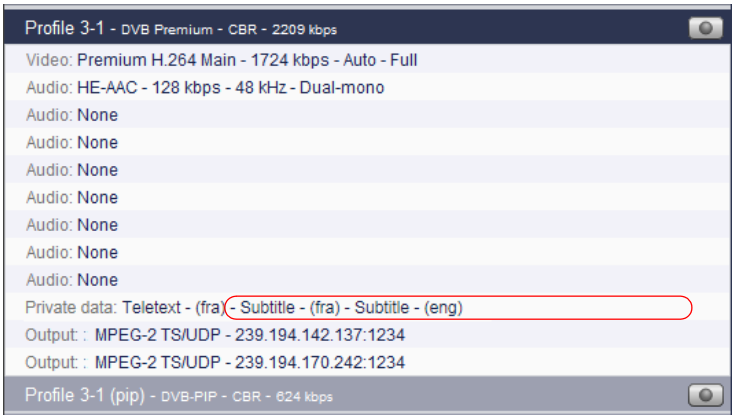
Possible values: **Not Indicated**, **Lt/Rt Downmix preferred**, **Lo/Ro Downmix preferred** or **Dolby Prologic II downmix preferred**[1]

1. Only available with Dolby Digital Plus audio codec.

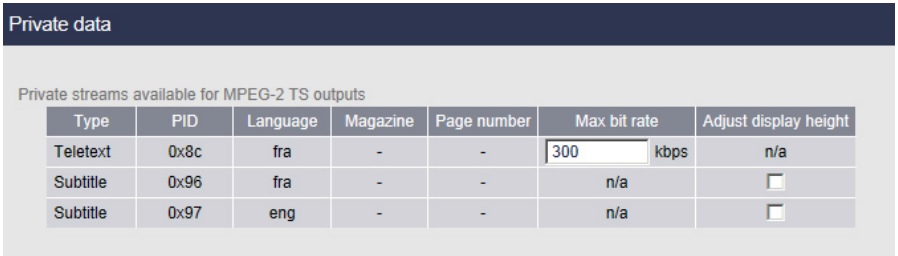
Setting the private stream parameters

To display the private stream parameters, follow these steps:

1. Set the **Export type** to **DVB** or **ATSC**. The **Private data** subtab is added to the profile subtabs.



2. Click the **Private data** subtab, the private stream parameters are displayed in the right panel.



The private stream management depends on the selected mode: **Ethernet**/**On-board**).

- In **Ethernet** mode, you cannot modify the input PID. Teletext and Subtitle information is extracted from the input
- In **On-board** mode, only one private stream is available, Teletext and two additional Teletext subtitle lines are displayed. You cannot add additional information.

The default PID value is 0x80 and only the Teletext PID can be changed. The Teletext subtitle PID is not displayed as it is not a separate stream.

3. The following information is displayed for each private stream:

Type	Possible values: **Teletext**, **Subtitle, SCTE-35**
PID	PID of the input private stream.
Language	Input language.
Magazine	Magazine number.
Page number	*Only displayed in On board mode*. Page number.
Max bit rate	*Only available with teletext*. You can define the bandwidth reserved for the teletext DVB data. Possible values: from **30** to **600** kbps
Adjust display height	Check this option to adjust the DVB subtitle display height.

4. Check the private streams that will be sent to the output(s).

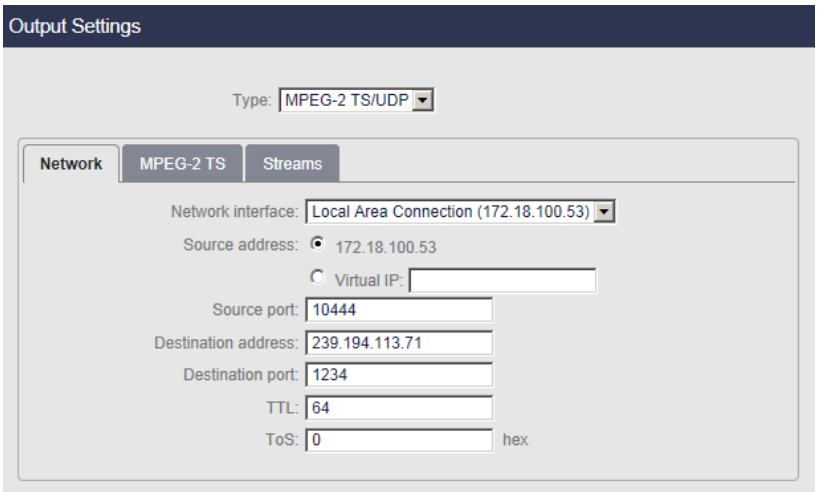
Setting the output parameters

To display the output parameters, follow these steps:

1. Click the **Output** subtab, the output parameters are displayed in the right panel.

NOTE Output parameters are displayed in several tabs that you have to fill one after the other.

Figure 66. Output parameters: MPEG-2 TS over UDP



2. Select the appropriate value for each parameter.

NOTE Depending on the export type, you can configure one or two outputs independently.

Type Select the broadcast type.

Possible values: **None**, **MPEG-2 TS/RTP** or **MPEG-2 TS/UDP**

Depending on the selected output type, specific parameters are displayed.

NOTE: MPEG-2 TS is compliant with ISO/IEC IS 13818-1.

Network

Network interface

Select the appropriate network interface.

Possible values: **Ethernet 1, Ethernet 2, Ethernet 5** and **Ethernet 6, Ethernet 1 (backup: Ethernet 2),** or **Ethernet 6 (backup: Ethernet 5).**

Select **Ethernet 1 (backup: Ethernet 2)** –or **Ethernet 6(backup: Ethernet 5)**– if you want redundant Ethernet interfaces. In this case, Ethernet 1 will be used to stream by default, but if Ethernet 1 fails, then Ethernet 2 will be used instead.

Source address

You can either select the Ethernet address configured via the web interface (see "Configuring the Ethernet connectors" on page 31) or enter a virtual IP address.

NOTE: Using a virtual interface is a convenient way to decorrelate the streaming from the hardware that is doing it, that is the Ethernet connectors. Virtual IP addresses are useful:

- When two network interfaces are bonded, for preserving the stream when a network interface fails (no need to replace the source IP with the backup network interface)
- In general, for preserving the stream when using backed up encoders

The virtual IP address MUST be on the same subnet as the primary IP it virtualizes. Otherwise, an error is displayed.

When the virtual IP applies to **Ethernet 1 (backup: Ethernet 2)**, both network interfaces 1 and 2 should be on the same subnet, otherwise a warning is displayed.

Source port

Source port for streams.

Possible values: from **1024** to **65534** (even number)

Destination address

The first time, a random multicast IP address is initially set, you can modify this address by entering a unicast or multicast address.

NOTES: A multicast address is between **224.0.0.0** and **239.255.255.255**.

Destination port

Destination port for streams.
Note that if you are using RTP, the RTCP stream will be incremented from this port number, for example:
RTP stream: **5020,** RTCP stream: **5021**
Possible values for RTP: from **1024** to **65534** (even)

Possible values for UDP: from **1024** to **65535**

TTL TTL stands for Time To Live.

A value in the range 0 through 255 defines the scope within which multicast packets should be sent over a network using Internet Protocol (ιπ).
Each router decrements the TTL by one. When the value reaches a predefined lower limit, the router throws the packet away. By default the TTL is set to 64.

ToS ToS stands for Type Of Service byte (for QOS purpose).

Possible value: numerical value (3 hexadecimal bytes), from **0** to **0x3f**.

A flag is added to the IP packet headers to show which kind of information is embedded in the IP stream so that switches can identify the traffic type without having to understand the traffic.

MPEG-2 TS settings

PMT PID PID (Packet IDentifier) of the PMT (Program Map Table) stream.

Possible values: from **0x0010** and **0x1ffe** (hexadecimal) or from 16 to 8190 (decimal).

PCR PID PID of the PCR, Program Clock Reference used to synchronize the video and audio packets.

Possible values: from **0x0010** and **0x1ffe** (hexadecimal) or from 16 to 8190 (decimal).

Target PCR Period PCR period (in milliseconds)

Possible values: from **20** to **1000** ms.

Target PSI Period PSI period (in milliseconds)

Possible values: from **20** to **1000** ms.

Insert TOT & TDT Check this option to generate a TOT/TDT table. The TOT/TDT tables are used to insert time information into MPEG-2 TS streams.

If you check this option, you must define the **Target TOT/TDT period** (from **1** to **30** s).

Program number Program identifier.

Possible values: from **0x0001** to **0xffff**.

Service name Service name in SDT.

Service provider Provider of the program stream.

Discard stream lost packets Check this option if you want that MFVP Encoding live drops all video and audio packets in case of input loss

Use LATM encapsulation for AAC By default, AAC audio is MPEG-2.
Check this option if you want AAC audio to be MPEG-4.

Align video frames to PES packets Check this option to align video frames to PES packets.

NOTE: This option guarantees the interoperability with set-top boxes but increases the bandwidth.

Transport streams

You can specify the composition of the output stream by selecting the video, audio and private streams that will be sent to the selected output.

NOTE The number of audio streams depends on the license.

Audio PID MPEG-2 TS parameter. PID of the audio stream.
Possible values: from **0x0010** and **0x1ffe** (hexadecimal)

NOTES: The audio PID defined is for the first audio. The second is increased by +1 and so on.

Video PID *MPEG-2 TS parameter*. PID of the video stream.
Possible values: from **0x0010** and **0x1ffe** (hexadecimal)

Private PID *MPEG-2 TS parameter*. PID of the private stream.
Possible values: from **0x0010** and **0x1ffe** (hexadecimal)

PIDs consistency rules

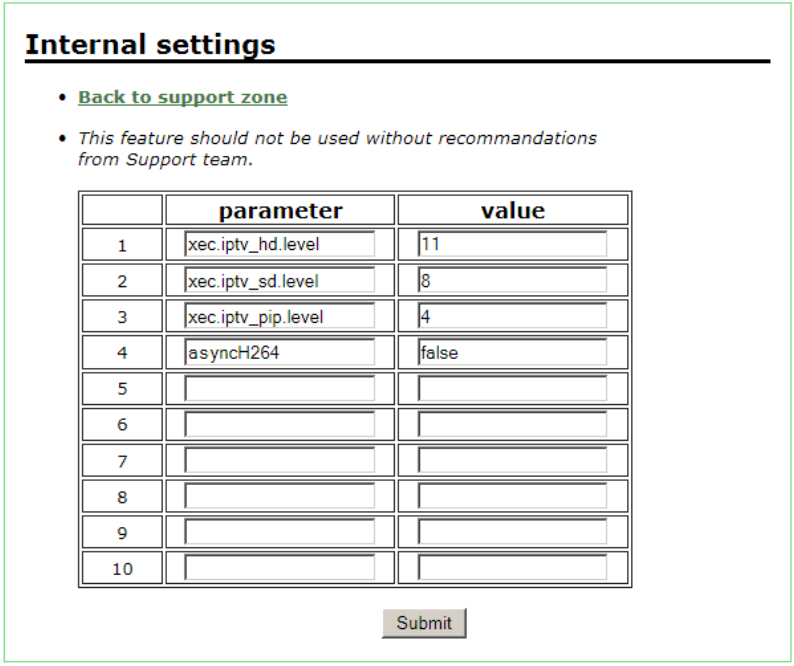
- PID must be between **0x0010** and **0x1ffe**.
- PMTPID must be different from video and audio PIDs.
- Video PID must be different of audio PID.
- PCR PID can be equal to PMT PID or to Video/Audio PIDs.

Specific Mediaroom parameters

To be compliant with Mediaroom, you need to configure specific parameters using advanced configuration settings.

To configure the **key seed**, follow these steps:

1. Open a browser and enter the following address:
   `http://<MFVP-EncodingLive_IP_Address>/Support.html`
2. Click the **Advanced configurations** green link at the top of the page.
3. Enter the following parameters:
   `xec.iptv_hd.level=11`
   `xec.iptv_sd.level=8`
   `xec.iptv_pip.level=4`
   `asyncH264=false`



Starting/Stopping encoding

The start/stop button, at each profile level, lets you respectively start and stop the current live broadcast.

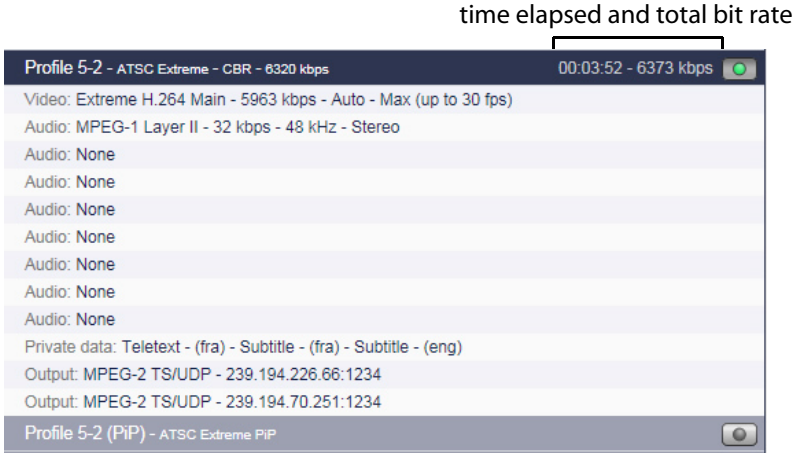
- To start encoding a profile, click the button.
- To stop encoding a profile, click the button.

When you start encoding the following information is displayed in the profile left panel.

The following parameters are regularly updated:

- time elapsed since the beginning of the encoding,
- total bit rate (in kbps),

Figure 67. Encoding information



Chapter 7

Configuring a Mobile Streaming profile

Basic workflow

Below is the basic workflow for generating a mobile streaming stream.

Step #1: Set the service parameters

- Enter a service name
- Select the input type used to capture the video stream.
  If **On Board** input is selected, select the input type and the standard
  If **Ethernet** input is selected, select the Ethernet interface, then enter the source IP address and port where your input source is broadcasting, and select the audio and video PIDs detected in the stream

Step #2: Set the profile parameters

- Enter a profile name
- Select the export type and the bit rate

Step #3: Set the profile video and audio parameters

- Select the audio and video input
- Select the audio and video codecs

Step #4: Set the profile output parameters

- Select the transport protocol, then set the output parameters

Step #5: Start the live broadcast session

Step #6: Display alarms

CAUTION When you modify a parameter and you press ENTER, the modification is automatically saved in the current configuration.

Setting up the encoding profiles

Depending on license, up to four services can be displayed and a service can include up to 32 profiles that can be configured and started independently one from the other profiles.

Creating a profile

By default only one profile by service is already activated and configured. To create a new profile, you can either add a new profile or duplicate an existing one.

Adding a new profile

To add a profile, click the **Service** tab, then select the **Add profile** button in the right panel. MFVP Encoding live creates a new profile using default parameters.

Duplicating a profile

To duplicate a profile, follow these steps:

1. Click the **Profile** tab of the profile you want to duplicate, then select the **Duplicate** button in the right panel.
2. MFVP Encoding live creates a new profile using the selected profile parameters.

Deleting a profile

To delete a profile, click the **Profile** tab, then click the **Delete** button in the right panel. Note that all the profile parameters will be lost.

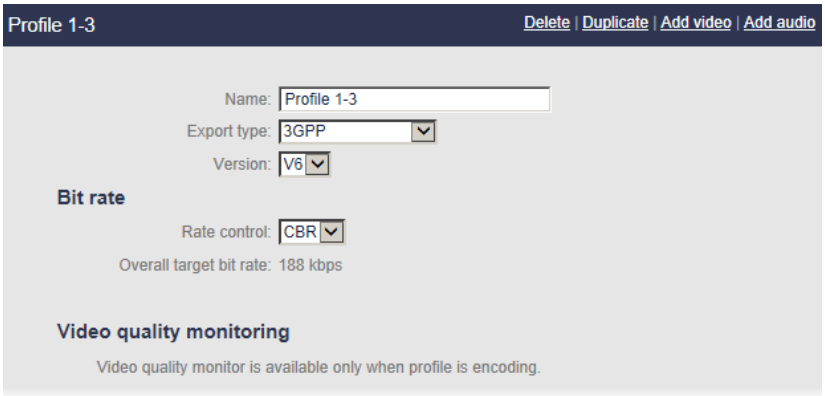
Setting the profile general parameters

NOTE During encoding, all the parameters are greyed and cannot be modified.

To set the profile general parameters, follow these steps:

1. Click a **Profile** tab. The profile general parameters are displayed in the right panel.

Figure 68. Profile 1-1 general parameters



2. Select the appropriate value for each parameter.

Name Enter a profile name. We recommend using a name that clearly identifies the channel.

Export type Select **3GPP** among the possible values.

If a codec-linked restriction is not respected, an error message is displayed when you start encoding.

Export type Version Select the 3GPP version.

Possible values: **V6** [1] or **V5**.

1. If you select 3GPPv6, you can have up to four video streams and two audio streams per multirate profile.

Bit rate control

For more information, see "Advanced Rate Control" on page 205.

Rate control

Possible values: **CBR**, **VBR**, or **ABR**.

NOTES: In case of H.264, the stream remains compliant with the normative HRD.

- With **CBR** (Constant bit rate), you can stream content over a limited bit rate channel such as a network. The output stream fits in one bit rate, which you specify as a parameter. Constant bit rate means that the bit rate is constant according to the leaky bucket concept (*see "Appendix G" on page 266.*).
- With **VBR** (Variable bit rate), the output stream will never exceed the Overall max bit rate, and will try to reach the Target network bit rate. This is usually used to save bandwidth. This model is also called "Capped VBR".
- With **ABR** (Available Bit Rate), MFVP Encoding live will never exceed the maximum bit rate, but may go lower than the Target network bit rate if no more bit rate is needed for encoding.

Overall target bit rate

Set the overall bandwidth target (in kbps) used by the stream.

Instead of configuring the video and audio bandwidth separately, which, added to the network overhead, constitute the overall bandwidth, you only have to specify the overall bandwidth and the audio bit rate. The video bit rate is computed from these two values.

NOTES: In **3GPP v6**, the Overall target bit rate is automatically calculated from the video and audio target bit rates and cannot be modified.

NOTES: In RTP, the total bit rate is the resulting IP bit rate (it includes all overhead down to and including IP headers).

Overall max bit rate

VBR mode only.

Display the maximum overall bit rate (in kbps).

Video quality

When encoding a profile, you can display a graphical representation of the video quality (see "Video quality" on page 41) by clicking the **video quality monitor** link.

Image Overlay

See "Image overlay" on page 88.

Displaying the video, audio, and output parameters

You can quickly check the video, audio, and output parameters of a profile, by clicking the **Profile #** tab. The selected profile subtabs are displayed below.

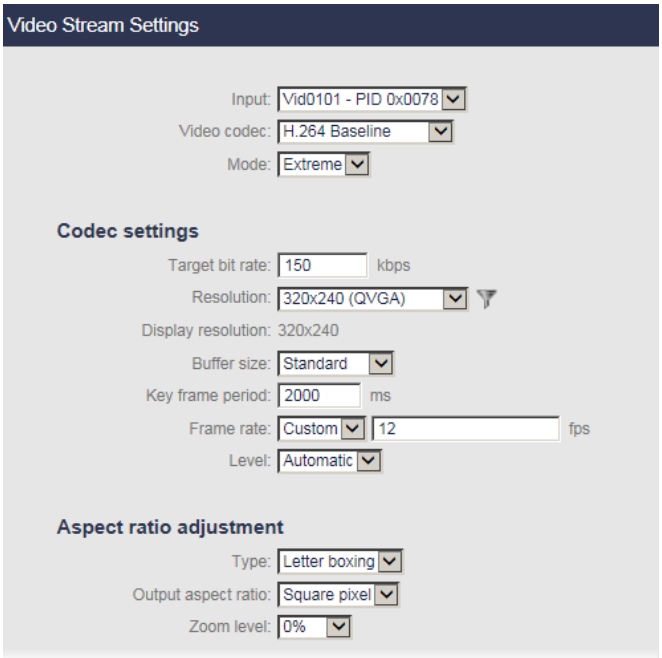
Figure 69. Profile subtabs

Setting the video stream parameters

To display the video parameters, follow these steps:

1. Click the **Video** stream subtab, the video parameters are displayed in the right panel.

Figure 70. Profile video parameters: Extreme H.264 Baseline video codec



2. Select the appropriate value for each parameter.

Input Select the appropriate video input channel
None means audio-only mode.

Video codec Select the video codec.
Elite H.264 Baseline, Extreme H.264 Baseline, MPEG-4 SP, H.263

Mode Only available with 3GPPv6 export type and H.264 Baseline codec.
Possible value: **Extreme**

Codec settings

Target bit rate Specify the video bit rate value in kilobits per second.
The specified value does not include the packet overhead.

Possible values: from 20 kbps to 1000 kbps.

Resolution Select a resolution value.

Possible values: *see Table 14 below*.

NOTE: Resolutions are sorted by the total number of pixels per frame.

Table 14. Possible resolutions per video codec

Resolution Width	x Height	MPEG-4 SP	H.263	H.264(1)
480	360			x
480	352			x
480	320			x
480	270			x
384	288			x
320	320			x
352	288	x	x	x
416	240			x
400	224			x
352	240	x		x
320	240	x		x
320	192			x
320	180	x		x
240	240			x
320	176	x		x
240	192	x		x
240	176	x		x
220	176	x		x
240	160			x
176	144	x	x	x
176	128	x		x
176	120	x		x
144	140	x		x

Resolution Width	x Height	MPEG-4 SP	H.263	H.264[1]
160	120	x		x
160	90	x		x
128	96	x	x	x
144	80	x		x
80	64	x		x

1. Elite and Extreme H.264 Baseline codecs

Display resolution Information field. Depends on the aspect ratio management and on the encoding resolution.

Buffer size *Only available with 3GPPv6 export type and H.264 codecs.*

Possible values: **Low delay**, **Standard** or **High quality**

Defines the size of the VBV (Video Buffering Verifier). If the mobile device is not performing correctly (desynchronization), you can adjust the delay settings.

Low delay: VBV size = 1 second
Standard: VBV size = 2 seconds
High quality: VBV size = 5 seconds

Key frame period Possible values: **1 000** to **10 000** ms

Enter the maximum time between two key frames. You can specify how often you want key frames to be inserted into the video stream. Key frames are inserted into the stream periodically to synchronize the decoder and enable it to recover from errors.

Refreshing the image more often (by setting a shorter key frame period) reduces the recovery time but requires a higher bit rate to maintain encoding quality.

Frame rate Possible values: **Full** (up to 25/30 fps), **Half**, **Quarter** or **Custom**

Custom: enter a specific value from **5** to **25** fps (PAL input) or from **5** to **29.97** fps (NTSC input).

Level The level is a measure of the projected decoding complexity of the file or stream. Because video codecs have level definitions with overlapping underline settings, you can create content with the right level of compatibility with your target device.

Possible values for H.263 codec:
Automatic, or **10**, **20**, **30**

Possible values for MPEG-4 SP codec:
Automatic, or **0**, **0b**, **1**, **2**, **3**

Possible values for H.264 codec:
Automatic, or **1**, **1b**, **1.2**, **1.3**, **2**, **2.1**, **2.2**, **3**

Aspect ratio adjustment

Select the aspect ratio adaptation type (*see Appendix D for more details*).

NOTE We recommend launching the encoding, and modify the **Aspect ratio adjustment** value to adjust it with accuracy.

Possible values: **Letter boxing** or **Stretching**.

- If you select **Letter boxing**, you can define **Output aspect ratio** (square pixel, 4/3 or 16/9) and the **Zoom level** (in%).
  Possible values for Zoom level: 0 (no zoom, black stripes on the top and bottom), 10, 20, 30, 40, 50, 60, 70, 80, 90, 100 (the video is zoomed, and black stripes are removed).
- If you select **Stretching**, you can define the **Output aspect ratio** (square pixel, 4/3 or 16/9) and the stretching values (in%) for Top, Bottom, Right and Left.

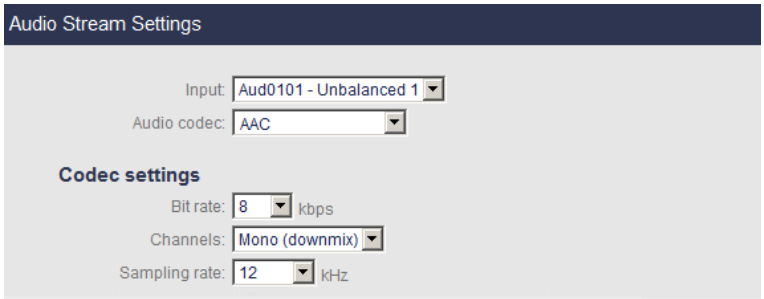
Setting the audio stream parameters

NOTE Depending on license, one or two audio streams are allowed per profile.

To display the audio parameters, follow these steps:

1. Click the **Audio** stream subtab, the audio parameters are displayed in the right panel.

Figure 71. Audio parameters: MPEG-4 AAC codec



2. Select the appropriate value for each parameter.

Input Select the appropriate audio input channel.
None means video-only mode.

Audio codec Specify the audio codec.
Possible values: *see Table 15 below*

Table 15. Audio codecs depending on Export type

Export type	Audio codec
3GPPv5	AMR-NB, AMR-WB or AAC
3GPPv6	AMR-NB, AMR-WB, AAC, HE-AAC, or HE-AAC v2

Codec settings

Bit rate Specify the audio bit rate value in kilobits per second.

Possible values: *see Table 16 below*.

NOTE: Audio bit rate corresponds to "raw" audio bit rate, and does not include bit rate overhead due to encapsulation (such as LATM or ADTS headers, PES headers, MPEG-2 TS headers).

Table 16. Bit rates depending on codec

Audio codec	Bit rate
AMR-NB	4.75, 5.15, 5.9, 6.7, 7.4, 7.95, 10.2, 12.2
AMR-WB	6.6, 8.85, 12.65,14.25, 15.85, 18.25, 19.85, 23.05, 23.85
AAC	8, 10, 12, 16, 20, 24, 28, 32, 40, 48, 56, 64, 80, 96, 112, 128, 160, 192, 224, 256, 320
HE-AAC	8, 10, 12, 16, 20, 24, 28, 32, 40, 48, 56, 64, 80, 96, 112, 128
HE-AAC v2	16, 20, 24, 28, 32, 40, 48

Channels Specify the channel mode.

Possible values: *see Table 17 below*.

Sampling rate Specify the sampling rate in kilohertz (kHz).

Possible values: *see Table 17 below*.

Table 17. Channel modes and sampling rate depending on bit rate and codec

Codec type	Bit rate (kbps)	Channel mode	Sampling rate
AMR-NB	All bit rates	Mono downmix Mono left Mono right	8
AMR-WB	All bit rates	Mono downmix Mono left Mono right	16

Codec type	Bit rate (kbps)	Channel mode	Sampling rate
AAC	8 and 10	Mono downmix Mono-left Mono-right	8, 11.025, 12
	12	Mono downmix Mono left Mono right	8, 11.025, 12, 16
	16	Mono downmix Mono left Mono right	8, 11.025, 12, 16, 22.05, 24
	20	Mono downmix Mono-left Mono-right	11.025, 12, 16, 22.05, 24
	24 and 28	Mono Mono left Mono right	11.025, 12, 16, 22.05, 24, 32
		Dual mono Stereo	11.025, 12, 16, 22.05, 24
	32	Mono Mono left Mono right	11.025, 12, 16, 22.05, 24, 32, 44.1, 48
		Dual mono Stereo	11.025, 12, 16, 22.05, 24
	40	Mono Mono left Mono right	16, 22.05, 24, 32, 44.1, 48
		Dual mono Stereo	16, 22.05, 24, 32
	48	Mono Mono left Mono right	22.05, 24, 32, 44.1, 48
		Dual mono Stereo	22.05, 24, 32

Codec type	Bit rate (kbps)	Channel mode	Sampling rate
AAC	56	Mono Mono left Mono right Dual mono Stereo	22.05, 24, 32, 44.1, 48
	64 and over	Mono Mono left Mono right Dual mono Stereo	32, 44.1, 48
HE-AAC	8 to 20	Mono left Mono right Mono downmix	For 8, 10, 12 kbps: **16, 22.05, 24, 32** For 16 kbps and over: **32, 44.1, 48**
	24 to 64	Mono left Mono right Mono downmix Dual mono Stereo	
	80 and over	Dual mono Stereo	
HE-AAC v2	All bit rates	Stereo	32, 44.1, 48

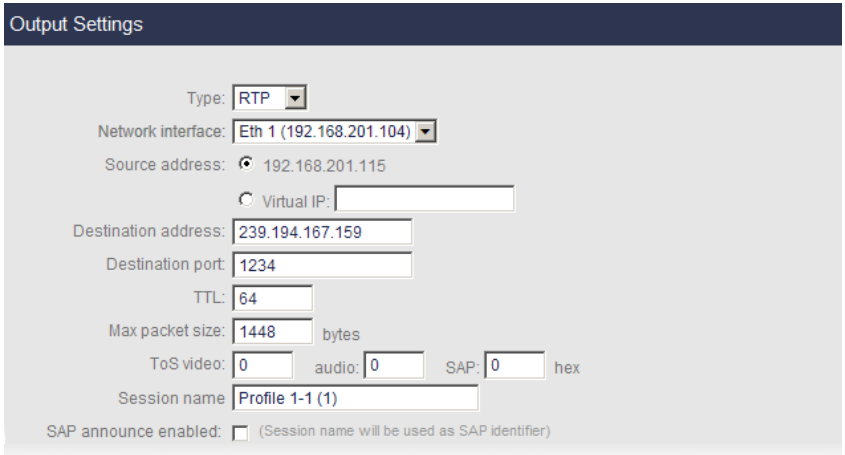
Setting the output parameters

NOTE Two outputs are allowed per profile.

To display the output parameters, follow these steps:

1. Click the **Output** subtab, the output parameters are displayed in the right panel.

Figure 72. Network parameters: RTP



2. Select the appropriate value for each parameter.

Network parameters

Type Select the broadcast type.

Possible values: **None**, **RTP**, **RTSP** or **RTSP announce (Automatic Unicast)**

Depending on the selected output type, specific parameters are displayed.

Network interface Select the appropriate network interface.

Possible values: **Ethernet 1, Ethernet 2, Ethernet 5, Ethernet 6**, **Ethernet 1 (backup Ethernet 2),** or **Ethernet 6 (backup: Ethernet 5)**

CAUTION: The selected Ethernet interface must be configured at the **System** level (*see page 31*).

Source address You can either select the Ethernet address configured via the Web interface or enter a virtual IP address.

NOTE: Using a virtual interface is a convenient way to decorrelate the streaming from the hardware that is doing it, that is the Ethernet connectors.
Virtual IP addresses are useful:

- When two network interfaces are bonded, for preserving the stream when a network interface fails (no need to replace the source IP with the backup network interface)
- In general, for preserving the stream when using backed up encoders

CAUTION: The virtual IP address MUST be on the same subnet as the primary IP it virtualizes. Otherwise, an error is displayed.

When the virtual IP applies to **Ethernet 1 (backup: Ethernet 2)**–or Ethernet6 (backup: Ethernet5)–, both network interfaces 1 and 2 should be on the same subnet, otherwise a warning is displayed.

Specific RTP output parameters

Destination address The first time, a random multicast IP address is initially set, you can modify this address by entering a unicast or multicast address.

A multicast address is between **224.0.0.0** and **239.255.255.255**.

Destination port Destination port for streams.
Note that if you are using RTP, the RTCP stream will be incremented from this port number, for example:

RTP stream: **5020**
RTCP stream: **5021**

Possible values: from **1024** to **65534** (even)

TTL TTL stands for Time To Live.

A value in the range **0** through **255** defines the scope within which multicast packets should be sent over a network using Internet Protocol (ιπ).

Each router decrements the TTL by one. When the value reaches a predefined lower limit, the router throws the packet away.

By default the TTL is set to **128**.

Max packet size Defines the maximum size of the RTP packet (in bytes). It includes payload and RTP header.

Possible values: from **300** to **65495**

See page 201 for calculating the maximum packet size.

ToS video, audio & SAP **ToS** stands for Type Of Service byte (for QOS purpose).

Possible value: numerical value (3 hexadecimal bytes). One separate flag can be set for video/audio/SAP packets.

A flag is added to the IP packet headers to show which kind of information is embedded in the IP stream so that switches can identify the traffic type without having to understand the traffic.

Session name Session name used to access the live broadcast when using SAP. A default SAP session name is initially set.

Enable SAP announce Check this option to activate SAP announce.

Content protection

This parameter is reserved for future use.

Specific RTSP output parameters

Listen port The port the encoder will listen on for connections.

Possible values: from **1** to **65535**

Session name A default session name is initially set.

Stream name The stream name that will be exposed through any server that connects.

Max packet size Defines the maximum size of the RTSP packet (in bytes). It includes payload and RTSP header.

Possible values: from **300** to **65535**

See page 201 for calculating the maximum packet size.

Specific RTSP Announce (Automatic Unicast) output parameters

Publishing point Enter the IP address of your publishing point.

URL: rtsp://<server address>/

NOTE: If you defined credentials to access this URL (*see* "Adding a credential" on page 235), the user icon, should appear in color.

Session name A default SAP session name is initially set.

Stream name The stream name that will be exposed through any server that connects.

Max packet size Defines the maximum size of the RTSP packet (in bytes). It includes payload and RTSP header.

Possible values: from **300** to **65536**

See below for calculating the maximum packet size.

Defining the maximum packet size

To define the maximum packet size, follow these steps:

1. Identify the MTU size of your network.
2. Identify the packet structure used on your network.
3. Following your streaming configuration, subtract the size of the packet headers from the MTU to get the size available for the RTP packet.

Example 2. RTP streaming over UDP over Ethernet:

The MTU size is 1500, and the packet structure is as follows:

20 bytes IP header	8 bytes UDP header	12 bytes RTP header	N bytes RTP payload

The value to set for Max. packet size when streaming over UDP is:

1500 - 20 - 8 = 1472

Table 18. Some packet header sizes

Header	Size (in bytes)
IP	20
UDP	8
TCP	28
RTP	12
RTSP interleave	4
PpoE	8

Table 19. Some typical MTU sizes

RFC	Description	MTU
894	Minimally required	68
1051	ARCNet	508
1356	X.25, ISDN	576
1055	Serial Line IP (SLIP)	1 066
1042, 2516	IEEE 802.3 / 802.2, PPPoE	1 492
894, 895	Ethernet	1 500
1390	FDDI	4 352
1042	4 Mbit Token Ring	4 464
1042	802.4 Token Bus	8 166
none	16 Mbit Token Ring	17 914
1374	HIPPI	65 535

Starting/Stopping encoding

The start/stop button, at each profile level, lets you respectively start and stop the current live broadcast.

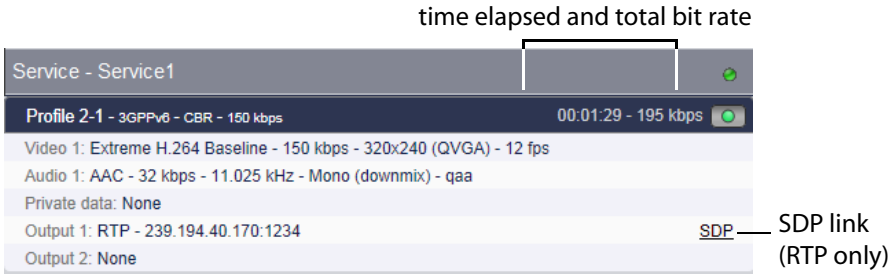
- To start encoding a profile, click the button.
- To stop encoding a profile, click the button.

When you start encoding the following information is displayed in the profile left panel.

The following parameters are regularly updated:

- time elapsed since the beginning of the encoding,
- total bit rate (in kbps): displays the bit rate measured on the network.

Figure 73. Encoding information



Previewing the live broadcast

When you use RTP, an **SDP** link is created when you start encoding (*see Figure 73 on page 203*). This link lets you access the SDP file.

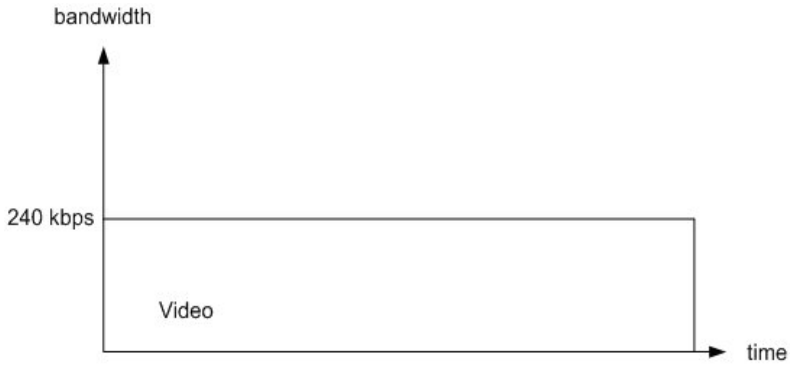
Chapter 8

Advanced Rate Control

CBR mode

In CBR mode, the total rate is set and always maintained.

Figure 74. CBR mode



Example:

Over a Satellite link limited to 1.5 Mbps, the encoder can be set at a target bit rate of 1.4 Mbps. We guarantee that the encoder will never go over or under the expected bit rate.

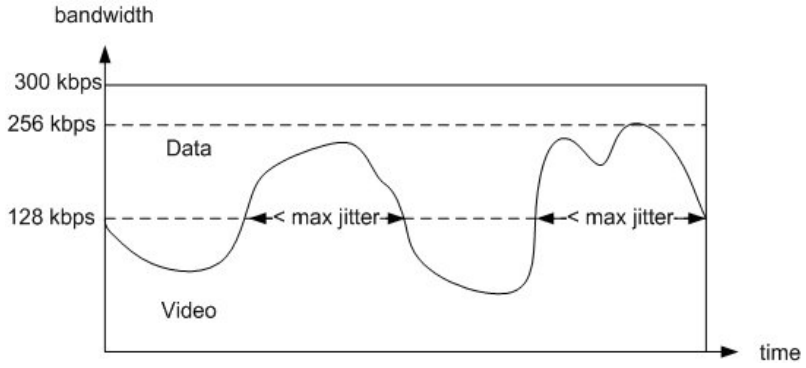
Benefit:

This mode can be used in video contribution or any other application where a limited bandwidth is fully allocated to the video feed. By maintaining the data rate constant, you can easily check that the feed is not dropping. This mode is particularly useful with IP Satellite modems.

Capped VBR mode

In IPTV, the capped VBR bit rate allows a user to set a nominal **Target bit rate** (average bit rate over time) and a maximum instantaneous bit rate, **Max bit rate**. The target bit rate is guaranteed over a configurable time window.

Figure 75. Capped VBR Mode



Example:

Over a DSL line limited to 4 Mbps, MFVP Encoding live **Target bit rate** can be set at 2 Mbps with a 3.5 Mbps **Max bit rate**. It guarantees that the maximum excursion above the targeted bit rate will not exceed a certain time, called the maximum jitter (usually set around 1 sec.).

The remaining bandwidth available for asynchronous data is equal to 2 Mbps in average, but can be lower. The jitter created by this bandwidth reduction will not exceed the maximum jitter defined above.

Benefit:

This mode can be used in IPTV deployments or shared bandwidth networks with heterogeneous data. With a slight impact on the data jitter, one can achieve better video quality results while not affecting the overall data transmissions[(1)].

1. Other capped VBR implementations do not guarantee this maximal jitter. The video could take over completely the bandwidth originally set for data.

Optimized Available Bit Rate mode (based on VBR)

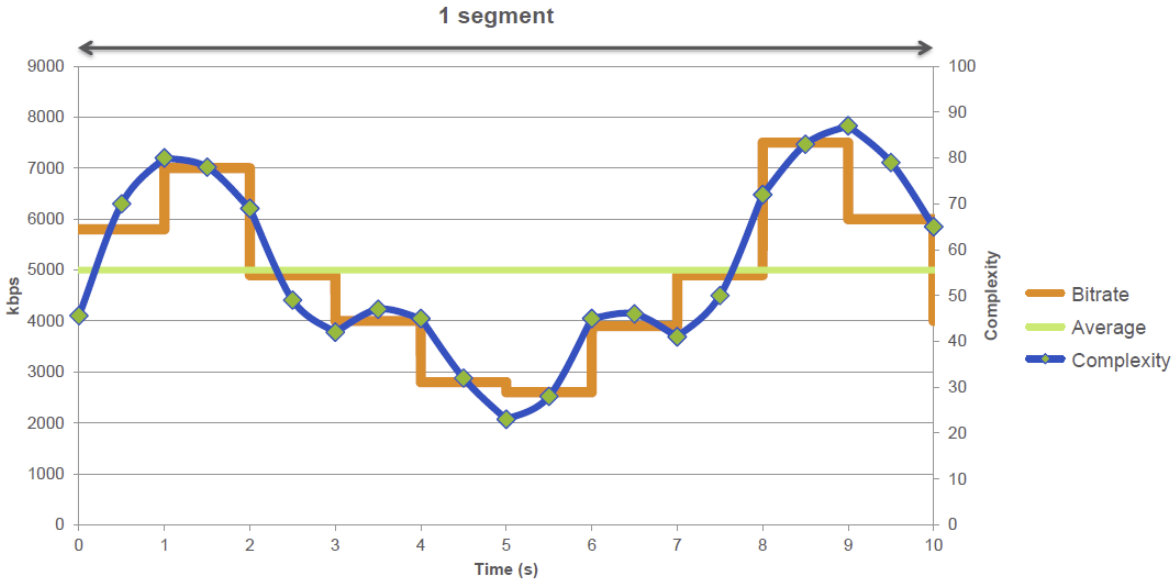
NOTE This mode is only available in Internet TV.

In Standard ABR (Available Bit Rate), each segment targets a constant bitrate, whatever the complexity.

"Optimized ABR" targets at delivering constant video quality, i.e. proportionate the bandwidth allocation to complexity, **PER segment**, thus avoiding to over-allocate bitrate on simple sequences.

"Optimized ABR" is an optimization technique based on using non constant bitrate inside an ABR segment, but keeping average network bitrate constant.

Figure 76. Optimized ABR Mode



"Optimized ABR" implements a specific look-ahead processing that enables deliver best overall quality for each segment. It needs to provision an additional delay to enable the lookahead. It is advised to set the segment duration (*see "GOP duration" on page 102*) to a value around 8-10 s seconds to have maximum benefit of the technology. The target bit rate (*see "Target bit rate" on page 99*) will be respected along this duration. Additional delay is then equal to Segment Duration in seconds minus 1.

Channels that alternate complex and simpler sequences benefit most of "Optimized ABR", such as movies or general purpose channels.
Due to additional delay, live Sports channels may choose not to adopt it.

Benefit:

"Optimized ABR" is a Bandwidth optimization technique, enabling to save up to 50% bandwidth on a CDN network, while keeping the video quality.

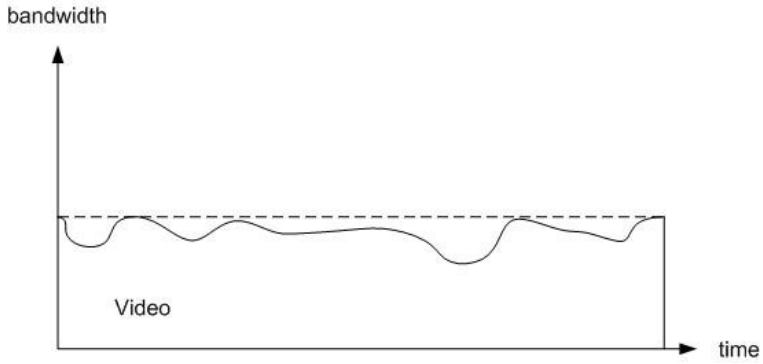
"Optimized ABR" is based on measuring the scenes complexity over time and allocating bitrate accordingly, for each ABR segment. Technology tunes up the bandwidth consumption to JUST what is needed to achieve the targeted quality.

This gives statistical benefits and enables substantial bandwidth savings.

Available Bit Rate (ABR) mode

This mode is obtained by setting the **Rate control** parameter to **ABR**. In this configuration, MFVP Encoding live will never exceed the target bit rate, but may go lower than the targeted bit rate.

Figure 77. ABR Mode



Benefits:

This mode can be used in mobile streaming applications or any application not requiring a full bandwidth allocation. This mode uses the necessary amount of bits to transmit video. The savings in bit rate can reduce the communication costs (for example, extended battery life for a cell phone) while always respecting the maximum channel capacity.

Combined with ToS (Type of Service) tagging, this mode can also be used to mix video with asynchronous data over a fixed capacity channel. The asynchronous data is opportunistically sent when the video does not take all the bandwidth.

StatMux, statistical rate control

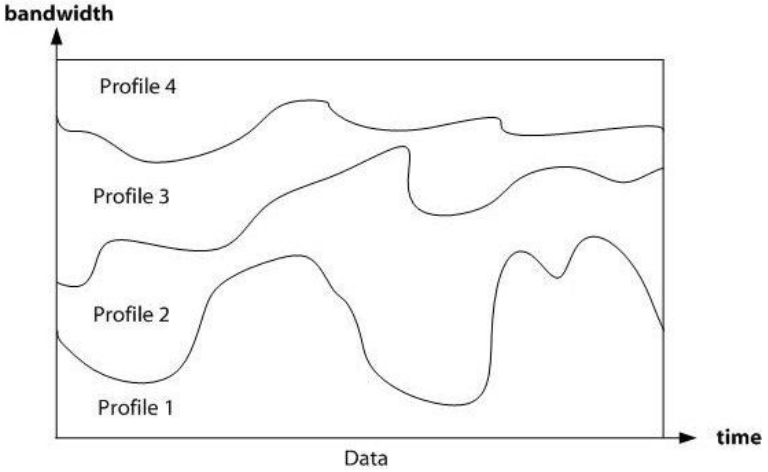
To answer to the requirement to fit multiple channels in a bandwidth-constrained network optimally, MFVP Encoding live offers an IP-based, statistical rate control module for broadcast mobile TV networks providing the highest quality video services.

By adding statistical rate control capabilities within the core of MFVP Encoding live, we enables service providers to both optimize the use of their bandwidth, and also avoid the cost of purchasing proprietary statmux technology.

The latter case is more suited to video transmission. Given the heterogeneous nature of video signals, it is more appropriate to use more bandwidth on parts of the video that are hard to encode and less on the easier parts.

Statistically, there is a high chance that different sources do not need high bandwidth at the same time, so there is an interest in dynamically assigning the bit rate for the transmission of each source.
This method is called Statistical Variable Bit Rate mode, or Statistical VBR, and the allocation mechanism Statistical Rate Control.

Figure 78. Aggregated bandwidth of 4 profiles in Statistical VBR



The bandwidth savings in this case are fairly significant. A 50% bit rate reduction has been measured with 8 profiles.

Benefits:

With scarce spectrum resources, the use of statistical rate control in DVB-H allows doubling the number of channels available. The overall subjective quality of the service is also greatly improved.

How it works?

In MFVP Encoding live, a profile set to work in statistical rate control mode must be assigned to a statmux group. This group represents all the profiles which are acting under the same statistical rate control and sharing the bandwidth.

The group receives the information from all the profiles and sends the dynamic bit rate allocation instructions to the pool. The group is in charge of the overall allocation.
The communication between the profiles is IP-based. In fact, all the profiles from the same statmux group communicate in a single multicast group.

Statistical rate control redundancy

To prevent from a potential failure of the statistical rate control, it is advised to set two groups. In this case, the first started group is acting as “Main” group while the second one is acting as backup.

If the main group fails, the backup group preempts the control of the bit rate and becomes Main. An alarm is raised when this preemption happens.

In this configuration the statistical rate control is protected by 1:1 redundancy.

NOTE Statmux profiles can be either on the same encoder as the statmux group or on a different encoder.

An additional level of protection is brought by MFVP Management, which adds N:M redundancy to the encoders. Indeed, as any other encoder, the entire configuration of the groups is backed up by MFVP Management. In case of failure, MFVP Management applies the configuration to a spare encoder, which can take the role of Backup group (or exceptionally Main group) depending on the original role of the failed encoder.

Configuration example

Let's consider a statmux group called **Group_1**. The group gets the overall bandwidth assignment from the operator or the management system, listens to the associated streams requests and allocates the bit rates dynamically. All communication is done via IP protocols.

NOTE You can also create your statmux group using the SOAP API (*See "External SOAP Interface" Technical Note*).

Creating a statmux group

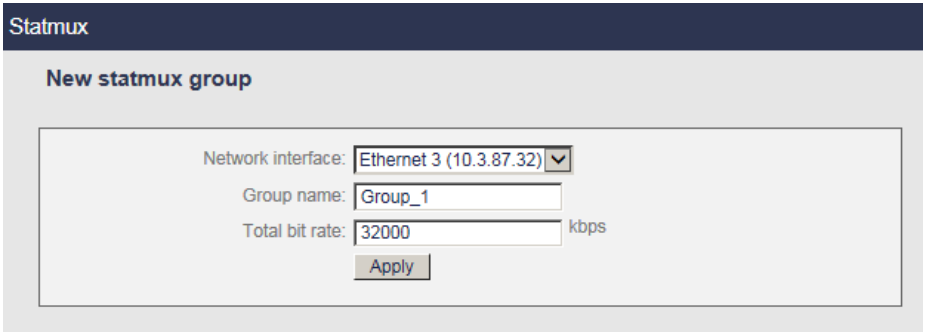
To create a group, follow these steps:

1. Go to the **Statmux** menu in the left panel.
2. In the **New statmux group** section, configure the following parameters:
   - **Network interface**, select the network interface that will be used for communication between the group profiles, **Ethernet 3**.

   NOTE You can use logical NICs to configure your statmux communications.

   - **Group name,** enter a unique group identifier, **Group_1**.
   - **Total bit rate** for this group, enter **32000** kbps.

Figure 79. Creating a new Statmux group



3. Click the **Apply** button to validate the creation of the statmux group.
   The **Group_1** group is now created and added to the list of statmux groups: other profiles can join it.
   You can edit, start or delete the group.

4. When you will have added one or several profiles to the group (see below), you can select **Group_1** and click the **Start** button. The group starts sending announces.

Adding one or several profiles to the group

5. Configure one or more profiles with the following parameters:
   - From the **Profile** submenu, go to the **Bit rate** section then set the **Rate control** to **Statmux**.
     The Overall min, max and constant bit rates are automatically calculated based on the Audio bit rate and the Video min and max bit rates.
   - In the **Statmux** section, select the same **Network interface** as the one selected by the group, **Ethernet 3**.
   - For the **Group name,** select the group to which the profile belongs, **Group_1**.

NOTE You can automatically switch from VBR to CBR (and vice-versa) on SCTE-35 trigger by indicating the reference stream PID; or manually by ticking the button. This can be useful in case of ad insertion splicing to guarantee the quality (CBR).

6. When you start encoding a profile, this profile is added to the group and shares the group bandwidth. Adding or removing a channel from a group can be done dynamically.

NOTES The sum of all the **Min network bit rates** must be lower that the group **Total bit rate**.

Maximum value for **Total bit rate** is 80 Mbps.

For better results, it is advised to leave a large difference between **min & max bit rates** (for example, HD channel should use min=1 000 kbps and max=16 000 kbps).

A group can include up to 20 channels.

CAUTION When two encoders have the same service and profile names, they have the same statmux ID. In a statmux group, two profiles having the same Statmux ID cannot encode together.

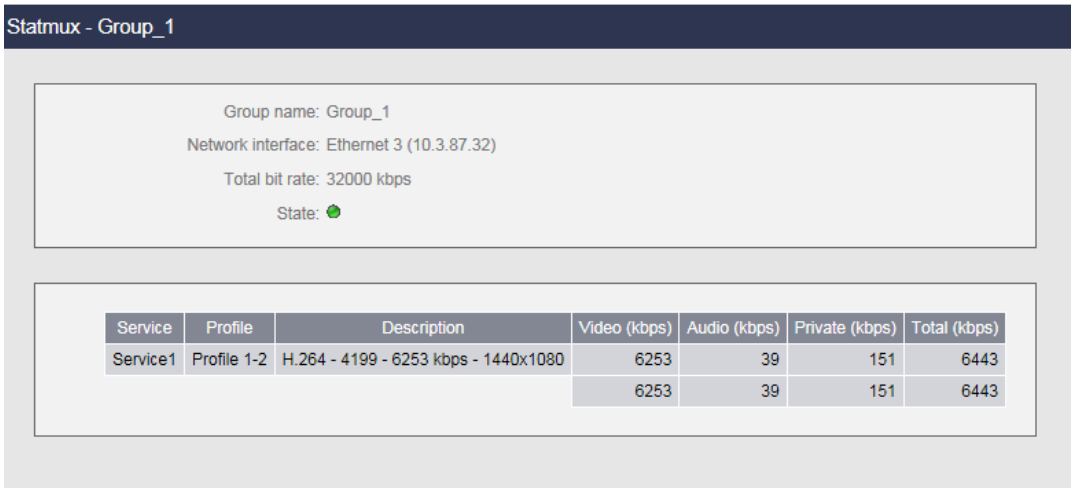
Viewing the statmux group information

Once the group and the profile are started you can check the bandwidth consumption for each profile.

1. From the **Statmux** menu, select a Statmux group. The following parameters are displayed for each group:

- Service name
- Profile name
- Profile description (video codec, min/max bit rate, resolution)
- Video, audio (includes all the audios) and private (includes all the private streams) bit rates (in kbps)
- Total bit rate (in kbps) (does not include the PST tables)

Figure 80. Viewing Statmux group information



Service	Profile	Description	Video (kbps)	Audio (kbps)	Private (kbps)	Total (kbps)
Service1	Profile 1-2	H.264 - 4199 - 6253 kbps - 1440x1080	6253	39	151	6443
			6253	39	151	6443

Chapter 9

DRM and Encryption

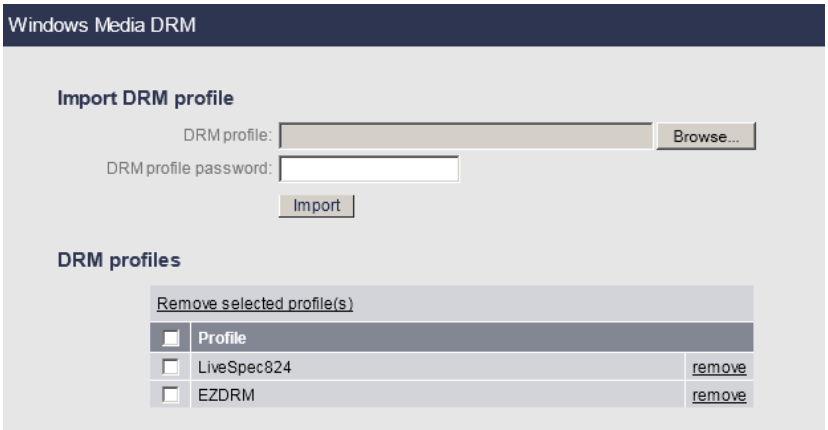
Windows media DRM

You can import DRM profiles that will be used by encoding profiles.

Importing a DRM file

To import a DRM file, follow these steps:

1. From the **System** tab, select the **Windows Media DRM** subtab. The following window is displayed:



2. In the **Import DRM profile** section, click the **Browse** button to select the DRM file then enter the associated password and click the **Import** button.

   The DRM profile is added to the DRM profiles table. It is now available from the output parameters (*see* "Activating a DRM profile" on page 218).

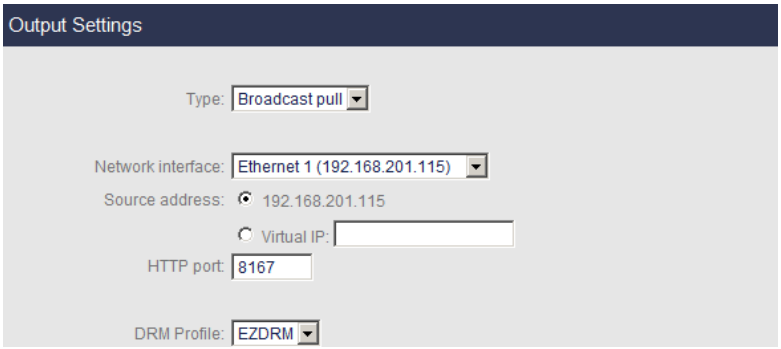
NOTE In the **DRM profiles section**, you can click the **remove** button in front of the DRM Profile to delete it if it is not used or click the **Remove selected profile(s)** link to delete all the checked profiles.

Activating a DRM profile

To activate a DRM profile for a specific encoding profile, follow these steps:

1. Click a Windows Media **Profile #** tab, then select the **Output** subtab in the left menu. The following window is displayed:
2. Select the appropriate value for each parameter:

Figure 81. Output parameters: Broadcast Pull mode



Network interface
Source address
HTTP port

See "Setting the output parameters" on page 128.

DRM profile

By default, DRM profile is set to **None**.

To activate the DRM profile, select one of the DRM profiles that you imported.

DRM Key ID

A default DRM key is automatically generated, but you can change it.

Encryption for Smooth Streaming profile

Encryption type Possible values: **None**, **Fixed key, Fixed key seed, Fixed protected key seed, External key generation**

Specific parameters for fixed key

Key Key used for encrypting the video content.

16 Byte hexadecimal value.

NOTE: When the video is encoded over different resolutions and bit rates (this is the case for Smooth Streaming), the same key is used

Key ID Unique identifier of the encryption key. This information is embedded into PlayReady header, and passed by the Silverlight client player to PlayReady DRM server.

16 Byte base 64 value.

Domain service ID Unique identifier of PlayReady service provider. This ID is granted by Microsoft to companies willing to license PlayReady technology.

License acquisition URL URL of the PlayReady license server where the Silverlight player will need to connect to get the description key.

Licence UI URL By default, set the same value as License acquisition URL.

Specific parameters for fixed key seed

Key Key used for encrypting the video content.

16 Byte hexadecimal value.

NOTE: When the video is encoded over different resolutions and bit rates (this is the case for Smooth Streaming), the same key is used

Key seed Key seed used for encrypting the video content.

30-byte base64-encoded key seed.

Key ID Unique identifier of the encryption key. This information is embedded into PlayReady header, and passed by the Silverlight client player to PlayReady DRM server.

16-Byte base64 value.

Click the **Generate key ID** button to generate a random key ID.

Domain service ID Unique identifier of PlayReady service provider. This ID is granted by Microsoft to companies willing to license PlayReady technology.

16-Byte base64 value

License acquisition URL URL of the PlayReady license server where the Silverlight player will need to connect to get the description key.

Licence UI URL By default, set the same value as License acquisition URL.

Specific parameters for fixed protected key seed

Key Key used for encrypting the video content.

16 Byte hexadecimal value.

NOTE: When the video is encoded over different resolutions and bit rates (this is the case for Smooth Streaming), the same key is used

Protected key seed Key seed used for encrypting the video content.

Click the **Generate protected key seed** button to generate a random key seed for the Smooth Streaming output and protect the key seed by encrypting it.

48-byte base64 hexadecimal value.

Key ID Unique identifier of the encryption key. This information is embedded into PlayReady header, and passed by the Silverlight client player to PlayReady DRM server.

16-byte base 64 value.

Click the **Generate key ID** button to generate a random key ID.

Domain service ID Unique identifier of PlayReady service provider. This ID is granted by Microsoft to companies willing to license PlayReady technology.

16-Byte base64 value

License acquisition URL URL of the PlayReady license server where the Silverlight player will need to connect to get the description key.

Licence UI URL By default, set the same value as License acquisition URL.

Specific External key generation parameters

Key server URL URL of the PlayReady license server where the Silverlight player will need to connect to get the description key.

Content ID Unique identifier of the encryption key.

Encryption for HTTP Live Streaming

Encryption type Possible values: **None**, **Segment based, TS packet based** or **Authentec**

Key source
- If you selected **Segment based** as **Encryption type,** possible values are **Internal key generation** or **External key generation**
- If you selected **TS packet based** as **Encryption type,** possible value is **HTTP ECMG and TS packet encryption**
- If you selected **Authentec** as **Encryption type,** possible values are **External key generation** or **Fixed key**

Specific internal key generation parameters

Key source Select the **Internal key generation** value.

Crypto period Defines how long (in seconds) a crypto word will be used. Default value: **60** s.

The crypto period must be greater than the segment duration.

Content ID By default, this parameter is empty, but you can specify a specific ID.

Separate keys Click this checkbox if you want the key to be published in another location than the streams, then fill in the different parameters. *See "Specific HTTP Live Streaming parameters" on page 134.*

Separate keys **Network interface:** select which network interface is used to publish keys.

Publishing point: specify the location where keys will be published.
Secondary publishing point: for redundancy aspects, a secondary publishing point can be configured.

Distribution point: specify the URL to which iPhone will connect to retrieve decryption key. This URL will be referenced within the playlist.

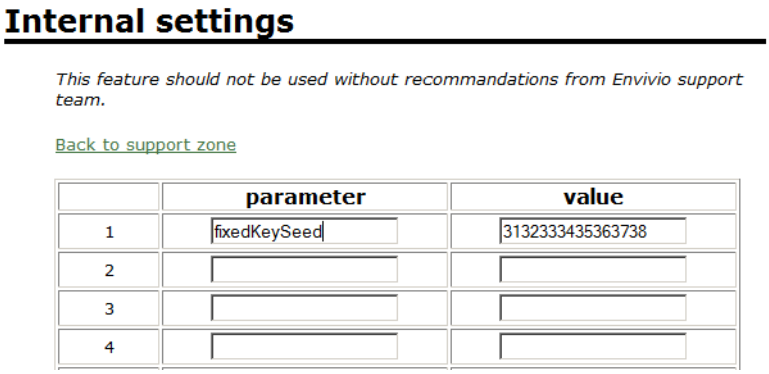
Secondary distribution point: for redundancy aspects, a secondary distribution point can be configured.

Key seed

You can configure the **key seed** parameter (identifier that enables the key generator to identify the content) using advanced configuration settings.

To configure the **key seed**, follow these steps:

1. Open a browser and enter the following address:
   `http://<MFVP-EncodingLive_IP_Address>/Support.html`
2. Click the **Advanced configurations** green link at the top of the page.
3. Enter the following parameter:
   `fixedKeySeed=my_initial_key` where **`my_initial_key`** is an hexadecimal value.



Specific external key generation parameters

Key source	Select the **External key generation** value.
Network interface	Select which network interface is used to publish keys.
Key server URL	Enter the URL of the key server.
Crypto period	Defines how long (in seconds) a crypto word will be used.
Content ID	Unique identifier of the encryption key.

Specific HTTP ECMG and TS packet encryption parameters

Key source Select the **HTTP ECMG and TS packet encryption** value.

Network interface Select which network interface is used to publish keys.

ECMG URL Enter the URL of the key server.

CA PID Enter the MPEG2-TS PID used for the CA (Conditional Access).

Possible values: from 0x0010 to 0x1FFE

NOTE: The CA PID must be different from the video, audio, private, image, PMT and PCR PIDs declared in the HLS TS settings.

CA system ID Enter the identifier of the Conditional Access system.

Possible values: from 0x000 to 0FFFF

Scrambling algorithm Select a scrambling algorithm.

Possible values: AES-128 ECB (L), AES-128 ECB (R), AES-128 CBC (L), AES-128 CBC (R), or ATIS IDSA

Crypto period Defines how long (in seconds) a crypto word will be used.

Encrypt video
Encrypt video headers
Encrypt RAP frames
Encrypt non-RAP frames
Check this option if you want to encrypt the video stream, then specify if you want to encrypt the video headers, the RAP (Random Access Point) and non-RAP frames.

Encrypt audio
Encrypt audio headers
Check this option if you want to encrypt the audio stream, then specify if you want to encrypt the audio headers.

Specific parameters for fixed key seed

Key source Select the **Fixed key** value.

Key Key used for encrypting the video content.

16 Byte hexadecimal value.

NOTE: When the video is encoded over different resolutions and bit rates (this is the case for Smooth Streaming), the same key is used

Key ID Unique identifier of the encryption key. This information is embedded into PlayReady header, and passed by the Silverlight client player to PlayReady DRM server.

16 Byte base 64 value.

Click the **Generate key ID** button to generate a random key ID.

Domain service ID Unique identifier of PlayReady service provider. This ID is granted by Microsoft to companies willing to license PlayReady technology.

License acquisition URL URL of the PlayReady license server where the Silverlight player will need to connect to get the description key.

Licence UI URL By default, set the same value as License acquisition URL.

Chapter 10

Administration

Getting information about MFVP Encoding live

To get information on your MFVP Encoding live, follow these steps:

1. Click the **System** tab, then click the **About** link.
2. Information on MFVP Encoding live is displayed in the right panel, especially:

- Serial number
- Firmware and system versions
- Licensing information
- Current date and time
- Copyright information

Managing the encoder configuration

You can save and restore the MFVP Encoding live encoding settings, that means its services & profiles configurations. The complete system settings are gathered into a single XML file, **config.xml**.

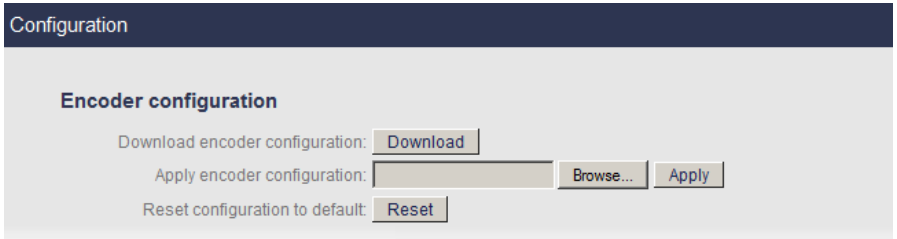
To access the configuration panel, click the **System** tab, then click the **Configuration** subtab. The configuration options are displayed in the right panel.

Saving the current configuration

To get the current configuration and save it on a remote computer, follow these steps:

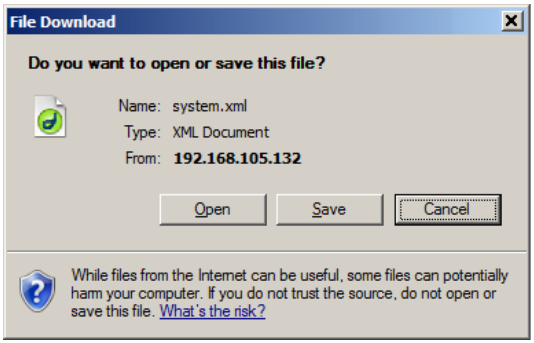
1. Click the **Download encoder configuration** button.

Figure 82. Downloading the configuration



2. A new window is opened.

Figure 83. Saving the configuration



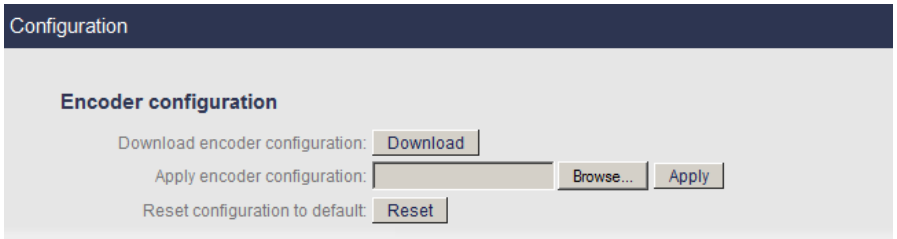
3. Click the **Save** button to save the encoder configuration file then select the destination folder and click **Save**.

Uploading an encoder configuration file

To upload an existing encoder configuration, follow these steps:

1. Next to the Apply encoder configuration parameter, select the appropriate configuration file by clicking the **Browse...** button, then click the **Apply** button.

Figure 84. Uploading a configuration



Resetting the encoder configuration

To reset the encoder to its default configuration, click the **Reset** button next to the Reset configuration to default parameter.

Managing the system configuration

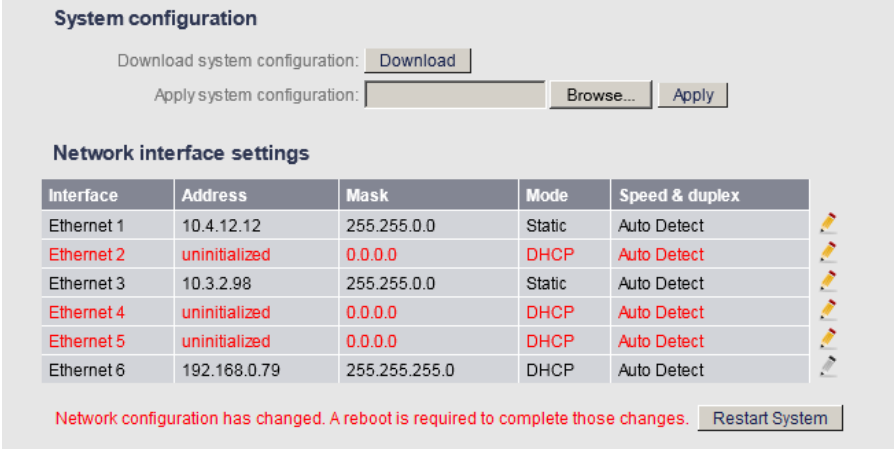
You can save and restore the MFVP Encoding live system settings, that is Ethernet interfaces configuration. The complete system settings are gathered into a single XML file, **system.xml**.

Saving the current configuration

To get the current system configuration and save it on a remote computer, follow these steps:

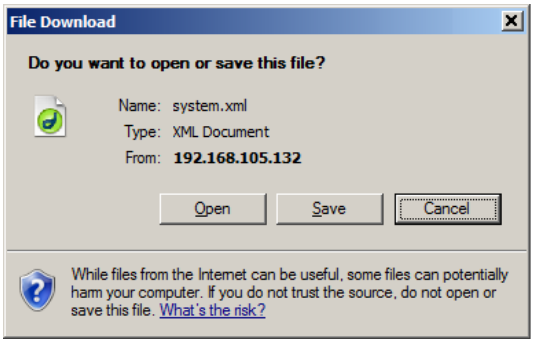
1. Click the **System** tab, then click the **Configuration** subtab. The configuration options are displayed in the right panel.
2. From the **System configuration** part, click the **Download** button next to the Download system configuration parameter.

Figure 85. Downloading the configuration



3. A new window is opened.

Figure 86. Saving the configuration



4. Click the **Save** button to save the system configuration file then select the destination folder and click **Save**.

Uploading a system configuration file

To upload an existing system configuration, follow these steps:

1. Select the **System** tab, then click the **Configuration** subtab. The configuration options are displayed in the right panel.
2. Select the appropriate configuration file by clicking the **Browse...** button, then click the **Apply** button next to the Apply encoder configuration parameter.

Figure 87. Uploading a system configuration

Restricting access to the remote configuration

You can restrict the access to the remote configuration web interface by creating a password.

Restricting the access to the web interface

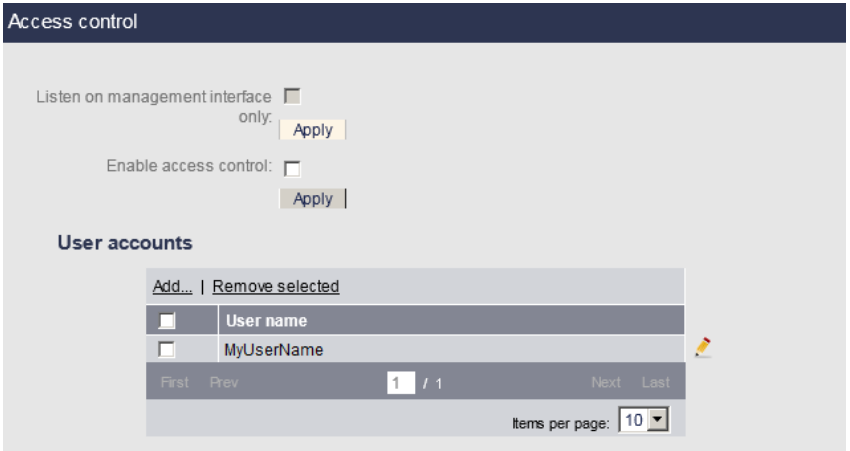
You can select the **Listen on management interface only** checkbox to restrict the web interface access via the recommended network interface, Ethernet 6.

Enabling the access control

To enable the access control, follow these steps:

1. From the web interface, click the **System** tab, then select the **Access control** subtab, the access control parameters are displayed in the right panel.

Figure 88. Access control



2. Select the **Enable access control** checkbox then click the **Apply** button.

Disabling the access control

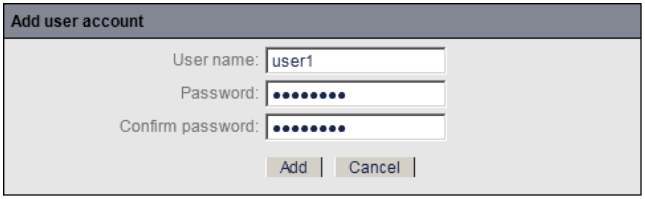
To disable the access control, deselect the **Enable access control** checkbox.

Creating a user

To add a user, follow these steps:

1. From the **Users** section, click the **Add** link, a new window is displayed.

Figure 89. Creating a new user



2. Enter the user name, the password twice, and click the **Add** button.
3. The new user is added to the list of users.

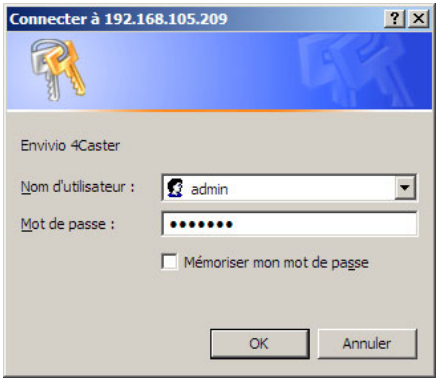
Changing the password

To change the password, follow these steps:

1. Click the **edit** button, in front of the user name, a new window is displayed.
2. Enter the current password, then enter the new password twice, and click the **Apply** button.

NOTE The user name (***admin***) cannot be modified.

When you try to access MFVP Encoding live web interface, you will be asked to enter the user name and password.



Removing a user

To remove a user, follow these steps:

1. Check the user you want to delete then click the **Remove selected** link.
2. A message asks you to confirm the deletion of the user. Click **OK**.
3. The user is removed from the list of user accounts.

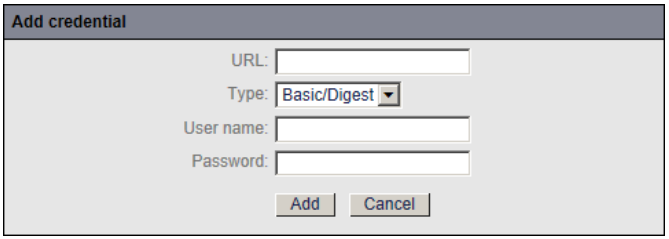
Managing the certificates and credentials

Adding a credential

You can create credentials to access a server for input or output files.
To create a credential, follow these steps:

1. From the web interface, click the **System** tab, then select the **Credentials** subtab.
2. In the **Manage credentials** section click the **Add** link. A new window is opened.

Figure 90. Adding a credential



3. Select the appropriate value for each parameter.
4. The new credential will be added to the list of credentials.

URL Enter the URL to the server.

Type Possible values: **Token**[1], **Basic/Digest**, or **Certificate**
Depending on the credential type, additional parameters are displayed.

1. Especially for HTTP streaming publishing with CDNs.

Specific Token authentication parameters

Token seed Enter the Token seed.

Refresh period Specify the Token refresh period.

Specific Basic/Digest authentication parameters

Corresponds to the basic/digest authentication.

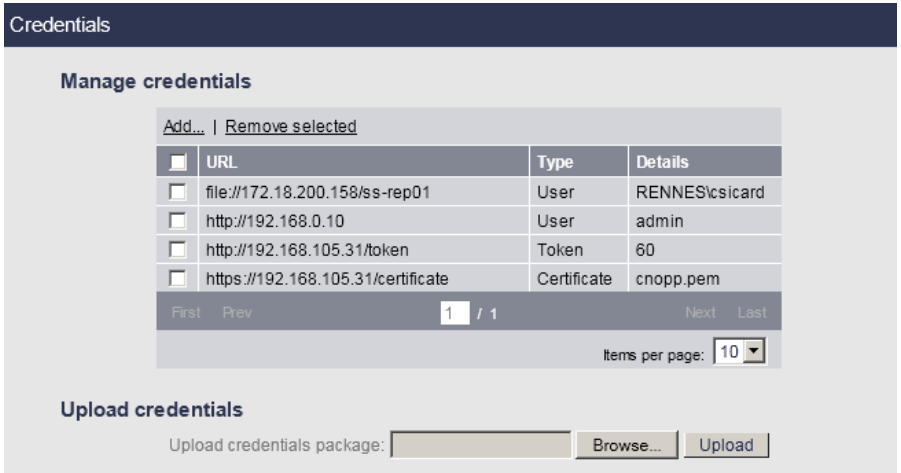
User name Specify the user name (server_ip_address\user_name or domain_name\user_name to access the server.

Password Specify the password to access the server.

Specific Certificate authentication parameters

1. To download a certificate, follow these steps:
2. Click the **Browse** button then select the appropriate certificate file and click the **Add** button.
3. The new certificate is added to the list of credentials.

Figure 91. Credentials



Uploading a credential file

To upload a credential file, go to the **Upload credentials** section, click the **Browse** button next to the **Upload credentials package** field then select the appropriate file and click the **Upload** button. The credentials are added to the list of credentials.

Zip archive must contain:

- An xml credential configuration file (mandatory),
- Required certificates files (optional, needed if the credential configuration contains a certificate-based credentials).

Example of a credential file:

```
<credentials>
<credential type="token">
<url>http://192.168.105.31/token</url>
<seed>Yemv71DtcbUVkIg/UxxigQ==</seed>
<refreshPeriod>60</refreshPeriod>
</credential>
<credential type="user">
<url>rtsp://192.168.105.31/Toto2</url>
<username>myusername</username>
<password>mypassword</password>
</credential>
<credential type="certificate">
<url>https://192.168.105.31/certificate</url>
<path>cnopp.pem</path>
</credential>
</credentials>
```

In our example, you will create a ZIP file including both the above XML file and the **cnopp.pem** file.

Using a credential

Once you have defined credentials, when entering the server URL, the user icon, should appear in color.

Deleting a credential

You can delete a credential by selecting its checkbox and clicking the **Remove selected credential(s)** link.

Playready encrypted key seeds

To define encrypted key seeds, follow these steps:

1. From the web interface, click the **System** tab, then select the **Playready** subtab.
2. Fill in the required parameters.

Certificate to encrypt key seeds

You can upload a certificate containing a public key to use for encrypting key weeds when they are published to external system.

To Upload the certificate, click the **Browse** button then select the appropriate file and click the **Upload** button.

The format of the certificate is a PEM file.

Key seeds publishing URL

You can publish all the key seeds present in the encoder configuration to an external FTPS server. This will extract all the key seeds and key IDs for all PlayReady-encrypted Smooth Streaming outputs and generate a file and upload it to an FTPS server.

Enter the URL of the external FTPS server.
URL format: `ftps://host/path`

Click the **Publish** button to upload the key seed mapping file to the FTPS server.

NOTE: If you defined credentials to access this URL (*see* "Adding a credential" on page 235), the user icon, should appear in color.

Setting the date and time

To set the date and time, follow these steps:

1. Click the **System** tab, then click the **Configuration** subtab. The configuration options are displayed in the right panel.
2. The date and time parameters are displayed in the right panel.

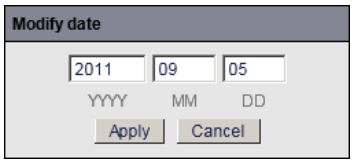
Figure 92. Date and time settings

Setting the date

To edit the date, follow these steps:

1. Click the pencil button, next to the date. A new panel is displayed.

Figure 93. Setting the date



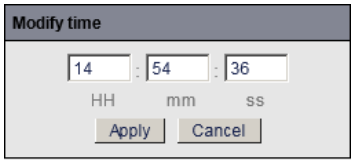
2. Enter the new date then click the **Apply** button.

Setting the time

To edit the time, follow these steps:

1. Click the pencil button, next to the time. A new panel is displayed.

Figure 94. Setting the time



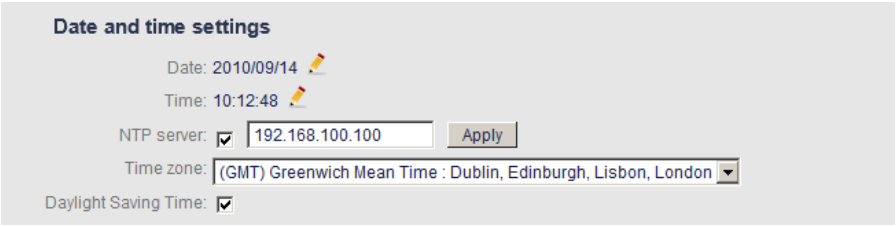
2. Enter the new date then click the **Apply** button.

Enabling the NTP server

If NTP[1] is enabled, MFVP Encoding live time will be given by a specific server.

If you check the NTP server option, you will have to specify the NTP server IP address is.

Figure 95. Enabling the NTP server



NOTE When using MFVP Encoding live on Linux, please refer to the Installation Guide, "MFVP Encoding live - Basic system commands".

1. See "NTP" in Glossary on page 284.

Setting the time zone

MFVP Encoding live uses Windows API for the time zone.
To set the time zone select a value from the drop-down list.

NOTE When using MFVP Encoding live on Linux, please refer to the Installation Guide, "MFVP Encoding live - Basic system commands".

Enabling the daylight saving time

Check this parameter if you observe the daylight savings time (DST).

Defining T0 for smooth streaming

Check this parameter if you want to define a reference date (T0) for smooth streaming using the standard time information used in the encoder.

To specify T0:

1. Click in the time area.
2. Set the date, time and time zone, then click the **Done** button to validate,
3. or click the **Now** button to get the current time.

Enabling SNMP

The SNMP agent allows users to receive traps when alarms occur.

To enable SNMP, follow these steps

1. Click the **System** tab, then click the **Configuration** subtab. The configuration options are displayed in the right panel.
2. In the **SNMP settings** section, check the **Enable SNMP** option.

NOTE When using MFVP Encoding live on Linux, please refer to the Installation Guide, "MFVP Encoding live - Basic system commands".

Shutting down or restarting the system

You can restart or shut down MFVP Encoding live from the web interface:

- From the web interface, click the **System** tab, then click the **Shutdown/restart** subtab.
- Click the **Shutdown system** or the **Restart system** button.

Figure 96. Shutting down or restarting MFVP Encoding live

NOTE A message will ask you to confirm your action.

Appendix A

Encoding parameters Guidance

Encoding recommendations

NOTES

These recommendations are based on requirements from available standards and configurations of our customers. They have to be considered as guidance to define a service configuration.

These recommendations are generic; it means that adjustments are required to meet each network and each device capabilities and specificities. These recommendations may also differ from some available standards.

Table 1. SD / HD source, 16:9 content

Network	Typical Device	Typical Protocol	Bit rate	Encoding Resolution	Codec	Frames (fps)	Audio codec	ES Audio (kbps)	Mode	Sample rate (kHz)
IPTV	TV	Mpeg-2 TS	6 Mbps	1920x1080 (HD)	H.264 High	Full	AC3 (pass through)/ MPEG1 Layer 2	196 / 128	Stereo	48
			4Mbps	1280x720 (HD)						
			1.6 Mbps	720x576 720x480	H.264 Main					
Broadband / WIFI	Connected TV OTT STB	HLS / SS / Flash / MPEG-2 TS	4Mbps	1280x720 (HD)	H264 Main	Full (30fps or 25 fps)	HE-AAC	48	Stereo	44,1 / 48
	Tablet / PC / OTT STB / Connected TV		2,6Mbps	1280x720 (HD)						
	Tablet / PC / OTT STB / Connected TV		1,8 Mbps	960x540 (HD)						
	SmartPhone / Tablet		1.4Mbps	640x360						
			1Mbps	640x360						
			650kbps	640x360						
3,5G / 4G	SmartPhone / tablet	3GPP / HLS / SS / Flash	440kbps	480x270	H264 Baseline	Full (30fps or 25 fps)	HE-AAC	24	Mono	32
3,5G / 3G	SmartPhone / tablet	3GPP / HLS / SS / Flash	300kbps	320x160			HE-AAC	24	Mono	32
3G	SmartPhone / Other phones	3GPP / HLS / SS / Flash	150kbps	320x160		Half	AAC	24	Mono	32
2,5 G	Other phones	3GPP	70kbps	176x144	MPEG-4 SP	Half / Quarter	AMR	12,2	Mono	8kHz
2G	Other phones	3GPP	50kbps	176x144	MPEG-4 SP	Half / Quarter	AMR	12,2		

Table 2. SD source, 4:3 content

Network	Typical Device	Typical Protocol	Bit rate	Encoding Resolution	Codec	Frames (fps)	Audio codec	ES Audio (kbps)	Mode	Sample rate (kHz)
IPTV	TV	Mpeg-2 TS	1.6 Mbps	720x576 720x480	H.264 Main	Full	AC3 (pass through) / MPEG1 Layer 2	196 / 128	Stereo	48
Broadband / WIFI	SmartPhone / Tablet/ PC / OTT STB / Connected TV	3GPP / HLS / SS / Flash / MPEG-2 TS	1.4Mbps	640x480						
			1Mbps	640x480						
			650kbps	640x480						
3,5G / 4G	SmartPhone / tablet	3GPP / HLS / SS / Flash	440kbps	480x360	H264 Baseline	Full (30fps or 25 fps)	HE-AAC	24	Mono	32
3,5G / 3G	SmartPhone / tablet	3GPP / HLS / SS / Flash	300kbps	320x240			HE-AAC	24	Mono	32
3G	SmartPhone / Other phones	3GPP / HLS / SS / Flash	150kbps	320x240		Half	AAC	24	Mono	32
2,5 G	Other phones	3GPP	70kbps	176x144	MPEG-4 SP	Half / Quarter	AMR	12,2	Mono	8kHz
2G	Other phones	3GPP	50kbps	176x144	MPEG-4 SP	Half / Quarter	AMR	12,2		

Appendix B

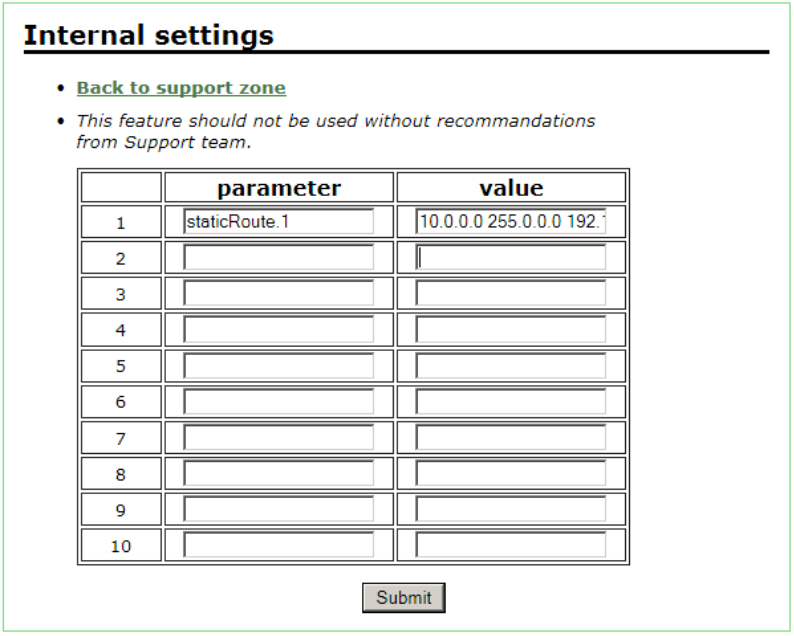
Static Routes Declaration

Adding a static route

To add static routes, you need to configure specific parameters using advanced configuration settings.

To add a route, follow these steps:

1. Open a browser and enter the following address:
   `http://<MFVP-EncodingLive_IP_Address>/Support.html`
2. Click the **Advanced configurations** green link at the bottom of the page.
   The following page is displayed:



3. Enter the following parameter:
   `staticRoute.n`
   … where **`n`** depends on the number of static routes you want need.
4. Then enter the following value:
   `target_IP_address mask gateway_IP_address NICs`
   where NICs can be: **`1, 2, 3, 4, 5, 6, 1-2, 3-4, 5-6`**

   Example: **`staticRoute.1=10.0.0.0 255.0.0.0 192.168.232.1 1-2`**

Static route checking

Use the following commands to check the persistent routes:

http://<Encoder_IP_Address>/support/runcmd.py

route print

Output

Command: route print send

```
===========================================================================
Interface List
 21...00 1e 67 3e 07 25 ......Intel(R) I350 Gigabit Network Connection #6
 19...00 1e 67 3e 07 24 ......Intel(R) I350 Gigabit Network Connection #5
 17...00 1e 67 54 76 9b ......Intel(R) I350 Gigabit Network Connection #4
 15...00 1e 67 54 76 9a ......Intel(R) I350 Gigabit Network Connection #3
 13...00 1e 67 54 76 99 ......Intel(R) I350 Gigabit Network Connection #2
 11...00 1e 67 54 76 98 ......Intel(R) I350 Gigabit Network Connection
  1...........................Software Loopback Interface 1
 12...00 00 00 00 00 00 00 e0 Microsoft ISATAP Adapter
 16...00 00 00 00 00 00 00 e0 Microsoft ISATAP Adapter #3
 18...00 00 00 00 00 00 00 e0 Microsoft ISATAP Adapter #4
 20...00 00 00 00 00 00 00 e0 Teredo Tunneling Pseudo-Interface
 22...00 00 00 00 00 00 00 e0 Microsoft ISATAP Adapter #5
 23...00 00 00 00 00 00 00 e0 Microsoft ISATAP Adapter #6
===========================================================================

IPv4 Route Table
===========================================================================
Active Routes:
Network Destination        Netmask          Gateway       Interface  Metric
          0.0.0.0          0.0.0.0         10.2.0.1       10.2.12.16     40
         10.0.0.0        255.0.0.0    192.168.232.1       10.4.12.16     15
         10.2.0.0      255.255.0.0         On-link        10.2.12.16    276
       10.2.12.16  255.255.255.255         On-link        10.2.12.16    276
     10.2.255.255  255.255.255.255         On-link        10.2.12.16    276
         10.3.0.0      255.255.0.0         On-link        10.3.12.16    266
       10.3.12.16  255.255.255.255         On-link        10.3.12.16    266
     10.3.255.255  255.255.255.255         On-link        10.3.12.16    266
         10.4.0.0      255.255.0.0         On-link        10.4.12.16    266
       10.4.12.16  255.255.255.255         On-link        10.4.12.16    266
     10.4.255.255  255.255.255.255         On-link        10.4.12.16    266
        127.0.0.0        255.0.0.0         On-link         127.0.0.1    306
        127.0.0.1  255.255.255.255         On-link         127.0.0.1    306
  127.255.255.255  255.255.255.255         On-link         127.0.0.1    306
        224.0.0.0        240.0.0.0         On-link         127.0.0.1    306
        224.0.0.0        240.0.0.0         On-link        10.3.12.16    266
        224.0.0.0        240.0.0.0         On-link        10.2.12.16    276
        224.0.0.0        240.0.0.0         On-link        10.4.12.16    266
  255.255.255.255  255.255.255.255         On-link         127.0.0.1    306
  255.255.255.255  255.255.255.255         On-link        10.3.12.16    266
  255.255.255.255  255.255.255.255         On-link        10.2.12.16    276
  255.255.255.255  255.255.255.255         On-link        10.4.12.16    266
===========================================================================
Persistent Routes:
  Network Address          Netmask  Gateway Address  Metric
          0.0.0.0          0.0.0.0         10.2.0.1      20
===========================================================================
```

Appendix C

Prerequisites and Standards Compliance

Prerequisites

MFVP Encoding live supports IP ingest of MPEG-2-TS streams with the following characteristics

TS Stream

IP ingest

- Total input bandwidth of 360 Mbps
  - 30 Mbps max. per SPTS
  - 90 Mbps max. per MPTS

ASI ingest

- Total input bandwidth of 160 Mbps
  - 60 Mbps max. per MPTS / SPTS
- ASI characteristics:
  - Data Bit Rate: 270 Mbps
  - Supported modes: Byte & Burst
  - Packet Size can be 188 or 204 bytes.

Video

- Single video track
- Codec:
  - MPEG-2 HP@HL (up to 30 Mbps)
  - H.264 HiP@Level4 (up to 30 Mbps)
- Resolution and frame rate:
  - 576i and 480i @25 / 29.97 fps
  - 1080i/p@ 25 / 29.97 / 30 fps
  - 720p @ 50 / 59.94 / 60 fps

Audio

- Up to 8 audio tracks
- Codec:
  - AAC (HE-AAC, HE AAC v2)
  - MPEG-1 Layer 2
  - AC-3, Enhanced AC-3 (pass-through)

MFVP Encoding live Input/Output Standards compliance

Type		Norm Reference	Parameters to measure
TS	SPTS	ETR 101290	Level 1,2 and part of 3 of norm
	MPTS	ETR 101290	Level 1,2 and part of 3 of norm
IP	Transport	IETF RFC 4445 IETF RFC 768 (UDP) IETF RFC 3550 (RTP) IETF RFC 2326 (RTSP) IETF RFC 2429 (RTP Payload Format for H263+)	• Protocol compliance • Off-air period • Number of PIDs per service • Scramble status • Program loss: MLR, MLT-15, MLS-15, MLT24, MLS24 • Program Bit Rate alarms: count of alarms and duration • Program outage alarms • Flow outage alarms • Bit rate statistics for the last 15 minutes (current/min/max/average) • Jitter and Delay factor (MDI –DF)
	MPEG over IP	ISO 14496 part 8 IETF RFC 3267 (AMR-NB & AMR-WB) IETF RFC 2658 (QCELP) IETF RFC 3984 (H264 Video) IETF RFC 3640 (Transport of MPEG-4 ES) IETF RFC 3016 (MPEG-4 AV Streams payload format) IETF RFC 3551 (RTP Profile for Audio and Video Conferences with Minimal control) IETF RFC 4566 (SDP)	• Protocol compliance
ASI		ETSI TR 101 891 BS EN50083-9	• Peak-to-peak signal amplitude • Maximum Rise and fall times • Deterministic Jitter (DJ) and Random Jitter (RJ)

Type		Norm Reference	Parameters to measure
Serial video	SDI	SMPTE 259M ITU-R BT.601/656 SMPTE 272M	• Peak-to-peak signal • Rise and fall times • Signal waveform jitter • No illegal code • Carry supported video formats • Correct values in group format • No EDH group error • No CRC error
	HD-SDI	SMPTE 292M SMPTE 299M SMPTE 291M SMPTE RP 184	• Peak-to-peak signal • Rise and fall times • Signal waveform jitter • No illegal code • Carry supported video formats
	PAL	ITU BT R 1700/B	• Signal amplitude • Time measurement (sync mostly) • Noise level
MPEG	Codec	ISO/IEC 11172-3 ISO 13818-2 ISO 13818-3 ISO 14496 part 3/10/14 ITU-T Rec. H.263 Video	• Format/norm compliance

Appendix D

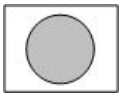
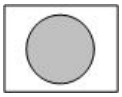
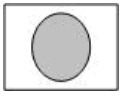
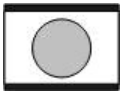
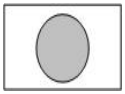
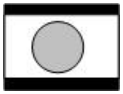
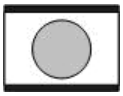
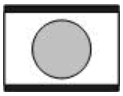
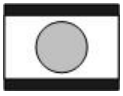
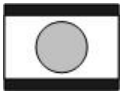
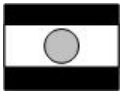
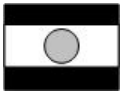
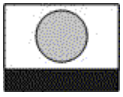
Output aspect ratio and cropping management

Picture aspect ratio handling

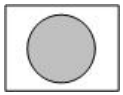
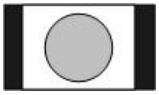
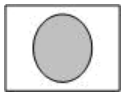
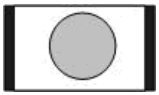
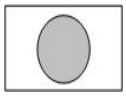
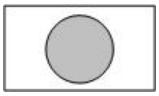
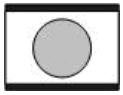
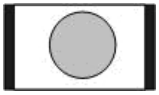
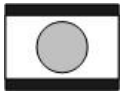
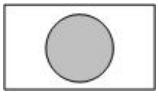
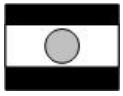
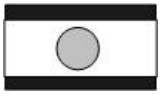
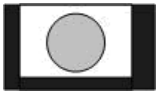
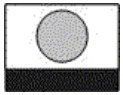
"Auto" aspect ratio handling

If the input video signal embeds WSS information and if the **Aspect ratio** parameter is set to **Auto**, MFVP Encoding live can extract the picture aspect ratio (PAR) information and transforms the input video as follows.

NOTE The signal always comes as 4:3, but the picture could be 16:9 anamorphic.

Video Input	WSS Flag	Processing	4:3 Video Output
	4:3 Full	Nothing	
	14:9 Full [1]	Restore PAR and fit in letterbox	
	16:9 Full	Restore PAR and fit in letterbox	
	14:9 Letterbox (center) [1]	Nothing	
	16:9 Letterbox (center) [1]	Nothing	
	> 16:9 Letterbox (center) [1]	Nothing	
	14:9 Letterbox (top) [1]	Nothing	
	16:9 Letterbox (top) [1]	Nothing	

1. Only available in PAL.

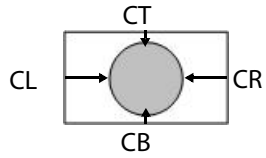
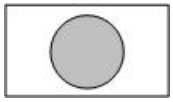
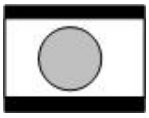
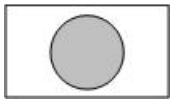
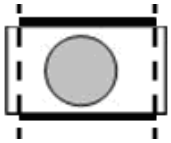
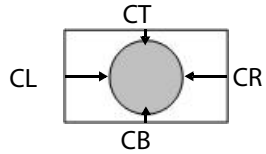
Video Input	WSS Flag	Processing	16:9 Video Output
	4:3 Full	Fit in pillar box	
	14:9 Full [1]	Restore PAR and fit in pillar box	
	16:9 Full	Restore PAR	
	14:9 Letterbox (center) [1]	Remove black stripes and fit in pillar box	
	16:9 Letterbox (center) [1]	Remove black stripes	
	> 16:9 Letterbox (center)	Remove black stripes and fit in letterbox	
	14:9 Letterbox (top) [1]	Fit in pillar box	
	16:9 Letterbox (top) [1]	Fit in pillar box	

1. Only available in PAL.

NOTE Only the aspect ratio part of WSS is supported. Additional information such as enhanced services and subtitles is not used.

Cropping management

Cropping is managed in a different way depending on the output aspect ratio management.

Cropping method	Parameters	Input display	Output display
Dynamic	Crop Left, Right, Top, Bottom and keep Aspect ratio		
Letter Boxing	Adaptation of input 16:9 to output 4:3 zoom level 0%		
	Adaptation of input 16:9 to output 4:3 zoom level 50%		
Stretch to fit	Crop Left, Right, Top, Bottom and Stretch		

Appendix E

H.264 Codecs

Premium, Extreme and Elite video codecs

Premium H.264

- H.264 baseline profile
- Video quality inherits from the previous generation of encoders for backward compatibility purposes

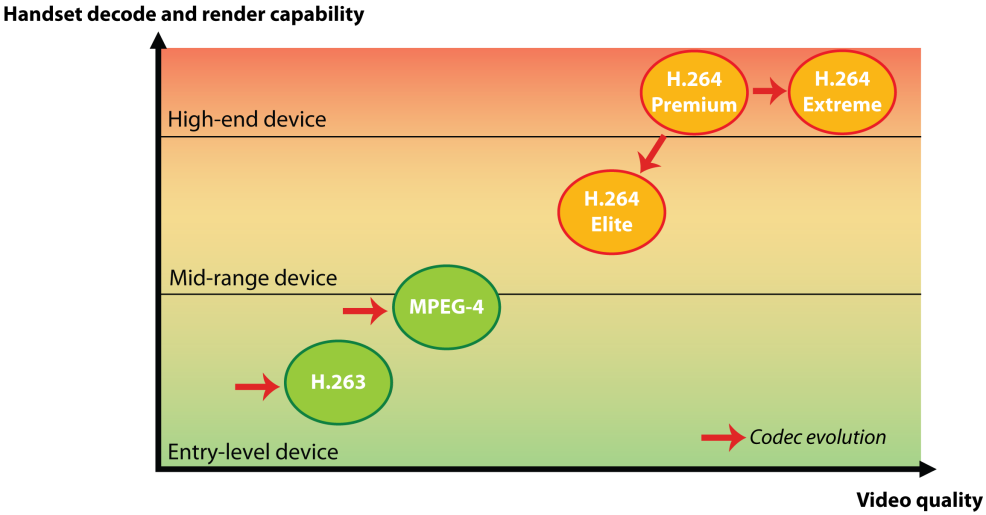
Extreme H.264

- H.264 baseline profile
- Best video quality at mobile bit rates:
  - Spatio-temporal adaptive rate control: provides better trade-off between picture and movement quality on difficult scenes
  - Rate Distortion Optimization (RDO): new advanced heuristics to determine the best coding strategy
  - Motion estimation improvements: additional gain brought by 1/2 PEL and increased number of reference frames
  - Adaptive area-based compression: improvement on edge and text readability

Elite H.264

- H.264 baseline profile and optimized algorithms for mid-range handsets
- Based on H.264 Extreme
- Further optimizes H.264 quality at targeted rates: 200 kbps
- Works on H.264 deployed players
- Cooperation with player manufacturers on simplification study and IOT testing
- Improves the decoding frame rate by 5 to 7 frames compared to Extreme H.264.

Figure 1. Handset versus video quality



Appendix F

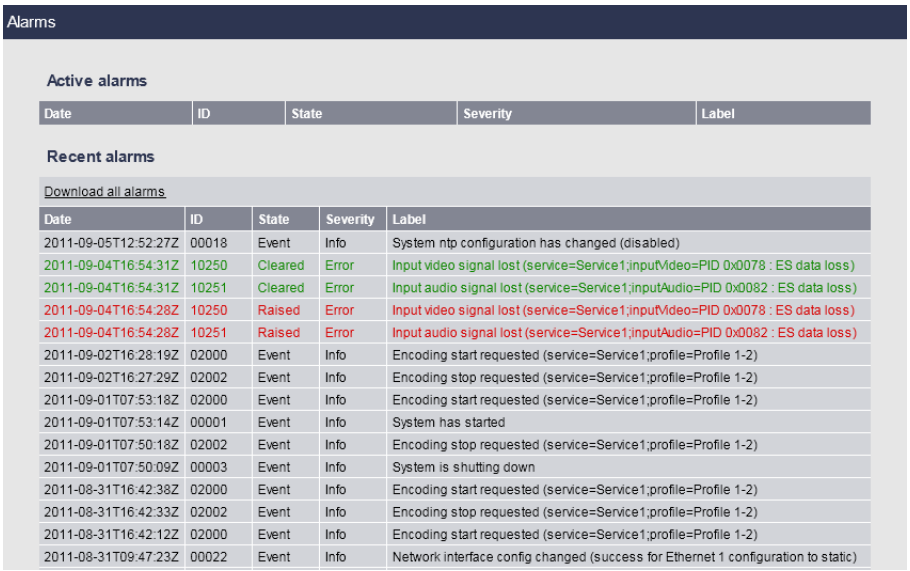
Alarms and Logs

Reviewing MFVP Encoding live alarms

To view MFVP Encoding live alarms, follow these steps:

1. Click the **System** tab, then click the **Alarms** subtab.
2. The list of alarms is displayed in the right panel.
   This list is periodically updated (every 3 seconds).

Figure 2. Alarms panel



Alarms

Active alarms

Date	ID	State	Severity	Label

Recent alarms

Download all alarms

Date	ID	State	Severity	Label
2011-09-05T12:52:27Z	00018	Event	Info	System ntp configuration has changed (disabled)
2011-09-04T16:54:31Z	10250	Cleared	Error	Input video signal lost (service=Service1;inputVideo=PID 0x0078 : ES data loss)
2011-09-04T16:54:31Z	10251	Cleared	Error	Input audio signal lost (service=Service1;inputAudio=PID 0x0082 : ES data loss)
2011-09-04T16:54:28Z	10250	Raised	Error	Input video signal lost (service=Service1;inputVideo=PID 0x0078 : ES data loss)
2011-09-04T16:54:28Z	10251	Raised	Error	Input audio signal lost (service=Service1;inputAudio=PID 0x0082 : ES data loss)
2011-09-02T16:28:19Z	02000	Event	Info	Encoding start requested (service=Service1;profile=Profile 1-2)
2011-09-02T16:27:29Z	02002	Event	Info	Encoding stop requested (service=Service1;profile=Profile 1-2)
2011-09-01T07:53:18Z	02000	Event	Info	Encoding start requested (service=Service1;profile=Profile 1-2)
2011-09-01T07:53:14Z	00001	Event	Info	System has started
2011-09-01T07:50:18Z	02002	Event	Info	Encoding stop requested (service=Service1;profile=Profile 1-2)
2011-09-01T07:50:09Z	00003	Event	Info	System is shutting down
2011-08-31T16:42:38Z	02000	Event	Info	Encoding start requested (service=Service1;profile=Profile 1-2)
2011-08-31T16:42:33Z	02002	Event	Info	Encoding stop requested (service=Service1;profile=Profile 1-2)
2011-08-31T16:42:12Z	02000	Event	Info	Encoding start requested (service=Service1;profile=Profile 1-2)
2011-08-31T09:47:23Z	00022	Event	Info	Network interface config changed (success for Ethernet 1 configuration to static)

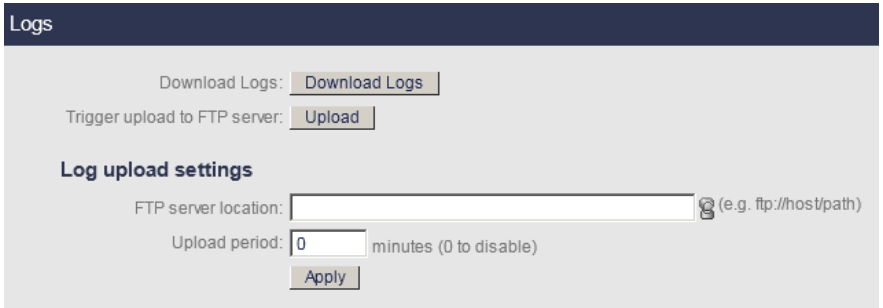
NOTE You can click the **Download all alarms** link to save the alarms file.

Reviewing MFVP Encoding live logs

To view MFVP Encoding live logs, follow these steps:

1. Click the **System** tab, then click the **Logs** subtab.
2. You can either download the logs on your PC or upload the logs to a remote server

Figure 3. Logs panel



Downloading the logs

To download the logs, click the **Download Logs** button.

Uploading the logs

To upload the logs, follow these steps:

1. Configure the remote server by entering the **Location** then enter the **Upload period** from 0 (disabled) to 1440 minutes).
2. Click the **Apply** button.
3. You can click the **Upload** button at any moment to upload the logs to the specified remote server.

Appendix G

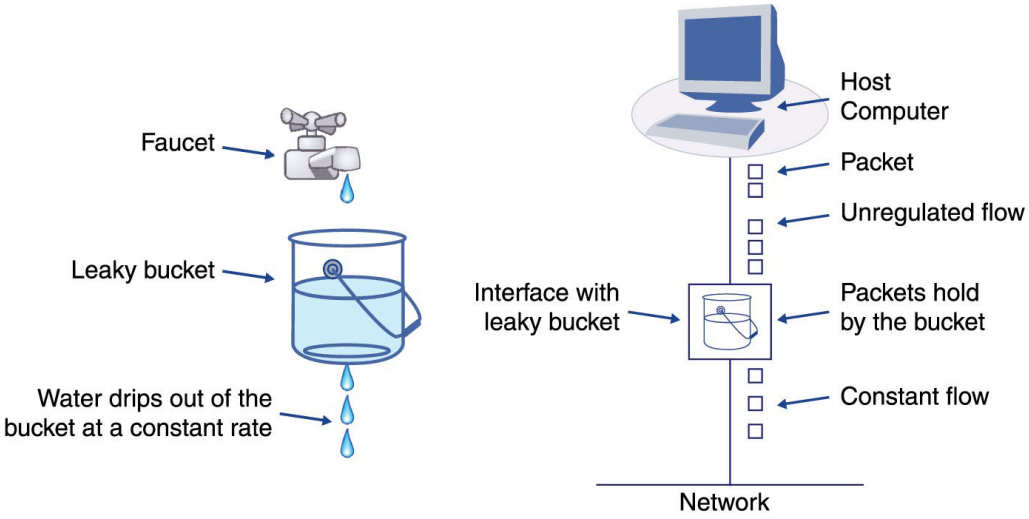
Leaky bucket concept

How it works

Leaky bucket is a flow-control mechanism designed to reduce the effect of the inevitable variability in input data streams as they are "injected" into communication networks. The leaky bucket algorithm performs the following functions:

- It controls what the encoder can inject into the network.
- It prevents the data "bucket" from overflowing by automatically regulating the bit rates in the event that the encoder generates too much data or very complex data.
- It prevents “burstiness” in the input stream, ensuring a smooth, even data flow.

Figure 4. Leaky bucket algorithm



Appendix H

Monitoring via SNMP

Using SNMP monitoring

MFVP Encoding live supports SNMP for monitoring purposes.

You can use an SNMP supervisor to remotely monitor several MFVP Encoding live encoders.

You can retrieve the MFVP Encoding live MIBs (Management Information Base) at:

http://<MFVP-EncodingLive-IP>/Snmp/Mibs/

SNMP community names

- The default Read SNMP community name for the SNMP agent is **public_envivio**.
- The default Write SNMP community name for the SNMP agent is **private_envivio**.

You can modify the SNMP community names using a SET SNMP command.

CAUTION ReadWriteCommunity and ReadCommunity must not have the same name.

Enabling SNMP trap reception

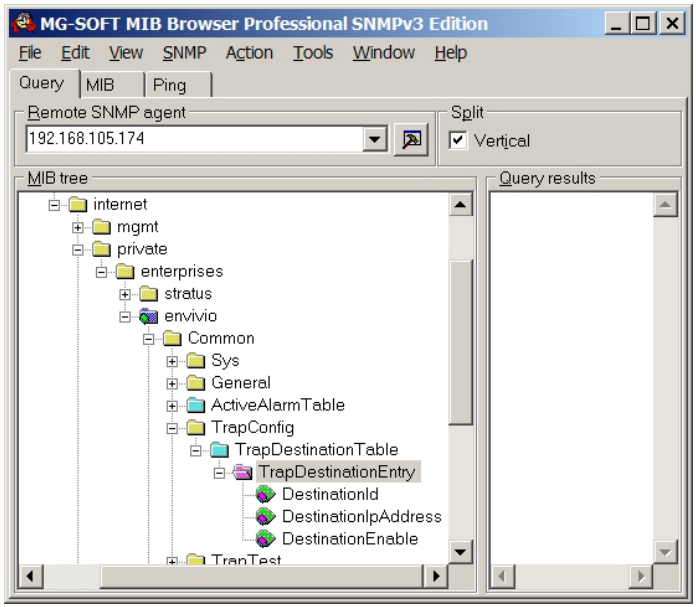
SNMP traps are used to monitor the events and alarms generated by MFVP Encoding live. To receive SNMP traps, you have to be registered as a trap listener on MFVP Encoding live SNMP agent.

NOTE Each encoder supports up to ten trap listeners.

Registering as a trap listener

To be registered as a trap listener, you must fill the destinations fields in the trap destination entry table.

Figure 5. Example using a MIB browser, MG-SOFT [1]



DestinationIPAddress is the target client IP address.

DestinationEnable must be set to *true* to receive the SNMP trap.

1. http://www.mg-soft.com/mgMibBrowserPE.html

Appendix I

Troubleshooting

Troubleshooting advice

- Make sure that the system is plugged into an electric power outlet and switched on.
- Ensure that the video, audio and network cables are correctly connected.

I get an error when I start encoding

Problem description

When I start encoding a profile from the web interface, I get an error message.

Tests to perform

1. Open MFVP Encoding live web interface, then check the error messages in the Alarms section *(see "Alarms and Logs" on page 263)* to identify the alarm.
2. Check your settings, especially the transport settings, check that you are streaming on a valid network interface.

NOTE If you cannot connect to the web interface, please refer to "I cannot access the web interface of my encoder" on page 274.

Solution

This error is usually a configuration issue. Checking the parameters should resolve the problem.

I cannot access the web interface of my encoder

Problem description

I cannot open MFVP Encoding live web interface http://<MFVP-EncodingLive_IP>.

Tests to perform

1. Check the IP connectivity, by *pinging* MFVP Encoding live from another device on the same IP subnet (from this device command prompt):

   ping <MFVP-EncodingLive_IP> (where **MFVP-EncodingLive_IP** is the IP address of MFVP Encoding live)

   If MFVP Encoding live does not reply, refer to the solution explained on page 275, "I cannot ping my encoder".

2. Try to access one of the following pages:

   - The updater web page: **http://<MFVP-EncodingLiveIP>/Updater/**
   - The support page: **http://<MFVP-EncodingLive_IP>/Support.html**

   If you can access one of these pages but you cannot access the main page, see solution below.

Solution

1. Restart your encoder to reset the startup configuration (*see* "How to restart MFVP Encoding live" on page 278).
2. If the problem still exists after restarting, get the support package and send it to the support team (*see* "How to report a problem to the support team" on page 279) who will provide you a way to overcome this issue.

I cannot ping my encoder

Problem description

When I ping MFVP Encoding live:

ping <MFVP-EncodingLive_IP> (where MFVP-EncodingLive_IP is the IP address of MFVP Encoding live),

… I receive no reply from the equipment.

Tests to perform /solution

Usually this problem is related to a network failure, but it can also be due to a bad network configuration.

1. Check at the back of the encoder that the network interface is connected to the network.
2. Check that you can ping another system located on the same network as MFVP Encoding live.

- If you cannot ping any equipment on the network – your IP switch or router may malfunction, or your console is not connected to the network – you should contact your network administrator.
- If you cannot ping MFVP Encoding live but you can ping other equipment, check the network connection between MFVP Encoding live and your network switch.
  On the switch side, check the LED corresponding to the port to which MFVP Encoding live is connected (refer to the switch's manufacturer documentation for the meaning of the LED status).
  - If the LED status means that MFVP Encoding live is not connected, check that the cable is properly inserted into both the switch and MFVP Encoding live Ethernet ports.
  - If the LED status means that MFVP Encoding live is still not connected, try to connect MFVP Encoding live to another port in the network switch.
  - If the LED status means that MFVP Encoding live is still not connected, connect the network switch to another of MFVP Encoding live Ethernet ports.

- If the LED status means that MFVP Encoding live is still not connected, try to use another Ethernet cable.
- If the LED status means that MFVP Encoding live is still not connected, contact support.

My encoding session is started but I cannot view the video stream on my decoder

Problem description

When my encoding session is started, the bit rate activity is displayed, but the decoder does not display the video source.

Tests to perform

Check the input signal: From the web interface, check the video input preview (*see* "Configuring the video input settings" on page 67).

- If you see no preview:
  - Check that the video standard is correct (PAL/NTSC).
  - Check the source signal by connecting it to a monitor.
- If you have a preview:
  - Check that an output is selected for the profile.
  - Check that a player can receive the session (either using VLC or a set-top box). If you cannot receive any session, check that you can ping MFVP Encoding live on the output specified in the session network parameters *(see "Configuring the Ethernet connectors" on page 31)*.

How to restart MFVP Encoding live

You can restart MFVP Encoding live via the web interface (*see* "Shutting down or restarting the system" on page 243).

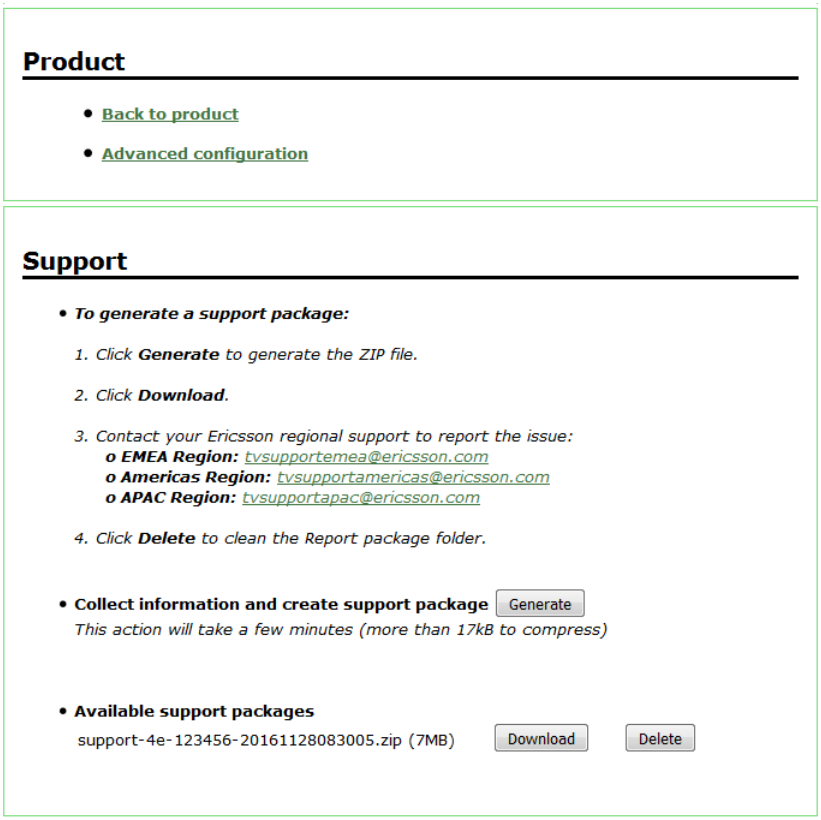
How to report a problem to the support team

To optimize the support process and bug tracking, a report package should be sent to the Support team.

CAUTION You should only generate a support package when the encoder is stopped or removed from active operation or during a non-critical time.

To generate a report package, follow these steps:

1. Open a browser and enter the following address to connect to the support web page:
   `http://<MFVP-EncodingLive_IP_Address>/Support.html`
   The following page is displayed:



2. Click **Generate** to generate the ZIP file.

3. Click **Download**.
4. Contact your Ericsson regional support (*See "Support information" on page 21*) to report the issue.
5. Once the support package is uploaded, click **Delete** to clean the Report package folder.

Glossary

AAC

Advanced Audio Coding, MPEG-2 or MPEG-4.

ABR

Available Bit Rate.

AFD

Active Format Descriptor. Standard set of codes that can be sent in the MPEG video stream or in the baseband SDI video signal that carries information about their aspect ratio and active picture characteristics.

artifacts

Flaws in the video encoding such as circles, blemishes, noise or spots.

ATSC

Advanced Television Systems Committee. An international digital television standard adopted by the United States, Canada, South Korea, Taiwan and Argentina. Broadcast over the air (terrestrial), via satellite or cable, ATSC transmits in a 6 MHz channel and uses MPEG-2 video compression and Dolby Digital audio compression.

AVI

Audio Video Interleaved. A container for a video and audio format.

B-frame

Bi-directional MPEG video frames are calculated from the content of the preceding video frame and the following video frame.

bandwidth

Bandwidth has many meanings, depending on context. It started as a radio term and has been expanded to include other kinds of communications. It is the numerical difference between the highest and lowest frequencies of a radio band or channel.

Bandwidth means the same thing in audio, although in most applications, the lowest frequency is sufficiently low enough to be considered equal to zero. Hence, audio bandwidth usually means the highest frequency of an audio signal or the highest frequency that can be carried by an audio system.

Typical examples of bandwidth are 300 - 3,400 Hz for telephone, 20 - 20,000 Hz for Hi-fi, and 20 - 15,000 Hz for fm.

bit rate

The rate (in Mbits per second) for transmitting data over a network. MFVP Encoding live refers to several bit rates. The target bit rate is the channel capacity of the network. The maximum bit rate is a limit, less than the target bit rate, that the fitter uses to fit data into the actual bandwidth for a streaming server. The audio encoder uses an average target bit rate and a maximum target bit rate for variable bit rate encoding.

broadband

A general term for DSL lines, cable modem, and t1 lines.

CAPEX

CAPital EXpenditure. Capital expenditures refer to the cost of developing or providing non-consumable parts for the product or system.

CBR

Constant bit rate.

CELP

Code Excited Linear Prediction. Audio encoding scheme used for very low bit rate encoding, mainly for speech.

CIF

Common Interchange Format. A video size (352 x 288) well suited to higher bit rates.

codec

The software that encodes and decodes a file. The word is a shortened form of coder-decoder or compression-decompression. MFVP Encoding live has an audio codec and a video codec.

contrast

The tonal difference between light, midtone and dark zones of an image.

D1

Video resolution standard. For NTSC, D1 is 720 x 480 pixels, in the PAL and SECAM systems, D1 is 720 x 576.

ECM

Entitlement Control Messages.

ECMG

Entitlement Control Messages Generator.

Fps

Frames per second. Number of video frames displayed within one second.

frame size

Size (in pixels) of a video frame (for example, 640 x 480).

GOP

Group of pictures. In Video coding, a group of pictures, or GOP structure, specifies the order in which intra- and inter-frames are arranged.

GMT

Greenwich Mean Time.

hinted movies

Movies that can be streamed for broadcast over the Internet. A hint track has information about the media file so that it can be streamed properly.

HRD

Hypothetical Reference Decoder.

I-frames

An MPEG-4 video term for key frames.

image source

Optical device (camera, webcam, TV, or VCR) that can create a digital representation of a view, scene or picture.

ISO

International Standards Organization.

kbps

Kilobits per second.

key frame

A video frame that fully refreshes its contents and can be used as a reference point for other frames in the sequence. In MPEG terms, this is an I-frame.

LATM

Low Overhead Audio Transport Multiplex.

Lo/Ro

Left only/Right only: A downmix from a multichannel to a two-channel output that is compatible for stereo or mono reproduction.

Lt/Rt

Left total/Right total: A downmix from a multichannel to a two-channel output that is Dolby Surround compatible.

MPEG

As defined on the MPEG website, MPEG (pronounced M-peg), stands for Moving Picture Experts Group, and is the name given to a family of International Standards used for coding audio-visual information in a digital compressed format. The MPEG family of standards includes MPEG-1, MPEG-2, MPEG-4, MPEG-7, and MPEG-21, which are formally known as iso/iec-11172, iso/iec-13818, iso/iec-14496, iso/iec-15938, and iso/iec-21000.

MTU

Maximum Transmission Unit.

multicast

Process where a single stream is served from one source to multiple receivers. It is a one-to-many communication.

The multicast address range is: 224.0.0.0 - 239.255.255.255.

multimedia

A single presentation with multiple forms of media—text, graphics, sound, video, and animation. Most personal computers can display multimedia, and the source is usually a CD because the files are large and require considerable storage capacity. MPEG-4 gives broadcasters the tools to deliver multimedia presentations over the Web in smaller file size.

noise

Random flaws that appear in audio or video.

NTP

Network Time Protocol. NTP is a protocol designed to synchronize the clocks of computers over a network.

NTSC

National Television Systems Committee. Determines the standards for color picture broadcasting. Used in the USA, Canada and Japan.

OPEX

OPerational EXpenditure. Operating expenditures are the on-going costs for running a product, business, or system.

PAL

Phase Alternating Line. Standard for television broadcast and reception used in Europe.

PCR

Program Reference Clock. MPEG-2 concept. The PCR is usually used for synchronization.

PID

Packet IDentifier. Each packet in the transport stream, whether it contains audio, video, tables or data, is identified by a number called a PID. PIDs enable the decoder to sort through the packets in a transport stream.

PiP

Picture in Picture. Allows one to watch more than one TV program at the same time on television sets or other devices. With PiP feature, one program will be displayed on the entire TV screen, and another program or programs will be displayed in individual smaller squares on the screen.

pixel

A picture element, which is one point in a graphic image.

PMT

Each PMT, or Program Map Table literally maps out a specific program, listing the PID values for the packets containing the program's video, audio and data components.With this information, the decoder can easily locate, decode and display the program's contents.

POIS

Placement Opportunity Information Service. The POIS is the system and process that identifies and provides descriptions of placement opportunities for media (such as the availability to insert ads).

program

For MPEG-2, a program is a collection of video, audio, data elementary streams, and tables carried on a channel that share a common clock reference.

QCIF

Quarter Common Interchange Format. A video size (176 x 144) well suited to lower bit rates.

real-time

Rapid transmission and processing of event-oriented data and transactions as they occur instead of being stored and retransmitted or processed as batches.

resolution

Determined by the number of pixels displayed per line or for a given area. Higher resolution means better picture quality.

RTCP

Real-time Transport Control Protocol. Protocol that works in conjunction with RTP. RTCP provides control services.

RTP

Real-time Transport Protocol. Transport protocol used to deliver live media to one or more viewers simultaneously. RTP is the transfer protocol for RTSP streaming

sample rate

The speed at which analog audio data is collected.

SD

Standard Definition. A set of digital television (DTV) standards with 480 lines of pixels from top to bottom for NTSC format (or 576 lines for PAL format).

set-top box

A device that provides access to the Broadband broadcast and Internet and displays information on a TV screen.

streaming

Sending media streams from a live encoder or streaming server to a player as a steady, continuous stream.

TDT

Time and Date Table

TOT

Time Offset Table

unicast

Unicast is communication between a single sender and a single receiver over a network.

URL

Uniform Resource Locator. A method of naming documents or places on the Internet.

VBI

Vertical Blanking Interval. The part of a TV signal that is sent between each video frame. Its purpose is to allow the TV time to reposition its electron beam from the bottom of the current frame (screen) to the top of the next one. This non-viewable part of the signal is used to transmit closed-caption content.

VBR

Variable bit rate.

VBV

The Video Buffering Verifier implements the leaky bucket algorithm. Leaky bucket is a flow control mechanism that is designed to reduce the effect of the inevitable variability in the input stream into a node of a communication network. The key concept is that if the encoder receives too much data or very complex data, the bitrate will not be respected and the bucket may overflow.

WSS

Wide Screen Signaling is a digital stream embedded in the TV signal describing qualities of the broadcast, in particular the intended aspect ratio of the image. This can be used by a widescreen TV to switch the correct display mode.

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