ADOBE® MEDIA SERVER 5.0.1
Developer's Guide

Legal notices

Contents

Chapter 1: Getting started streaming media

Chapter 2: Content protection

Chapter 3: Getting started developing applications

Chapter 4: Developing streaming media applications

Chapter 5: Working with live video

Chapter 6: Building peer-assisted networking applications

Chapter 7: Developing social applications

Chapter 8: Securing applications

Chapter 9: Developing Plug-ins

Chapter 1: Getting started streaming media

Supported clients, encoders, codecs, and file formats

Supported clients and servers for streaming services

Streaming services are pre-built Adobe Media Server applications. Use streaming services to stream media to Flash, AIR, and Apple (iOS and QuickTime) clients. The services are installed to *rootinstall*/applications/*servicename*.

The following table lists the streaming services and their earliest supported server versions, server editions, and client runtimes:

Streaming type	**Service name**	**Server version**	**Server edition(earliest supported)**	**Client runtime (earliest supported)**
On-demand streaming over RTMP. See Stream on-demand media (RTMP).	vod	3	All	Flash Player 6 AIR 1
On-demand streaming over HTTP. See "Stream on-demand media (HTTP)" on page 19.	None	4.5	All	Flash Player 10.1 AIR 2 iOS 3.0 QuickTime X
Live streaming over RTMP. See "Stream on-demand media (RTMP)" on page 24.	live	3	All	Flash Player 6 AIR 1
Live streaming over HTTP. See "Stream live media (HTTP)" on page 5.	livepkgr	4	4—Adobe Media Server Extended, Adobe Media Server Professional 4.5—All	4—Flash Player 10.1, AIR 2 4.5—Flash Player 10.1, AIR 2, iOS 3.0, QuickTime X

Streaming type	Service name	Server version	Server edition(earliest supported)	Client runtime (earliest supported)
Multicast streaming over RTMFP. See "Multicast media (RTMFP)" on page 36.	multicast	4	Adobe Media Server Professional supports IP Multicast Adobe Media Server Extended supports IP Multicast, Application-level Multicast, and Multicast Fusion. Multicast Fusion technology combines IP and Application-level Multicast. The server editions have changed with server version 5. Adobe Flash Media Streaming Server is Adobe Media Server 5 Standard. Adobe Flash Media Interactive Server is Adobe Media Server 5 Professional. Adobe Flash Media Enterprise Server is Adobe Media Server 5 Extended. Adobe Flash Media Development Server is Adobe Media Server 5 Starter.	Flash Player 10.1 AIR 2

***Important:** On Adobe Media Server 5 Standard, you cannot modify the server-side code in the streaming services. In all other editions of the server, you can modify the code.*

Supported file formats and codecs

Adobe Flash Platform

For a complete list of supported file formats and codecs, see Supported file formats and codecs.

Live HTTP Dynamic Streaming supports the H.264, VP6, MP3, and AAC codecs.

For on-demand HTTP Dynamic Streaming, the just-in-time packager supports F4V/MP4 files. This document explains how to use the just-in-time packager. To package FLV files for HTTP Dynamic Streaming, use the offline File Packager tool. See File Packager reference.

In addition, see the following Flash Platform articles:

Smart phone and tablet video encoding recommendations for Flash Player and AIR by Adobe encoding evangelist, Maxim Levkov, and Adobe Product Manager, Tom Nguyen.

Video encoding and transcoding recommendations for HTTP Dynamic Streaming on the Flash Platform by Adobe encoding evangelist, Maxim Levkov.

For a longer list of articles about encoding, see the Video Encoding page on Adobe Developer Connection.

Apple HTTP Live Streaming

Adobe Media Server support for Apple HTTP Live Streaming includes H.264 and AAC/HE-AAC for audio-video content and AAC/HE-AAC for audio-only content.

For more information, see Apple HTTP Live Streaming Overview - FAQ.

For recommended encoding settings, see Best Practices for Creating and Deploying HTTP Live Streaming Media for the iPhone and iPad and Recommended Encoding Settings for HTTP Live Streaming Media.

Supported encoders

Use the following encoders to publish a live stream to Adobe Media Server:

- Flash Media Live Encoder
- Server-side scripts running on Adobe Media Server Extended, Adobe Media Server Professional, and Adobe Media Server Starter. When you call the Server-side ActionScript method `Stream.play()`, the stream is considered "live".
- A custom-built Flash Player or AIR application. See "Working with live video" on page 235.
- Third-party encoding solutions. See Adobe Media Server Solution Partners.

Pre-built media players

Flash and AIR media players

Media player	**Supported protocols**	**Media player location**
Flash Media Playback	RTMP/x, HTTP Note: Flash Media Playback doesn't support Set-level Manifest files.	www.adobe.com/products/flashmediaplayback/
Strobe Media Playback	RTMP/x, HTTP	osmf.org/developers.html
Adobe Media Server sample video player The AMS sample video player is Strobe Media Playback in a wrapper. The wrapper includes sample and generates HTML embed code.	RTMP/x, HTTP To use this player for multicast playback, use the multicast configurator to create a manifest file. Use the manifest file as the source.	*rootinstall*/samples/videoPlayer
Adobe Media Server multicast sample player	RTMFP	*rootinstall*/tools/multicast/multicastplayer
Flash FLVPlayback component	RTMP/x	fl.video.FLVPlayback FLV playback 2.5 supports DVR
Flex components	RTMP/x	mx.controls.VideoDisplay spark.components.VideoPlayer spark.components.VideoDisplay

***Note:** The previous table is not a complete list of supported media players. Many third parties have developed excellent media players. When using a third party media player, please check with the third party to verify which features the media player supports.*

Play media in Flash Media Playback

Flash Media Playback is a compiled SWF file hosted on Adobe.com. Flash Media Playback is built on the Open Source Media Framework (OSMF). Flash Media Playback is not open source, but it is fully configurable and supports dynamically loaded plug-ins from third-party service providers. It runs in Flash Player 10.1 and AIR 2 and supports multi-bitrate streaming, Adobe HTTP Dynamic Streaming (streaming over HTTP), and DVR.

***Note:** Flash Media Playback does not support set-level manifest files; use Strobe Media Playback instead. Or, to use multi-bitrate HTTP Dynamic Streaming with Flash Media Playback, use a Manifest.xml file.*

For more information about Flash Media Playback, see www.adobe.com/products/flashmediaplayback.

1. Load the Flash Media Playback Setup page in a web browser: www.osmf.org/configurator/fmp/.
2. Enter the Video Source.
3. Indicate whether the video uses HTTP Dynamic Streaming or Adobe Access.
4. To use DVR, click the Advanced tab and select Stream Type DVR.
5. Click Preview to update the embed code.
6. Click Play to test the code.
7. To use the player in your own HTML page, copy the embed code and paste it into your page.

Play media in Strobe Media Playback

Strobe Media Playback is built with the Open Source Media Framework (OSMF). It supports progressive download, RTMP streaming, HTTP Dynamic Streaming (including adaptive bitrate manifest files), multicast streaming, and content protection with Adobe® Flash® Access™. You must host Strobe Media Playback on your own server.

Download the latest Strobe Media Playback and its documentation from osmf.org.

Play media in the Adobe Media Server sample video player

The sample player that installs with Adobe Media Server to *rootinstall*/samples/videoPlayer is based on Strobe Media Playback. You can use the sample player to generate HTML embed code to use in your own HTML page.

***Important:** To play the sample F4M manifest files in the sample video player, enter a localhost address in the Stream URL box. For example, http://localhost/vod/hds_sample1_manifest.f4m.*

Embed the Adobe Media Server sample video player in an HTML file

1. Browse to the sample video player and open it in a browser. The file is located at *rootinstall*/samples/videoPlayer/videoplayer.html.
2. Enter the URL of the video to play and click Stream. The sample player adds the URL of the video file to the embed code.

   To play an on-demand file from the vod service, copy the file to the *rootinstall*/applications/vod/media folder. The URL is something like the following:

   ```
   rtmp://localhost/vod/mp4:sample1_1500kbps.f4v
   ```

   For more information, see "URLs for playing on-demand media files over RTMP" on page 26, "URLs for publishing and playing live streams over RTMP" on page 18, "URLs for publishing and playing live streams over HTTP" on page 12, and "URLs for playing on-demand streams over HTTP" on page 23.

3 Find the section on the page labeled “Embed Code”.

4 Copy the code from the text field at the bottom of the page. The code begins and ends with `<object>` tags.

5 Create a new HTML page.

6 Paste the code from the sample video player into the HTML file between the `<body></body>` tags.

7 Save the HTML page to the *rootinstall*/webroot folder.

__Important:__ Do not name the file “index.html”. The Adobe Media Server Start Screen is the index.html file in the webroot folder. If you save a file named index.html to the webroot folder, the file will overwrite the Start Screen.

The embed code looks for the SampleMediaPlayback.swf file at swfs/SampleMediaPlayback.swf. The SampleMediaPlayback.swf is installed to this folder by default. To serve the HTML page from a different web server, use the same folder structure.

8 Open your HTML file in a browser.

Apple media players

Apple HTTP Live Streaming supports both live content and on-demand content.

Media player	Supported streaming type
Devices running iOS 3.0 and later include built-in client software.	On-demand and live (HTTP)
On Mac OS 10.6 and later, Safari 4.0 and QuickTime X.	On-demand and live (HTTP)

For more information about Apple media players, see Apple HTTP Live Streaming Overview.

Stream live media (HTTP)

Prerequisites for live streaming over HTTP

To use HTTP Dynamic Streaming (HDS) and HTTP Live Streaming (HLS) to serve live streams to clients over HTTP, publish the streams to the HTTP Live Packager service on Adobe Media Server (*rootinstall*/applications/livepkgr). The livepkgr service ingests the streams, packages them into fragments, and delivers the fragments to Flash and iOS clients in real-time.

To complete these tutorials, use the following software:

- Adobe Media Server 5. See the video Install Adobe Media Server 5 and verify HTTP streaming to Flash and iOS.
- Flash Media Live Encoder

  __Note:__ Flash Media Live Encoder on Windows doesn't support AAC encoding. To add support for AAC, purchase the MainConcept AAC Encoder.

- (Adobe HTTP Dynamic Streaming) Adobe Media Server sample video player (which uses Strobe Media Playback) and Flash Player 10.1.
- (Apple HTTP Live Streaming) iOS 3.0 or later device or Mac OS 10.6 with Safari 4.0 or QuickTime X.

For more information, see “Supported clients, encoders, codecs, and file formats” on page 1 and “Pre-built media players” on page 3.

Publish and play a single live stream over HTTP

1 Install Adobe Media Server 5 and choose to install Apache HTTP Server.

2 Install Flash Media Live Encoder and configure it to use absolute time.

a Close Flash Media Live Encoder.

b Open the Flash Media Live Encoder *rootinstall*\conf\config.xml file in a text editor.

The default installation location on Windows is C:\Program Files\Adobe\Flash Media Live Encoder 3.2.

The default installation location on Mac OS is Macintosh HD:Applications:Adobe:Flash Media Live Encoder 3.2.

c Set the tag //flashmedialiveencoder_config/mbrconfig/streamsynchronization/enable to true:

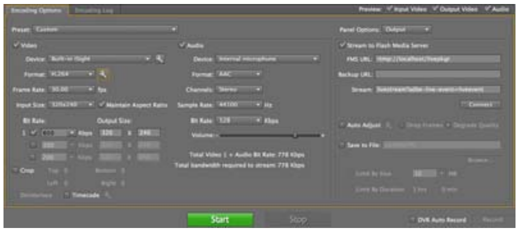
```
<flashmedialiveencoder_config>
    <mbrconfig>
        <streamsynchronization>
            <!-- "true" to enable this feature, "false" to disable.-->
            <enable>true</enable>
```

d Save the file.

3 To publish a live stream to Adobe Media Server, start Flash Media Live Encoder and do the following:

a In the Encoding Options panel, from the Preset pop-up menu, choose High Bandwidth (800 Kbps) — H.264. For Audio Format, choose AAC.

Note: *Flash Media Live Encoder on Windows doesn't support AAC encoding. To add support for AAC, purchase the MainConcept AAC Encoder.*

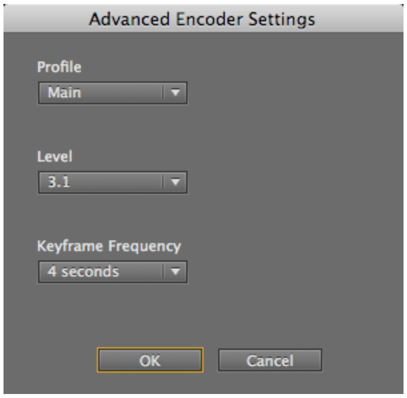


Flash Media Live Encoder Encoding Options panel

b Click the wrench next to Format to open Advanced Encoder Settings and do the following:

- Profile—Main
- Level—3.1
- Keyframe Frequency—4 or a multiple of `<FragmentDuration>` in the applications/livepkgr/events/_definst_/liveevent/Event.xml file. The default value of `<FragmentDuration>` is 4000 milliseconds.

  Note: *For information about profile and level support, see "Supported file formats and codecs" on page 2.*



Advanced Encoder Settings

Profile: Main

Level: 3.1

Keyframe Frequency: 4 seconds

Flash Media Live Encoder Advanced Encoder Settings

c In the AMS URL text box enter the following:

rtmp://localhost/livepkgr

Note: *Use the RTMP protocol to stream the content to the livepkgr application on Adobe Media Server. You can substitute an IP address or a domain name for "localhost".*

d In the Stream text box enter the following:

livestream?adbe-live-event=liveevent

Note: *The applications/livepkgr/main.asc file expects this value.*

e Deselect Save to File.

f Click Start.

4 (Adobe HTTP Dynamic Streaming) To play the media in Flash Media Playback, do the following:

a Open Flash Media Playback in a web browser:

http://www.osmf.org/configurator/fmp/

Note: *Flash Media Playback requires Flash Player 10.1 to support HTTP Dynamic Streaming.*

b In Video Source, enter:

http://localhost/hds-live/livepkgr/_definst_/liveevent/livestream.f4m

For information about the request URL, see "URLs for publishing and playing live streams over HTTP" on page 12.

c Select Yes under the question "Are you using HTTP Streaming or Adobe Access 2.0?".

d Click Preview. Click Play.

5 (Adobe HTTP Dynamic Streaming) To play the media in Strobe Media Playback, do the following:

a Open the Adobe Media Server sample video player in a web browser. Browse to *rootinstall*/samples/videoPlayer/videoplayer.html

b In STREAM URL, enter:

http://localhost/hds-live/livepkgr/_definst_/liveevent/livestream.f4m

For information about the request URL, see "URLs for publishing and playing live streams over HTTP" on page 12.

   c. Select LIVE and Click PLAY STREAM.

6. (Apple HTTP Live Streaming) Use the following URL:

   http://10.0.1.11/hls-live/livepkgr/_definst_/liveevent/livestream.m3u8

   For information about where to use the URL to serve various iOS devices, QuickTime, and Safari, see HTTP Live Streaming Overview in the iOS Reference Library.

   For information about the request URL, see “URLs for publishing and playing live streams over HTTP” on page 12.

***Note:** Replace “localhost” or “10.0.1.11” with the domain name or IP address of the computer hosting Adobe Media Server. If you’re using a Adobe Media Server hosting provider, they can give you this value.*

More Help topics

“Troubleshoot issues with streaming media” on page 111

“Disk management” on page 70

“Content storage (HDS and HLS)” on page 56

“Configure HTTP Dynamic Streaming and HTTP Live Streaming” on page 51

“Stream live media (RTMP)” on page 17

Publish and play live multi-bitrate streams over HTTP

1. Install Adobe Media Server 5 and choose to install Apache HTTP Server.
2. Install Flash Media Live Encoder and configure it to use absolute time.
   a. Close Flash Media Live Encoder.
   b. Open the Flash Media Live Encoder *rootinstall*\Conf\config.xml file in a text editor.
   c. Set the tag //flashmedialiveencoder_config/mbrconfig/streamsynchronization/enable to true:

```
<flashmedialiveencoder_config>
    <mbrconfig>
        <streamsynchronization>
            <!-- "true" to enable this feature, "false" to disable.-->
            <enable>true</enable>
```

   d. Save the file.

3. Browse to rootinstall/applications/livepkgr/events/_definst_/liveevent and do the following:
   - Edit the Event.xml file to look like the following:

```
<Event>
    <EventID>liveevent</EventID>
        <Recording>
                <FragmentDuration>4000</FragmentDuration>
                <SegmentDuration>16000</SegmentDuration>
                <DiskManagementDuration>3</DiskManagementDuration>
        </Recording>
</Event>
```

   - Remove the Manifest.xml file from the liveevent folder or rename it.

4 (Adobe HTTP Dynamic Streaming) For multi-bitrate streaming, Flash and AIR media players request an F4M manifest file that contains the location and bitrate of each live stream. This type of F4M file is called a *set-level manifest*. To create an set-level manifest, do the following:

a Open *rootinstall*/tools/f4mconfig/configurator/f4mconfig.html in a browser.

b Select the f4m file type.

c Stream URIs can be absolute or relative to a Base URI. Enter the following for the Base URI:

http://localhost/hds-live/livepkgr/_definst_/liveevent

d Enter the following for each stream and click Add:

Stream URI	Bitrate
livestream1.f4m	150
livestream2.f4m	500
livestream3.f4m	700

e To preview the file, click View Manifest. The manifest file looks like this:

```
<manifest xmlns="http://ns.adobe.com/f4m/2.0">
    <baseURL>http://localhost/hds-live/livepkgr/_definst_/liveevent/</baseURL>
    <media href="livestream1.f4m" bitrate="150"/>
    <media href="livestream2.f4m" bitrate="500"/>
    <media href="livestream3.f4m" bitrate="700"/>
</manifest>
```

f Click Save Manifest and save the file as liveevent.f4m to *rootinstall*/webroot.

The media player requests this file from a web server. This tutorial saves the file to *rootinstall*/webroot, but the file can be served from any location on any webserver. This file does not need to live on Adobe Media Server. This file can also have any name.

5 (Apple HTTP Live Streaming) iOS devices request an M3U8 variant playlist file that contains the location, bitrate, and optionally the codec of each stream. To create an M3U8 file, do the following:

a If the Set-level F4M/M3U8 File Generator tool isn't open, double-click *rootinstall*/tools/f4mconfig/configurator/f4mconfig.html to open it in a browser.

b Select m3u8.

c Enter the Stream URI and bitrate for each stream. Stream URIs can be absolute or relative. If they are relative, they are relative to the m3u8 file.

***Note:** You can't use a Base URI when generating an M3U8 file.*

This tutorial uses the following Stream URI settings:

Stream URI	Bitrate
http://localhost/hls-live/livepkgr/_definst_/liveevent/livestream1.m3u8	150
http://localhost/hls-live/livepkgr/_definst_/liveevent/livestream2.m3u8	500
http://localhost/hls-live/livepkgr/_definst_/liveevent/livestream3.m3u8	700

d For m3u8 files, you can optionally add a codec for each stream.

If one stream is audio-only, specify an audio codec. Specify audio and video codecs for the other streams in the manifest. See "Publish an audio-only stream (HLS)" on page 11.

e To view the file, click View Manifest. The m3u8 file looks like this:

```
#EXTM3U
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=150000
http://localhost/hls-live/livepkgr/_definst_/liveevent/livestream1.m3u8
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=500000
http://localhost/hls-live/livepkgr/_definst_/liveevent/livestream2.m3u8
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=700000
http://localhost/hls-live/livepkgr/_definst_/liveevent/livestream3.m3u8
```

f Save the file as liveevent.m3u8 to the folder *rootinstall*/webroot.

The media player requests this file from a web server. This tutorial saves the file to *rootinstall*/webroot, but the file can be served from any location on any webserver. This file does not need to live on Adobe Media Server.

6 To publish a live stream to Adobe Media Server, start Flash Media Live Encoder and do the following:

a In the Encoding Options panel, from the Preset pop-up menu, choose Multi Bitrate - 3 Streams (1500 Kbps) - H.264. Choose Audio Format AAC.

***Note:** Flash Media Live Encoder on Windows doesn't support AAC encoding. To add support for AAC, purchase the MainConcept AAC Encoder.*

b Click the wrench next to Format to open Advanced Encoder Settings. For Keyframe frequency, select 4 seconds.

***Note:** This value matches the* `<FragmentDuration>` *value in the applications/livepkgr/events/_definst_/liveevent/Event.xml file. The* `<FragmentDuration>` *value is in milliseconds.*

c For Bit Rate, choose 150, 500, and 700.

d In the AMS URL text box, enter the following:

rtmp://localhost/livepkgr

***Note:** This application is installed with Adobe Media Server and contains a main.asc file and configuration files for live HTTP Dynamic Streaming.*

e In the Stream text box, enter the following:

livestream%i?adbe-live-event=liveevent

***Note:** The applications/livepkgr/main.asc file expects this value. Flash Media Live Encoder uses the variable %i to create multiple stream names: livestream1, livestream2, livestream3, and so on. To use another encoder, provide your own unique stream names, for example, **livestream1?adbe-live-event=liveevent**, **livestream2?adbe-live-event=liveevent**.*

f Deselect Save to File.

g Click Start.

7 (Adobe HTTP Dynamic Streaming) Flash Media Playback does not support set-level manifest files. To play the media, use Strobe Media Playback. Strobe Media Playback is used in the sample video player that installs with Adobe Media Server.

a Copy the videoPlayer directory from *rootinstall*/samples/videoPlayer to *rootinstall*/webroot.

b Browse to the sample player in a web browser:

http://localhost/videoPlayer/videoplayer.html

c In Video Source, enter the following:

http://localhost/liveevent.f4m

8 (Apple HTTP Live Streaming) In iOS, enter the following URL in Safari:

http://localhost/liveevent.m3u8

For information about where to use the URL to serve various iOS devices, QuickTime, and Safari, see HTTP Live Streaming Overview in the iOS Reference Library.

***Note:** You can replace localhost with the domain name or IP address of the computer hosting Adobe Media Server. If you're using a Adobe Media Server hosting provider, they can give you this value.*

Setting the record option while publishing to livepkgr

You can define the record option by adding a query parameter string `adbe-record-mode`. The value of this string can be "record" or "append". If the parameter is not specified, "append" will be used. In case of "record", livepkgr overwrites the previous recording for the same stream and event name. In case of "append", livepkgr application appends to the previous recording for the same stream and event name.

For example, to publish a stream in record mode, use:

```
livestream%i?adbe-live-event=liveevent&adbe-record-mode=record
```

To publish a stream in append mode (default case), use:

```
livestream%i?adbe-live-event=liveevent&adbe-record-mode=append
```

Packaging an audio-only stream (HLS)

To serve streams over a cellular network, one of the streams must be audio-only. For more information, see HTTP Live Streaming Overview.

AAC-encoded audio files (MP4/F4V) are packaged as .aac file. Also, the M3U8 file will list the file names as .aac instead of .ts. An audio-only stream functionality will be available only if the M3U8 URL has the following embedded tags:

Tag	Description
audio-only	Specifies whether the stream/file is audio-only. If the tag is not present, the audio-only stream will be extracted from the A/V stream.
audio-only-mp3/audio-only-aac	Specifies the audio codec of the requested stream file. If this tag is not present, the file/segment will be segmented as TS.

Here is an example of a valid M3U8 URL:

```
http://myserver/hls-vod/audio-only-aac/sample1_1500kbps.f4v.m3u8
```

Here is an example M3U8 file snippet:#EXTM3U_#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=1500000,CODECS="avc1.42001e, mp4a.40.2"http://10.40.23.222/hls-vod/sample1_1500kbps.f4v.m3u8#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=150000,CODECS="avc1.42001e, mp4a.40.2"_http://10.40.23.222/hls-vod/sample1_150kbps.f4v.m3u8#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=1000000,CODECS="avc1.42001e, mp4a.40.2"_http://10.40.23.222/hls-vod/sample1_1000kbps.f4v.m3u8#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=64000,CODECS="mp4a.40.2"http://10.40.23.222/hls-vod/audio-only-aac/sample1_1000kbps.f4v.m3u8

Publish an audio-only stream (HLS)

To publish an audio-only stream, enter the following in the Adobe Media Encoder Stream field:

livestream*%i?adbe-live-event=*liveevent*&adbe-audio-stream-name=*livestream1_audio_only*&adbe-audio-stream-src=*livestream1

If the encoder specifies individual query strings for each stream, use individual stream names instead of the variable `%i`:

livestream1*?adbe-live-event=*liveevent*&adbe-audio-stream-name=*livestream1_audio_only

livestream2*?adbe-live-event=*liveevent*&adbe-audio-stream-name=*livestream2_audio_only

***Note:** You can use the values in italics or replace them with your own values. The default live event is called "liveevent".*

Parameter	Description
`adbe-audio-stream-name`	The name of the audio-only stream.
`adbe-audio-stream-src`	The name of the stream from which to extract the audio for the audio-only stream. If this parameter is not specified, the server uses the stream on which the `adbe-audio-stream-name` parameter was specified.

To generate a set-level variant playlist when using an audio-only stream, specify the audio codec of the audio-only stream. Specify the audio and the video codec of the streams that contain audio and video. For more information about using the Set-level F4M/M3U8 File Generator, see "Publish and play live multi-bitrate streams over HTTP" on page 8.

***Note:** It is highly recommended to use the "Audio-only packaging" as mentioned in previous section rather than publishing audio-only streams.*

URLs for publishing and playing live streams over HTTP

***Important:** The format for HDS request URLs has changed in Adobe Media Server 5. The Apache httpd.conf file redirects requests in the 4.0 format to the 4.5 format. The path /live/events/livepkgr/events redirects to the path /hds-live/livepkgr.*

Use the following URL to publish a single live stream to Adobe Media Server for streaming over HTTP:

rtmp://*ams-ip-or-dns*/livepkgr/livestream?adbe-live-event=liveevent

Use the following URL to publish multiple live streams to Adobe Media Server for adaptive bitrate streaming:

rtmp://*ams-ip-or-dns*/livepkgr/livestream%i?adbe-live-event=liveevent

***Note:** Publish the stream over RTMP. Clients play the stream over HTTP.*

Use the following URLs to play live streams over HTTP:

Client	Single stream or Adaptive Bitrate	Request URL
Adobe HDS	Single stream	http://<ams-dns-or-ip>/**hds-live**/livepkgr/<instname>/<eventname>/<streamname>.**f4m**
Apple HLS	Single stream	http://<ams-dns-or-ip>/**hls-live**/livepkgr/<instname>/<eventname>/<streamname>.**m3u8**
Adobe HDS	Adaptive bitrate	The media player requests a set-level manifest file: http://<webserver-dns-or-ip>/<set-level-manifest>.**f4m** The set-level manifest file contains URLs for each live stream in the following format: http://<ams-dns-or-ip>/**hds-live**/livepkgr/<instname>/<eventname>/<streamname>.**f4m**

Client	Single stream or Adaptive Bitrate	Request URL
Apple HLS	Adaptive bitrate	The media player requests a set-level variant playlist file: http://<webserver-dns-or-ip>/<set-level-variant-playlist>.**m3u8** The set-level variant playlist contains URLs for each live stream in the following format: http://<ams-dns-or-ip>/**hls-live**/livepkgr/<instname>/<eventname>/<streamname>.**m3u8**

Important: *To play streaming media over HTTP, a media player requests a manifest file (F4M or M3U8). The server generates manifest files in real-time. The files are not written to disk; you cannot see them on the server.*

The path /hds-live is a `<Location>` directive in the Apache httpd.conf file that tells the server to package the content for Adobe HTTP Dynamic Streaming (HDS).

The path /hls-live is a `<Location>` directive in the Apache httpd.conf file that tells the server to package the content for Apple HTTP Live Streaming (HLS).

For adaptive streaming, the client requests a set-level manifest. For HDS, this file is a .f4m file. For HLS, this file is a .m3u8 file. The set-level manifest can live on any web server. You can create multiple set-level manifest files for a single set of live streams.

The set-level manifest contains the paths to the F4M manifest files (HDS) and M3U8 variant playlists (HLS) of each live stream. The paths must begin with /hds-live or /hls-live to tell the server to package the streams for HTTP.

Note: *Do not include /hds-live or /hls-live in the URL that requests the set-level manifest.*

You can configure the `<Location>` directive settings and the content location in the Apache httpd.conf file. For more information, see Content storage (HDS and HLS).

Differences in HTTP live URLs from Flash Media Server 4.0 to Adobe Media Server 5

The default URL to play a live stream over HTTP has changed in Adobe Media Server 5. In Flash Media Server 4.0, the URL to play a live stream was http://<servername>/live/events/livepkgr/events/<eventname>. The Adobe Media Server 5 Apache httpd.conf file uses a 301 redirect to redirect requests that use a 4.0 URL to the 4.5 URL.

For more differences, see "Differences in HTTP Dynamic Streaming between Flash Media Server 4.0 and 4.5" on page 54.

Configure DVR (HDS)

Configure DVR on the server

1 Open the Set-level F4M/M3U8 File Generator in a browser: *rootinstall*/tools/f4mconfig/configurator/f4mconfig.html.

2 In addition to entering values for Stream URI and Bitrate, enter a value for DVR Window Duration. This value sets the amount of viewable content, in minutes, before the live point.

   Use the following guidelines to set a Window Duration:

   - Set Window Duration to a value greater than 0.

     Setting the Window Duration to 0 can cause a bad user experience.

- A value of -1 indicates that the available recorded content behind the live point is unlimited.

3 In the Event.xml file, configure the `DiskManagementDuration` element to specify the amount of content the server caches. By default, the server caches 3 hours of content. Use the following formula to determine the value of the Window Duration in relation to the value of `DiskManagementDuration`:

```
HDSWindowDuration <= (DiskManagementDuration - SegmentDuration)
```

For more information about disk managment, see "Disk management" on page 70.

Publish a DVR stream

To publish a DVR stream from Flash Media Live Encoder, do not click Record or check DVR Auto Record. Publish the stream just as you publish any live stream.

Play DVR streams

Strobe Media Playback supports DVR streams by default.

***Note:** Flash Media Playback does not support set-level manifest files. To use Flash Media Playback, configure a Manifest.xml file. See "Manifest.xml" on page 86.*

Configure a sliding window (HLS)

Configure a sliding window on the server

A *sliding window* is the seekable portion of the stream for Apple HTTP Live Streaming. Clients cannot seek beyond the sliding window length. HTTP Live Streaming clients use the sliding window to configure the seek bar.

Configure a sliding window at the following levels:

Level	**Configuration file**
Server	*rootinstall*/Apache2.2/conf/httpd.conf
Application	*rootinstall*/applications/livepkgr/Application.xml The livepkgr application is the default application for HTTP streaming. You can duplicate and rename this application.
Event	*rootinstall*/applications/livepkgr/events/_definst_/liveevent/Event.xml The liveevent folder is the default live event. You can create multiple live events within an application.

Use the following parameters:

Parameter	**Configuration file**	**Description**
`HLSSlidingWindowLength`	httpd.conf	The number of TS files available for seeking in a sliding window.
`SlidingWindowLength`	Application.xml and Event.xml	The number of TS files available for seeking in a sliding window.
`HLSMediaFileDuration`	httpd.conf	The length of a TS file, in milliseconds.
`MediaFileDuration`	Application.xml and Event.xml	The length of a TS file, in milliseconds.

The time within the sliding window is:

```
HLSSlidingWindowLength * HLSMediaFileDuration
SlidingWindowLength * MediaFileDuration
```

By default, `HLSSlidingWindowLength` is set to 6 seconds and `HLSMediaFileDuration` is set to 8000 milliseconds. Therefore, by default, all HLS live events are seekable within a window that is 48 seconds wide.

The sliding window is relative to the current position of the live stream. For example, if sliding window is configured to have 15 minutes of data and the event starts at time 0, when the live stream is at 30, the last seek position possible is 15.

To make an entire live event seekable, set `SlidingWindowLength` or `HLSSlidingWindowLength` to 0. However, doing so may impact performance.

Configure the sliding window to be smaller than the duration of content cached on disk. In the Event.xml file, the `DiskManagementDuration` element specifies the amount of content the server caches. By default, the server caches 3 hours of content. The size of the HLS sliding window must be as follows:

```
HLSSlidingWindow <= (DiskManagementDuration - SegmentDuration)
```

For more information about disk managment, see "Disk management" on page 70.

Configure a sliding window at the event level

The following Event.xml file creates a 1 hour sliding window for a single HLS live event:

```
<Event>
    <EventID>liveevent</EventID>
    <Recording>
        <FragmentDuration>4000</FragmentDuration>
        <SegmentDuration>400000</SegmentDuration>
        <DiskManagementDuration>3</DiskManagementDuration>
    </Recording>
    <HLS>
        <MediaFileDuration>8000</MediaFileDuration>
        <SlidingWindowLength>450</SlidingWindowLength>
    </HLS>
</Event>
```

Configure a sliding window at the application level

The following Application.xml file creates a 1 hour sliding window for all HLS live events within the livepkgr application:

```
<Application>
    <StreamManager>
        <Live>
            <AssumeAbsoluteTime>true</AssumeAbsoluteTime>
        </Live>
    </StreamManager>
    <HLS>
        <MediaFileDuration>8000</MediaFileDuration>
        <SlidingWindowLength>450</SlidingWindowLength>
    </HLS>
</Application>
```

Reload the livepkgr application.

Configure a sliding window at the server level

The following Apache configuration sets `HLSSlidingWindowLength` to `450`. This configuraiton creates a 1 hour sliding window for all HLS live events on the server:

```
...
<IfModule hlshttp_module>
<Location /hls-live>
    HLSHttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications"
    HttpStreamingContentPath "../applications"
    HLSMediaFileDuration 8000
    HLSSlidingWindowLength 450
    HLSFmsDirPath ".."
    HLSM3U8MaxAge 2
    HLSTSSegmentMaxAge -1
    Options -Indexes FollowSymLinks
</Location>
```

Restart Apache HTTP Server.

Publish streams with a sliding window

You don't need to configure any encoder settings to publish a stream with a sliding window.

Play streams with a sliding window

Devices that support HTTP Live Streaming support the sliding window feature by default.

Duplicate the livepkgr service

The server supports an unlimited number of instances of the livepkgr service.

❖ Duplicate the *rootinstall*/applications/livepkgr folder in the applications folder and give it a new name, for example, livepkgr2. In this case, the new livepkgr service is located at *rootinstall*/applications/livepkgr2.

You can create as many instances of the livepkgr service as you need.

Modify server-side code in the livepkgr service

Note: *You cannot modify server-side code on Adobe Media Server Standard.*

❖ Remove the *rootinstall*/applications/livepkgr/main.far file and replace it with the *rootinstall*/samples/applications/livepkgr/main.asc file.

Removing all HDS segments

To remove all the existing HDS segments when the application unloads, you can use the `clearOnAppStop` tag as shown below:

```
<ScriptEngine>
    <ApplicationObject>
        <config>
            <clearOnAppStop>true</clearOnAppStop>
        </config>
    </ApplicationObject>
</ScriptEngine>
```

Stream live media (RTMP)

Tutorial: stream live media (RTMP)

To complete this tutorial, install the following software:

- Flash Media Live Encoder

  Flash Media Live Encoder captures live audio and video, encodes it, and streams it to Adobe Media Server. Flash Media Live Encoder is free so it's a good idea to download the latest version.

- Adobe Media Server

  You can use any edition of Adobe Media Server to stream live media, including the free developer edition. For information about installing the server, see Installing the server.

- Flash Player

  This tutorial uses a video player that requires Flash Player 10.

For more information, see "Supported clients, encoders, codecs, and file formats" on page 1 and "Pre-built media players" on page 3.

Publish a live stream to Adobe Media Server

1 Connect a camera to the computer.

2 Open Flash Media Live Encoder and do the following in the Encoding Options panel:

   a From the Preset menu, choose High Bandwidth (300 Kbps) - H.264.

   You can choose any of the single stream options (not Multi Bitrate) from the Preset menu. The information on the left side of the panel is filled in when you choose a preset.

   b Select Stream to Adobe Media Server.

   c For AMS URL, enter **rtmp://localhost/live**.

   Use *localhost* for testing when Flash Media Live Encoder and Adobe Media Server are on the same computer. In a production environment, use the domain name or IP address of the computer hosting Adobe Media Server, for example, rtmp://ams.mycompany.com/live.

   If you're using a Adobe Media Server hosting provider, they can give you the domain name or IP address of the server.

   You must use the name *live* in the AMS URL unless you duplicate and rename the live service. The live service is a pre-built application on Adobe Media Server installed to *rootinstall*/applications/live. For more information, see "Duplicate the live service" on page 18.

   d For Stream, enter **livestream**.

   e To save a recording of the stream on your hard drive, select Save to File, click Browse, and choose a location. If you chose an H.264 preset, use an .f4v filename extension. If you chose a VP6 preset, use an .flv filename extension.

   Note that the server is not recording the file; Flash Media Live Encoder is recording the file. To serve the file on-demand when the live event is over, copy the file to the local disk of the server. For more information, see "Stream on-demand media (RTMP)" on page 24.

   To play an F4V file recorded by Flash Media Live Encoder without streaming it from the server, use the F4V Flattener tool to flatten the file. You can play FLV files without flattening them.

f Click Start to connect to the server and start streaming.

Use the Adobe Media Server sample player to play a live stream

1 Double-click the *rootinstall*/samples/videoPlayer/videoplayer.html file to open the sample video player in a browser.

***Note:** Substitute the Adobe Media Server installation directory for rootinstall.*

2 In the sample video player, do one of the following:

- In the list of videos, click “livestream”.
- Enter **rtmp://localhost/live/livestream**, check the LIVE checkbox, and click PLAY STREAM.

  If the media player isn’t on the same computer as Adobe Media Server, replace *localhost* with the domain name or IP address of the computer hosting Adobe Media Server.

  Use this URL to play this stream from any compatible video player, including Flash Media Playback and Strobe Media Playback. For more information about the sample video player and other video players, see “Pre-built media players” on page 3.

Use Flash Media Playback to play a live stream

1 Load the Flash Media Playback Setup page in a web browser: www.osmf.org/configurator/fmp/.

2 Enter the Video Source:

rtmp://localhost/live/livestream

You can replace *localhost* with the domain name or IP address of the server.

3 Click Preview to update the embed code.

4 Click Play to test the code.

5 To use the player in your own HTML page, copy the embed code and paste it into your page. Flash Media Playback is a compiled SWF file hosted by Adobe.

More Help topics

“Troubleshoot issues with streaming media” on page 111

“Stream live media (HTTP)” on page 5

URLs for publishing and playing live streams over RTMP

Use the following values to publish a single live stream to Adobe Media Server for streaming over RTMP:

AMS URL: **rtmp://*ams-ip-or-dns*/live**

Stream: ***streamname***

Use the following URL to play a live stream:

rtmp://*ams-ip-or-dns*/live/*streamname*

Duplicate the live service

You can create as many instances of the live service as you need.

1 Create a folder in the *rootinstall*/applications folder, for example, rootinstall/applications/live2

2 Copy the main.far, Application.xml, allowedHTMLdomains.txt, and allowedSWFdomains.txt files from the *rootinstall*/applications/live folder to the live2 folder.

3 Open the ams.ini file (located in *rootinstall*/conf) and add a parameter to set the content path for the new service, for example:

```
LIVE2_DIR = C:\Program Files\Adobe\Adobe Media Server 5\applications\live2
```

4 Open the Application.xml file in the *rootinstall*/applications/live2 folder and edit the virtual directory to the following:

```
<Streams>/;${LIVE2_DIR}</Streams>
```

5 Restart the server.

6 Clients can connect to the new publishing point at the URL:

```
rtmp://ams-ip-or-dns/live2
```

Modify server-side code in the live service

***Note:** You cannot modify server-side code in the live service on Adobe Media Server Standard.*

❖ Remove the *rootinstall*/applications/live/main.far file and replace it with the *rootinstall*/samples/applications/live/main.asc file.

Disable live services

❖ Move any live services folders out of the applications folder.

Stream on-demand media (HTTP)

Prerequisites for streaming on-demand media (HTTP)

To complete these tutorials, use the following software:

- Adobe Media Server 5. See the video Install Adobe Media Server 5 and verify HTTP streaming to Flash and iOS.
- (Adobe HTTP Dynamic Streaming) Adobe Media Server sample video player (which uses Strobe Media Playback) and Flash Player 10.1
- (Apple HTTP Live Streaming) iOS 3.0 or later device or Mac OS 10.6 with Safari 4.0 or QuickTime X

For more information, see "Supported clients, encoders, codecs, and file formats" on page 1 and "Pre-built media players" on page 3.

Play a single on-demand media file over HTTP

A community member has created a screencast that walks you through similar steps: Stream on-demand video to Flash and iOS over HTTP.

1 Install Adobe Media Server 5 and choose to install Apache HTTP Server.

2 Copy an F4V/MP4 file to the following location:

***rootinstall*/webroot/vod**

***Note:** Replace rootinstall with the Adobe Media Server installation folder.*

This tutorial uses the file sample2_1000kbps.f4v which installs with Adobe Media Server to the *rootinstall*/webroot/vod folder.

3 (Adobe HTTP Dynamic Streaming) To play the media in Flash Media Playback, do the following:

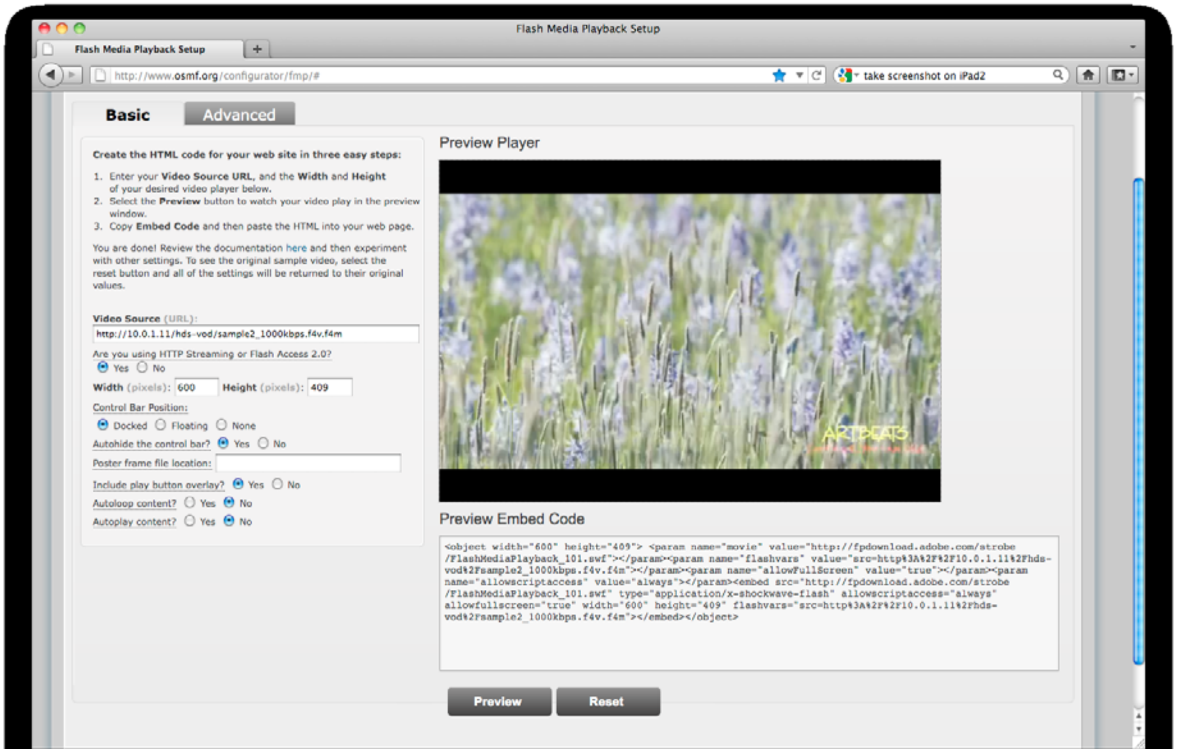
a Open Flash Media Playback in a web browser:

http://www.osmf.org/configurator/fmp/

Note: *Flash Media Playback requires Flash Player 10.1 to support HTTP Dynamic Streaming. To see which version of Flash Player is installed on your computer, go to Adobe Flash Player.*

b In Video Source, enter:

http://localhost/hds-vod/sample2_1000kbps.f4v.f4m


Flash Media Playback Setup
Flash Media Playback Setup
http://www.osmf.org/configurator/fmp/#
take screenshot on iPad2
Basic
Advanced
Preview Player

Create the HTML code for your web site in three easy steps:

1. Enter your **Video Source URL**, and the **Width** and **Height** of your desired video player below.
2. Select the **Preview** button to watch your video play in the preview window.
3. Copy **Embed Code** and then paste the HTML into your web page.

You are done! Review the documentation here and then experiment with other settings. To see the original sample video, select the reset button and all of the settings will be returned to their original values.

Video Source (URL): http://10.0.1.11/hds-vod/sample2_1000kbps.f4v.f4m

Are you using HTTP Streaming or Flash Access 2.0? ● Yes ○ No

Width (pixels): 600 **Height** (pixels): 409

Control Bar Position: ● Docked ○ Floating ○ None

Autohide the control bar? ● Yes ○ No

Poster frame file location:

Include play button overlay? ● Yes ○ No

Autoloop content? ○ Yes ● No

Autoplay content? ○ Yes ● No

Preview Embed Code

```
<object width="600" height="409"> <param name="movie" value="http://fpdownload.adobe.com/strobe
/FlashMediaPlayback_101.swf"></param><param name="flashvars" value="src=http%3A%2F%2F10.0.1.11%2Fhds-
vod%2Fsample2_1000kbps.f4v.f4m"></param><param name="allowFullScreen" value="true"></param><param
name="allowscriptaccess" value="always"></param><embed src="http://fpdownload.adobe.com/strobe
/FlashMediaPlayback_101.swf" type="application/x-shockwave-flash" allowscriptaccess="always"
allowfullscreen="true" width="600" height="409" flashvars="src=http%3A%2F%2F10.0.1.11%2Fhds-
vod%2Fsample2_1000kbps.f4v.f4m"></embed></object>
```

HDS on-demand streaming in Flash Media Playback

Although the media file lives in the /webroot/vod directory, the URL request is /hds-vod. The path /hds-vod is a `<Location>` directive in the Apache httpd.conf file. When a request URL begins with /hds-vod, the server looks for media in the /webroot/vod folder and packages it for HTTP Dynamic Streaming. For information about the request URL syntax, see "URLs for playing on-demand streams over HTTP" on page 23.

c Select Yes under the question "Are you using HTTP Streaming or Adobe Access 2.0?".

d Clear the text from the Poster frame file location text box.

e Click Preview to update the embed code.

f Click the Play button to test the code.

g To use the player in your own HTML page, copy the embed code and paste it into the body of the page. Flash Media Playback is a compiled SWF file hosted by Adobe. For more information, see Flash Media Playback.

4 (Adobe HTTP Dynamic Streaming) To play the media in Strobe Media Playback, do the following:

   a Open the Adobe Media Server sample video player in a web browser. Double-click *rootinstall*/samples/videoPlayer/videoplayer.html.

   b In STREAM URL, enter:

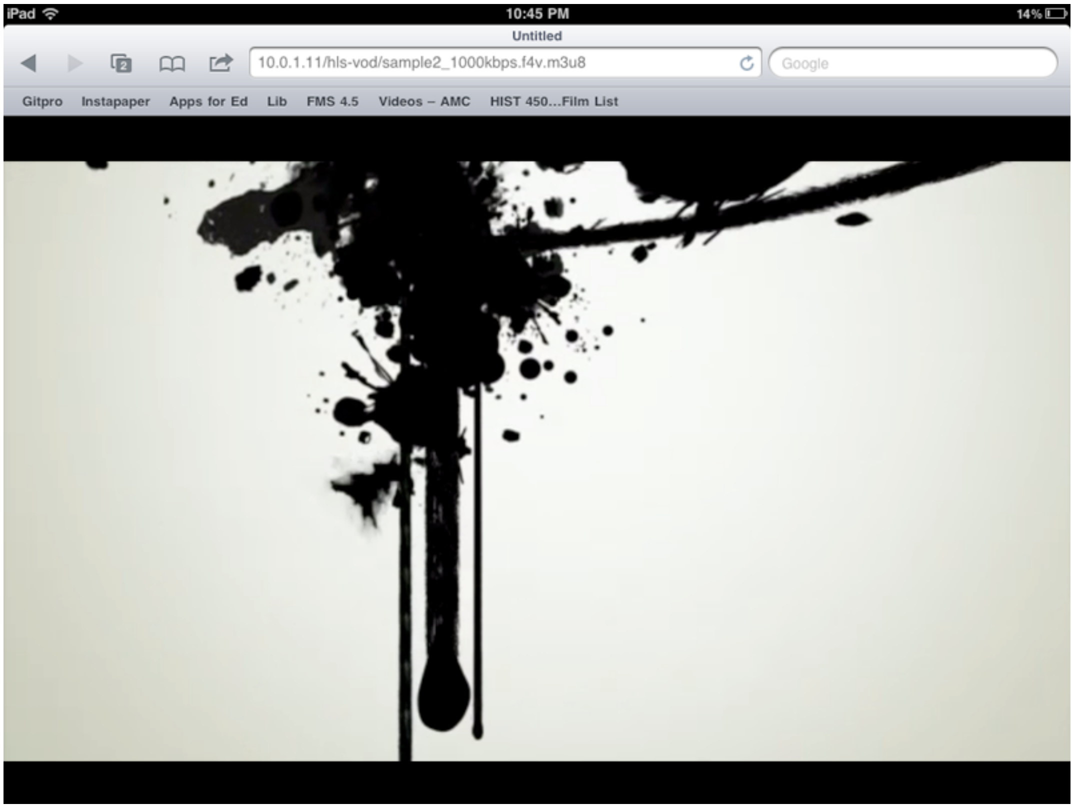
   http://localhost/hds-vod/sample2_1000kbps.f4v.f4m

   c Select VOD and click PLAY STREAM.

   d To use the player in your own HTML page, copy the embed code and paste it into the body of the page.

5 (Apple HTTP Live Streaming) On iOS, enter the following URL in the Safari address bar:

   http://10.0.1.11/hls-vod/sample2_1000kbps.f4v.m3u8



HLS on-demand streaming in Safari on an iPad

To test in Safari on Mac OS 10.6, use the following HTML code:

```
<video src="http://10.0.1.11/hls-vod/sample2_1000kbps.f4v.m3u8"
controls="controls"></video>
```

***Note:** You can replace "localhost" and "10.0.1.11" with the domain name or IP address of the computer hosting Adobe Media Server. If you're using a Adobe Media Server hosting provider, they can give you this value.*

More Help topics

"Troubleshoot issues with streaming media" on page 111

"Supported file formats and codecs" on page 2

"Content storage (HDS and HLS)" on page 56

"Configure HTTP Dynamic Streaming and HTTP Live Streaming" on page 51

"Stream on-demand media (RTMP)" on page 24

Play on-demand multi-bitrate media files over HTTP

A community member, Jody Bleyle has created a screencast that walks you through similar steps: HTTP adaptive bitrate streaming to Flash and iOS.

1 Install Adobe Media Server 5 and choose to install Apache HTTP Server.

2 Do one of the following:

- Encode an F4V/MP4 file at 3 different bitrates.
- Use the multi-bitrate sample files that install with Adobe Media Server to the ***rootinstall*/webroot/vod** folder.

This tutorial uses the following files installed to the webroot/vod folder:

***rootinstall*/webroot/vod/sample1_150kbps.f4v**

***rootinstall*/webroot/vod/sample1_700kbps.f4v**

***rootinstall*/webroot/vod/sample1_1500kbps.f4v**

3 (Adobe HTTP Dyanmic Streaming) To stream multi-bitrate content, Flash and AIR media players request a *set-level manifest file*. This is a .f4m file that contains the location and bitrate of each stream. To create a set-level F4M file, do the following:

a Open *rootinstall*/Adobe Media Server/tools/f4mconfig/configurator/f4mconfig.html in a browser.

b Select the f4m file type.

c Stream URIs can be absolute or relative to a Base URI. Enter the following for the Base URI:

http://localhost/hds-vod/

d Enter the following for each stream and click Add:

Stream URI	Bitrate
sample1_150kbps.f4v.f4m	150
sample1_700kbps.f4v.f4m	700
sample1_1500kbps.f4v.f4m	1500

e To view the file, click View Manifest. The manifest file looks like this:

```
<manifest xmlns="http://ns.adobe.com/f4m/2.0">
  <media href="http://localhost/hds-vod/sample1_150kbps.f4v.f4m" bitrate="150"/>
  <media href="http://localhost/hds-vod/sample1_700kbps.f4v.f4m" bitrate="700"/>
  <media href="http://localhost/hds-vod/sample1_1500kbps.f4v.f4m" bitrate="1500"/>
</manifest>
```

f Click Save Manifest to save the file as sample1.f4m to *rootinstall*/webroot.

The media player requests this file from a web server. This tutorial saves the file to *rootinstall*/webroot, but the file can be served from any location on any webserver. This file does not need to live on Adobe Media Server. This file can have any name.

4 (Apple HTTP Live Streaming)For multi-bitrate streaming, iOS devices request a set-level M3U8 variant playlist file that contains the location, bitrate, and optionally the codec of each stream. To create a set-level M3U8 file, do the following:

a If the File Generator tool isn't open, double-click *rootinstall*/Adobe Media Server/tools/f4mconfig/configurator/f4mconfig.html to open it in a browser.

b Select the m3u8 file type.

c Enter the Stream URI, bitrate, program-ID, and optionally a resolution and codec for each stream. Stream URIs can be absolute or relative. If they are relative, they are relative to the m3u8 file. The program-ID must be the same for each stream.

***Note:** For information about supported codecs, see Apple HTTP Live Streaming Overview.*

This tutorial uses the following Stream URI settings:

Stream URI	Bitrate
http://10.0.1.11/hls-vod/sample1_150kbps.f4v	150
http://10.0.1.11/hls-vod/sample1_700kbps.f4v	700
http://10.0.1.11/hls-vod/sample1_1500kbps.f4v	1500

d To view the file, click View Manifest. The m3u8 file looks like this:

```
#EXTM3U
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=150000
http://10.0.1.11/hls-vod/sample1_150kbps.f4v.m3u8
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=700000
http://10.0.1.11/hls-vod/sample1_700kbps.f4v.m3u8
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=1500000
http://10.0.1.11/hls-vod/sample1_1500kbps.f4v.m3u8
```

e Click Save Manifest and save the file as sample1.m3u8 to the folder *rootinstall*/webroot.

The media player requests this file from a web server. This tutorial saves the file to *rootinstall*/webroot, but the file can be served from any location on any webserver. This file does not need to live on Adobe Media Server.

5 (Adobe HTTP Dynamic Streaming) Flash Media Playback does not support set-level manifest files. To play the media, use Strobe Media Playback. Strobe Media Playback is used in the sample video player that installs with Adobe Media Server.

a Double-click *rootinstall*/samples/videoPlayer/videoplayer.html to open it in a browser.

b In Video Source, enter the following:

http://localhost/sample1.f4m

6 (Apple HTTP Live Streaming) On iOS, enter the following URL in the Safari address bar:

http://10.0.1.11/sample1.m3u8

To test in Safari on Mac OS 10.6, use the following HTML code:

```
<video src="http://10.0.1.11/sample1.m3u8" controls="controls"></video>
```

For more information, see HTTP Live Streaming Overview in the iOS Reference Library.

URLs for playing on-demand streams over HTTP

Use the following URLs to play on-demand streams over HTTP:

Client	**Live or VOD**	**Single stream or Adaptive Bitrate**	**Request URL**
Adobe HTTP Dynamic Streaming	VOD	Single stream	http://<ams-dns-or-ip>/**hds-vod**/<streamname>.<fileextension>.**f4m**
Apple HTTP Live Streaming	VOD	Single stream	http://<ams-dns-or-ip>/**hls-vod**/<streamname>.<fileextension>.**m3u8**
Adobe HTTP Dynamic Streaming	VOD	Adaptive bitrate	The media player requests a set-level manifest file: http://<webserver-dns-or-ip>/<set-level-manifest>.**f4m** The URLs in the set-level manifest point to the individual streams and use the following URL: http://<ams-dns-or-ip>/**hds-vod**/<streamname>.<fileextension>.**f4m**
Apple HTTP Live Streaming	VOD	Adaptive bitrate	The media player requests a set-level variant playlist file: http://<webserver-dns-or-ip>/<set-level-variant-playlist>.**m3u8** The URLs in the set-level variant playlist point to the individual streams and use the following URL: http://<ams-dns-or-ip>/**hls-vod**/<streamname>.<fileextension>.**m3u8**

The path /hds-vod is a `<Location>` directive in the Apache httpd.conf file. The directive tells the server to look for the content in the *rootinstall*/webroot/vod folder. It also tells the server to package the content for delivery to Flash and AIR over HTTP (called HTTP Dynamic Streaming or *HDS*).

The path /hls-vod is a `<Location>` directive in the Apache httpd.conf file. The directive tells the server to look for the content in the *rootinstall*/webroot/vod folder. It also tells the server to package the content for delivery to Apple HTTP Live Streaming.

***Note:** Apple HTTP Live Streaming supports live and on-demand streaming.*

For adaptive bitrate streaming, the client requests a set-level manifest file. For HDS, this file is an .f4m file. For HLS, this file is a .m3u8 file. Set-level manifest files contain the paths to the physical locations of the media files. The paths to the media files must begin with /hds-vod or /hls-vod to tell the server to package them for HTTP.

More information

"Content storage (HDS and HLS)" on page 56

Stream on-demand media (RTMP)

Tutorial: Stream on-demand media (RTMP)

Prerequisites

To complete this tutorial, install the following software:

- Adobe Media Server

  You can use any edition of Adobe Media Server to stream on-demand media, including the free developer edition. For information about installing the server, see Installing the server.

- Flash Player

  This tutorial uses a video player that requires Flash Player 10.

  To see which version of Flash Player is installed in a browser, go to www.adobe.com/software/flash/about/.

Copy on-demand files to Adobe Media Server

Use the vod (video on demand) service on Adobe Media Server to stream recorded media to clients. Simply copy recorded media files to the server and clients can stream them. The vod service streams files to media players from the following two folders:

***rootinstall*/applications/vod/media**

***rootinstall*/webroot/vod**

You can also create subfolders of these folders to hold media files. This tutorial creates a subfolder and copies a file to the subfolder to stream.

1 Browse to the following folder:

  ***rootinstall*/applications/vod/media**

  Note: *Replace rootinstall with the Adobe Media Server installation folder. For example, on Windows the default installation folder is C:\Program Files\Adobe\Adobe Media Server 5.*

2 Create the folder "tests" in the "media" folder, as follows:

  ***rootinstall*/applications/vod/media/tests**

3 Copy an F4V/MP4 or FLV file to the /tests folder.

  This tutorial uses the following file:

  ***rootinstall*/applications/vod/media/tests/polymorphics.f4v**

  If you don't have a video file, you can download a file from the Moving Images Archive at www.archive.org/details/movies. You can also use a sample file included with the server. For learning purposes, copy a sample file from *rootinstall*/applications/vod/media to *rootinstall*/applications/vod/media/tests.

  For information about supported file formats and codecs, see Supported file formats.

Use the Adobe Media Server sample player to play an on-demand file

1 To open the sample video player in a browser, double-click the *rootinstall*/samples/videoPlayer/videoplayer.html file.

2 Enter the address of the video in the STREAM URL textbox, check VOD, and click PLAY STREAM. This tutorial plays the following file:

  rtmp://localhost/vod/mp4:tests/polymorphics.f4v

  If you're using a Adobe Media Server hosting provider, replace *localhost* with the domain name or IP address of the server that they provided.

  Important: *Although you copy the media file to the vod/media folder, you do not need to specify /media in the path when you play the file. The server is configured by default to look for media files in the vod/media folder. If you specify /media in the path, the server looks in the /vod/media/media folder.*

  Begin paths to F4V/MP4 files with the prefix `mp4:`. Use the filename extension of the file, whether it's F4V, MP4, MOV, and so on.

Use Flash Media Playback to play an on-demand stream

1 Load the Flash Media Playback Setup page in a web browser: www.osmf.org/configurator/fmp/.

2 Enter the Video Source:

rtmp://localhost/vod/mp4:tests/polymorphics.f4v

You can replace *localhost* with the domain name or IP address of the server.

3 Click Preview to update the embed code.

4 Click the Play button to test the code.

5 To use the player in your own HTML page, copy the embed code and paste it into the body of the page. Flash Media Playback is a compiled SWF file hosted by Adobe. For more information, see "Play media in Flash Media Playback" on page 4.

More Help topics

"Supported file formats and codecs" on page 2

"Troubleshoot issues with streaming media" on page 111

"Stream on-demand media (HTTP)" on page 19

URLs for playing on-demand media files over RTMP

After completing the tutorial once, learn more about how to build a URL that requests a file from the server. Run through the tutorial again using different files and different paths.

The syntax for requesting a file from Adobe Media Server is as follows:

```
protocol://server-domain-or-IP/ams-app-name/[ams-app-instance-name/][codec-prefix:]file-path[filename-extension]
```

Element	**Required**	**Description**
protocol	Yes	The protocol for media delivery. For information about the protocols Adobe Media Server supports, see RTMP, RTMFP, and HTTP in the *Technical Overview*.
server-domain-or-IP	Yes	The domain name or IP address of the computer hosting Adobe Media Server. If the client is on the same computer as Adobe Media Server, you can use *localhost* for testing purposes.
ams-app-name	Yes	The Adobe Media Server application that the client connects to. The default folder that holds Adobe Media Server applications is *rootinstall*/applications. This tutorial uses the application *rootinstall*/applications/vod.
ams-app-instance-name	No	Applications can have an unlimited number of instances. For example, you could have clients connect to rtmp://localhost/vod/instance1, rtmp://localhost/vod/instance2, and so on.
codec-prefix	Required for F4V/MP4 and MP3 files	The F4V/MP4 and MP3 file formats require a codec prefix in the request URL. For F4V/MP4 files, use the prefix `mp4:`. For MP3 files, use the prefix `mp3:`. FLV files do not require a codec prefix.

Element	Required	Description
file-path	Yes	The path from the folder configured to hold media files to the media file. For the vod application, the following folders are configured to hold media files: *rootinstall*/applications/vod/media *rootinstall*/webroot/vod In the tutorial, the request URL is: rtmp://localhost/vod/mp4:tests/polymorphics.f4v. The file-path is tests/polymorphics. If the file polymorphics.f4v were in the /vod/media folder instead of in the /vod/media/tests folder, the full request URL would be rtmp://localhost/vod/mp4:polymorphics.f4v. The file-path would be polymorphics. Configure the folders to hold media files in the *rootinstall*/conf/ams.ini file.
filename-extension	Required for F4V/MP4 and MP3 files.	Use the filename extension of the file you want to play. For example, if the file is an F4V file, use .f4v. If the file is an MOV file, use .mov.

Use the following addresses to stream on-demand media files over RTMP:

File format	Address
F4V	rtmp://*server-domain-or-IP*/vod/mp4:*filename* rtmp://*server-domain-or-IP*/vod/mp4:*filename*.f4v rtmp://*server-domain-or-IP*/vod/mp4:*subfolder/fileName*.f4v
FLV	rtmp://*server-domain-or-IP*/vod/*filename* rtmp://*server-domain-or-IP*/vod/*filename*.flv

To download a file progressively over HTTP from the Apache web server installed with Adobe Media Server, use a standard HTTP address. The following file is in the *rootinstall*/webroot/vod folder:

http://*server-domain-or-IP*/vod/*filename.xxx*

***Note:** To use localhost over HTTP, append the port number 8134, for example, http://localhost:8134/vod/video.f4v. The server uses port 8134 internally for HTTP.*

Configure the location of media files

Two parameters in the *rootinstall*/conf/ams.ini file determine the locations of the folders in which the vod application looks for media files:

```
VOD_COMMON_DIR = C:\Program Files\Adobe\Adobe Media Server 5\webroot\vod
VOD_DIR = C:\Program Files\Adobe\Adobe Media Server 5\applications\vod\media
```

When a client connects to the vod application and plays a file, the server looks for the file in these two folders.

Files in the folder specified in the `VOD_DIR` parameter can stream over RTMP only.

Files in the folder specified in the `VOD_COMMON_DIR` parameter can stream over RTMP and can download progressively over HTTP. Write code in the client that checks for a successful NetConnection to the server over RTMP. If the client doesn't connect successfully, write code that requests the file over HTTP. The server does not automatically fall back to HTTP.

Important: *Although you copy media to the vod/media folder, do not specify /media in the path when you play the file. The server is configured by default to look for media in the vod/media folder. If you specify /media in the path, the server looks in the /vod/media/media folder.*

Change the storage location of media files

1 Open the *rootinstall*/conf/ams.ini file in a text editor.

2 Edit the `VOD_DIR` and `VOD_COMMON_DIR` parameters.

3 Restart Adobe Media Server.

Add new media storage locations

1 Open the *rootinstall*/conf/ams.ini file in a text editor.

2 Add a new parameter and point it to the desired location, for example:

```
VOD_DIR_2 = C:\hrvideos
```

3 Open the *rootinstall*/applications/vod/Application.xml file in a text editor.

The parameters in the ams.ini file are used in the Application.xml configuration file.

```
<Application>
    <StreamManager>
        <VirtualDirectory>
            <Streams>/;${VOD_COMMON_DIR}</Streams>
            <Streams>/;${VOD_DIR}</Streams>
        </VirtualDirectory>
    </StreamManager>
```

4 Add a `<Streams>` tag with the new parameter, for example:

```
<Streams>/hr;${VOD_DIR_2}</Streams>
```

To play streams stored in this folder, use the following address:

`rtmp://localhost/vod/`**`hr`**`/mp4:`*`somefilename`*`.f4v`

The `<Streams>` tag tells the server to look for media in the location specified in the `VOD_DIR_2` parameter if the media path starts with /hr.

5 Restart Adobe Media Server.

Duplicate the vod service

The server supports an unlimited number of instances of the vod service.

1 Duplicate the *rootinstall*/applications/vod folder in the applications folder and give it a new name, for example, vod2. In this case, the new vod service is located at *rootinstall*/applications/vod2.

You can create as many instances of the vod service as you need.

2 Clients can connect to the vod service at the URL `rtmp://`*`adobemediaserver`*`/vod2`.

3 Open the ams.ini file (located in *rootinstall*/conf) and do the following:

- Add a parameter to set the content path for the new service, for example: `VOD2_DIR = C:\Program Files\Adobe\Adobe Media Server 5\applications\vod2\media`.
- If you installed Apache and want the media files to be available over HTTP, add a new `VOD2_COMMON_DIR` parameter: `VOD2_COMMON_DIR = C:\Program Files\Adobe\Adobe Media Server 5\webroot\vod2`.

4 Open the Application.xml file in the *rootinstall*/applications/vod2 folder and do the following:

- Edit the virtual directory to the following: `<Streams>/;${VOD2_DIR}</Streams>`.
- Edit the virtual directory to the following: `<Streams>/;${VOD2_COMMON_DIR}</Streams>`.

5 Place recorded media files into the following locations:

- Place files that stream only over RTMP in the C:\Program Files\Adobe\Adobe Media Server 5\applications\vod\media folder.
- Place files that stream over RTMP or HTTP in the C:\Program Files\Adobe\Adobe Media Server 5\webroot\vod2.

***Note:** You do not have to specify the media folder in the URL; the media folder is specified in the path you set in the ams.ini file.*

Modify server-side code in the vod service

***Note:** You cannot modify server-side code in the live service on Adobe Media Server Standard.*

❖ Remove the *rootinstall*/applications/vod/main.far file and replace it with the *rootinstall*/samples/applications/vod/main.asc file.

Disable the vod service

❖ Move any vod service folders out of the *rootinstall*/applications folder.

Stream on-demand encrypted media (pRTMP)

Use protected RTMP (pRTMP) to encrypt and deliver on-demand content to Flash Player and AIR.

***Note:** Protected RTMP isn't a protocol. It delivers encrypted content over the RTMP protocol.*

System requirements

Protected RTMP is supported in origin-only configurations. Unlike RTMPE, pRTMP does not work in an edge-origin configuration.

Applications that share media files *must* use the same protected RTMP configuration settings for the shared files.

Flash Player 11.0 and AIR 3.0 are required to play protected RTMP content. Protected RTMP supports all the on-demand file formats and codecs that RTMP supports except F4M, RAW, and F4F.

***Note:** In Adobe Media Server 5, protected RTMP is supported for VOD streaming, it is not supported for live streaming.*

About protected RTMP

Adobe Media Server encrypts on-demand media files and embeds a Adobe Access 3.0 license in the DRM metadata of the content. Flash Player and AIR clients communicate with Adobe Media Server to play the media. Protected RTMP does not require a license server; the license is embedded in the content metadata and a client receives it along with the media.

Protected RTMP is more secure than RTMPE because it uses Adobe Access 3.0 DRM content protection. Protected RTMP encrypts the content whereas RTMPE protects the communication channel.

Because the content is encrypted, unauthorized replay is impossible. Protected RTMP content can be decrypted only by Flash Player and AIR clients that have the Protected Streaming private key.

There is no way to use an RTMPE connection between servers and decrypt the content on Adobe Media Server. Different applications that access common media files must have consistent setup with respect to Protected RTMP. That is, you should not access the same media file over both protected and unprotected RTMP.

Separate applications that include common media files must have consistent settings with respect to Protected RTMP. When AMS receives requests for media file playback (VOD) it applies the application-specified protection to the media and then caches it in memory. Each time the server receives a new request for an already cached media file it compares the protection settings of the cached media with the settings of the application in the new request. If those settings are inconsistent AMS replaces the cached media with content processed according to the settings of the application of the current request and then sends the media to the client. AMS generates an error log entry every time it identifies inconsistency between cached media settings and requested media settings.

Protected RTMP simplifies deployment and increases security because:

- Adobe Media Server is the only server required.
- No license server is required.
- No domain server is required.
- All media can be packaged with the same common key.
- Content is encrypted with Adobe Access 3.0 DRM protection.
- Protected RTMP uses 128-bit AES encryption. RTMPE uses 128-bit RC4.

Configure protected RTMP at the vhost, application, or application instance level. Use the Application.xml file, Server-Side ActionScript, and the Authorization Plug-in to configure protected RTMP.

***Note:** The DRM used for Protected RTMP is the same DRM used for Protected HTTP streaming..*

Quick start: Use protected RTMP to play on-demand media

To complete this tutorial, install the following software:

- Adobe Media Server 5

  For information about installing the server, see Installing the server.

- Flash Player 11

  This tutorial requires Flash Player 11.

Configure protected RTMP on Adobe Media Server

1 Configure protected RTMP for the VOD application.

2 Open *amsrootinstall*/applications/vod/Application.xml in a text editor.

3 Add the `<ProtectedRTMP enabled="true"></ProtectedRTMP>` element. The Application.xml file looks like this:

```
<Application>
    <ProtectedRTMP enabled="true"></ProtectedRTMP>
    ...
</Application>
```

4 Save and close the file.

5 If the VOD application is running, close it to reload the Application.xml file. If the Adobe Media Server Start Screen is open, the VOD application is running. Closing the Start Screen closes the VOD application (unless another client is also connected to the VOD application).

Play on-demand media

1 Double-click the Adobe Media Server Sample Video Player to open it in a browser: *amsrootinstall*/samples/videoPlayer/videoplayer.html.

2 Click any link in the /applications/vod/media list. Every file that the VOD application streams uses protected RTMP.

Configure protected RTMP

You can configure protected RTMP in the following locations:

- Application.xml file:
    - Configure the Application.xml file in an application folder to configure pRTMP for a single application.
    - Configure the Application.xml file in a vhost folder to configure pRTMP for all applications in a virtual host.
- Server-Side ActionScript:
    - Set properties on the application object.
- Authorization plug-in:
    - Set fields in the in the `E_APPSTART` event of the Authorization plug-in.

Use the Application.xml file to configure protected RTMP

To configure protected RTMP, add a `<ProtectedRTMP>` section to the Application.xml file at the application level or at the vhost level and set the enabled attribute to `"true"`. The other pRTMP parameters have default values. You can use the default values or set the parameters to new values. The following is a sample `<ProtectedRTMP>` section:

```
<Application>
    <ProtectedRTMP enabled="true">
        <CommonKeyFile>creds/common-key.bin</CommonKeyFile>
        <UpdateInterval>60</UpdateInterval>
        <SWFVerification enabled="true">
            <WhiteListFolder>whitelists</WhiteListFolder>
        </SWFVerification>
    </ProtectedRTMP>
</Application>
```

Use the following parameters in an Application.xml file to configure protected RTMP:

Parameter	Description	Default
`CommonKeyFile`	A relative path to a file containing a base key. The path is relative to the folder that contains the Application.xml file. The server generates the base key file during installation. The server uses the base key (along with the content ID) to generate the final content encryption key. The key data is scrambled and Base64 encoded using the scramble tool. The server generates the content ID automatically.	*rootinstall*/creds/common-key.bin
`UpdateInterval`	Optional configuration that specifies how often (in minutes) the server updates the DRM metadata.	60
`SWFVerification`	Container for SWFVerification. To enable SWF verification, set the `enabled` attribute to `"true"`.	`"false"`
`WhiteListFolder`	The folder that contains the SWF whitelist for SWF verification. The folder can contain more than one whitelist file. Relative paths are relative to the application folder.	The folder containing the application folder.

Use the Authorization plug-in to configure protected RTMP

To configure protected RTMP at the plug-in level, write code in the `E_APPSTART` event to set the `F_APP_PRTMP` field to true. The other pRTMP fields have default values. You can use the default values or set the fields to new values. Protected RTMP settings in the Application.xml file are overridden in the Authorization plug-in.

The `E_APPSTART` event is an application-level event that supports reading all application-level fields, including the following:

Field	Description	Read/Write
F_APP_URI	The URI of the application to which the client connected. The value does not include the server name or port information.	Read-only
F_APP_NAME	The application name.	Read-only
F_APP_NAME	The application instance name.	Read-only

The `E_APPSTART` event supports the following new fields for protected RTMP:

Field	Description	Read/Write
`F_APP_PRTMP`	A Boolean value indicating whether RTMP streaming is protected.	Read/Write
`F_APP_PRTMP_COMMON_KEY_FILE`	A string to specify the path to the common key file. The path can be absolute or relative. Relative paths are relative to the location of the application folder.	Read/Write
`F_APP_PRTMP_UPDATE_INTERVAL`	A number to specify how often (in minutes) the DRM metadata will be updated.	Read/Write
`F_APP_PRTMP_SWF_VERIFICATION`	A Boolean value indicating whether SWF Verification is enabled.	Read/Write

Field	**Description**	**Read/Write**
`F_APP_PRTMP_SWF_WHITELIST_FOLDER`	A string to specify the path to the whitelist folder for SWF verification. The path can be absolute or relative. Relative paths are relative to the location of the application folder.	Read/Write
`F_APP_PRTMP_SWF_WHITELIST`	A string containing one or more SWF digests, separated by '\n'. SWF digests are sha256 hashes of the SWF, base64 encoded. You generate a whitelist using the whitelist tool that ships with AMS. This string can also contain the comments and empty lines that are in the whitelist generated by the tool, which are ignored.	Read/Write
`F_APP_PRTMP_AIR_WHITELIST`	A string containing one or more AIR identifiers, separated by '\n'. AIR identifiers are derived from the signature.xml file that is used to sign an AIR app. You generate a whitelist using the whitelist tool that ships with AMS. This string can also contain the comments and empty lines that are in the whitelist generated by the tool, which are ignored.	Read/Write

The `E_PRTMP_WHITELIST` event is an application-level event that AMS calls every *n* minutes (according to the update interval), enabling you to update the SWF/AIR whitelists through the `F_APP_PRTMP_SWF_WHITELIST` and `F_APP_PRTMP_AIR_WHITELIST` fields.

Use Server-side ActionScript to configure protected RTMP

Use the following Server-side ActionScript properties to configure protected RTMP:

Property	**Data type**
`application.prtmpEnabled`	Boolean
`application.prtmpCommonKeyFile`	String
`application.prtmpUpdateInterval`	Int
`application.prtmpSWFVEnabled`	Boolean
`application.prtmpSwfWhitelistFolder`	String

Note: *The* `prtmpCommonKeyFile` *and* `prtmpSwfWhitelistFolder` *properties specify a path to a file or folder. The server resolves relative paths to an application directory.*

These properties are filled in by values retrieved from configuration files during the `application.onAppStart()` event. You can also set values in this event, as follows:

```
application.onAppStart = function()
{
    trace  ("----------prtmp on start----------");
    trac  e("prtmp " + application.prtmpEnabled);
    trace("prtmpCommonKeyFile " + application.prtmpCommonKeyFile);
    trace("prtmpUpdateInterval " + application.prtmpUpdateInterval);
    trace("prtmpSwfVerification " + application.prtmpSWFVEnabled);
    trace("prtmpSwfWhitelist "   + application.prtmpSwfWhitelistFolder);
    trace  ("------------prtmp on start-------  -");
    application.prtmpEnabled = false;
    application.prtmpCommonKeyFile = "TEST  ";
    application.prtmpUpdateInterval = 12000  ;
    application.prtmpSWFVEnabled = true;
    application.prtmpSwfWhitelistFolder = "";
}
```

Trace the new values in the `application.onConnect()` event:

```
{
    trace("----------prtmp on connect----------");
    trace("prtmp " + application.prtmpEnabled);
    trace("prtmpCommonKeyFile " + application.prtmpCommonKeyFile);
    trace("prtmpUpdateInterval " + application.prtmpUpdateInterval);
    trace("prtmpSwfVerification " + application.prtmpSWFVEnabled);
    trace("prtmpSwfWhitelist " + application.prtmpSwfWhitelistFolder);
    trace("------------prtmp on connect--------");
}
```

The previous code prints an application.log file similar to the following:

```
#Version: 1.0
#Start-Date: 2011-10-19 12:20:41
#Software: Adobe Media Server 5 d430 x64
#Date: 2011-10-19
#Fields: date time x-pid x-status x-ctx x-comment
2011-10-19 12:20:41 7832 (s)2641173 ----------prtmp on start---------- -
2011-10-19 12:20:41 7832 (s)2641173 prtmp true -
2011-10-19 12:20:41 7832 (s)2641173 prtmpCommonKeyFile c:\Program Files\Adobe\Adobe Media
Server 5\conf\_defaultRoot_\_defaultVHost_\streamtest\creds\common-key.bin -
2011-10-19 12:20:41 7832 (s)2641173 prtmpUpdateInterval 3600000 -
2011-10-19 12:20:41 7832 (s)2641173 prtmpSwfVerification false -
2011-10-19 12:20:41 7832 (s)2641173 prtmpSwfWhitelist c:\Program Files\Adobe\Adobe Media
Server 5\conf\_defaultRoot_\_defaultVHost_\streamtest -
2011-10-19 12:20:41 7832 (s)2641173 ------------prtmp on start-------- -
2011-10-19 12:20:41 7832 (s)2641173 ----------prtmp on connect---------- -
2011-10-19 12:20:41 7832 (s)2641173 prtmp false -
2011-10-19 12:20:42 7832 (s)2641173 prtmpCommonKeyFile c: \Program Files\Adobe\Adobe Media
Server 5\conf\_defaultRoot_\_defaultVHost_\streamtest\TEST -
2011-10-19 12:20:42 7832 (s)2641173 prtmpUpdateInterval 12000 -
2011-10-19 12:20:42 7832 (s)2641173 prtmpSwfVerification true -
2011-10-19 12:20:42 7832 (s)2641173 prtmpSwfWhitelist c:\Program Files\Adobe\Adobe Media
Server 5\WhiteListFolder -
2011-10-19 12:20:42 7832 (s)2641173 ------------prtmp on connect-------- -
```

Tune server performance

The pRTMP operations that use the most CPU power are generating the DRM metadata and generating a license. Adobe Media Server generates DRM metadata the first time a client requests a media file. The server caches the metadata and reuses it for subsequent request for the same media file.

The media license in the DRM metadata imposes a 24-hour time limit on media playback from the point at which the license was generated. Therefore, the server refreshes the license periodically and generates new DRM metadata with a new time-stamp. Configure the `<UpdateInterval>` parameter to control how often the server generates DRM metadata. The default value is 60 minutes. To use less CPU power, generate the license less frequently.

Certificates and policy files

All certificates and the policy file for protected RTMP are installed to the Adobe Media Server rootinstall/creds folder.

Add Protected Streaming certificates

Adobe Media Server uses Protected Streaming certificates (.cer files) for protected RTMP. The server installs with three Protected Streaming certificates. It uses different certificates for desktop, mobile, and set-top.

In the event of a security breach, Adobe will release a security alert telling you to download new Protected Streaming certificates.

To add a Protected Streaming certificate, copy it to the *`rootinstall`*`/creds/sd` folder.

Understand static security files

The following files are installed with Adobe Media Server to the *`rootinstall`*`/creds/static` folder. Do not replace or modify these files.

Policy file

Adobe Media Server installs with the policy file rootinstall/creds/static/creds_24hr_policy.pol. The policy is set as:

```
anonymous; not use license chaining; 24 hours limited license caching; and Protected Streaming
is permitted.
```

This policy file lets clients start playing content within 24 hours from when the server generated the DRM metadata. Users can continue watching the content until the end, even if that time is beyond the 24 hour window. The 24 hour window starts when the server generates the DRM metadata and stores it in a cache. The `<UpdateInterval>` parameter in the Application.xml file determines the frequency at which the server generates DRM metadata. The default value is 60 minutes.

License server certificate

An Adobe-issued DER-encoded license server certificate. The license server certificate specifies the private key used to sign the license.

Transport certificate

An Adobe-issued DER-encoded X.509 transport certificate file. The transport certificate file is used when the client communicates with a server (for example, an authentication server). This feature is not supported in Adobe Media Server 5 but the certificate is still required.

Packager credential

An Adobe-issued packager server credential (a certificate and its associated key) PFX file. The server uses this file to apply a signature to the metadata while encrypting content files.

Multicast media (RTMFP)

About the multicast service

Flash Media Server 4.0, Flash Player 10.1

Important: *The Multicast service (rootinstall/applications/multicast) does not run on Adobe Media Server Standard.*

The multicast service is part of the Adobe Media Server multicast solution. The multicast solution is an end-to-end solution for live, corporate, multicast events. The multicast solution delivers a single, live stream to Flash Player clients over IP multicast, P2P multicast, and the combination of both (called *fusion multicast*).

The multicast solution consists of the following components:

- The multicast config tool (*rootinstall*/tools/multicast/configurator)

  Collects information to describe and configure a multicast event. Generates a live stream name to copy and paste into Flash Media Live Encoder that embeds required multicast event settings as parameters within the stream name's query string. Generates a manifest.f4m file that you copy to the same folder as the multicast player.

- Flash Media Live Encoder

  Captures, encodes, and publishes the live video to the multicast service on the Adobe Media Server.

- The multicast service (*rootinstall*/applications/multicast)

  A server-side Adobe Media Server application that republishes the live stream as a multicast stream into a target RTMFP Group.

- The multicast player (*rootinstall*/tools/multicast/multicastplayer)

  A client-side Flash Player application that connects to the multicast service, joins the group, and plays the live stream.

Configure the multicast event

Do the following to configure the multicast event:

1. Open *rootinstall*/tools/multicast/configurator/configurator.html in a browser.
2. Select one of the following multicast types:
   - Fusion (simultaneous, cooperative IP and P2P multicast)
   - IP Multicast
   - Peer to Peer
   - Peer to Peer with Peer Discovery

   Note: *The tool contains descriptions of each type.*
3. For Fusion and Peer to Peer, enter the server name (or IP address) and the full path to the multicast application, for example, **rtmfp://ams.example.com/multicast**. If the server is configured to use a port other than 1935, specify the port, for example, **rtmfp://ams.example.com:1940/multicast**.

   Note: *IP Multicast events do not require a connection URI. IP Multicast events use the URI* `rtmfp:` *which puts the NetConnection object in "connectionless" mode.*
4. Enter the name of the live stream, for example, **CorpAllHandsQ2_2010**, or **livestream**.
5. Enter a publish password.

The password ensures that only the multicast server can publish a multicast stream into the group. Other peers do not have the publish password and can only play the stream, not publish a stream.

6 Enter a name for the group. To make the group name unique, select Make Unique.

Use unique group names for streaming events occuring at the same time.

7 For IP Multicast and Fusion, enter an IP multicast address and port to use for the live event. This is the address and port to which the live stream is broadcast.

***Note:** Get the IP multicast address from your IT department. The IT department sets up the multicast router and multicast address. The multicast address delivers data to all members of the group on the network.*

8 (Optional) If the server hosting Adobe Media Server has more than one network interface card (NIC), enter the IP address for one NIC in the Interface Address text box. Adobe Media Server uses this IP address to determine the appropriate interface to use when publishing.

9 (Optional) To use source-specific multicast, enter an IP address and port in the Source-specific Multicast Address text box. For more information, see "Source-specific IP multicast" on page 266.

10 Click Generate. The multicast config tool generates the following:

- A name for the live stream. To publish a stream, you'll click Copy and paste this value into the Flash Media Live Encoder Stream field.
- An F4M file to use with the client multicast player. To view the F4M file, click View Manifest File.

This tutorial uses the following settings:

11 To use the Multicast Sample Player, do the following:

a Click Save Manifest File.

b Save the manifest.f4m file to the same location as the multicastplayer.html and multicastplayer.swf files. By default, this location is *rootinstall*/tools/multicast/multicastplayer.

12 To use Strobe Media Playback, leave the Configurator open. After you set up Strobe Media Playback, you will return to the Configurator and save the manifest.f4m file to the same folder as Strobe Media Playback.

Publish a stream from Flash Media Live Encoder

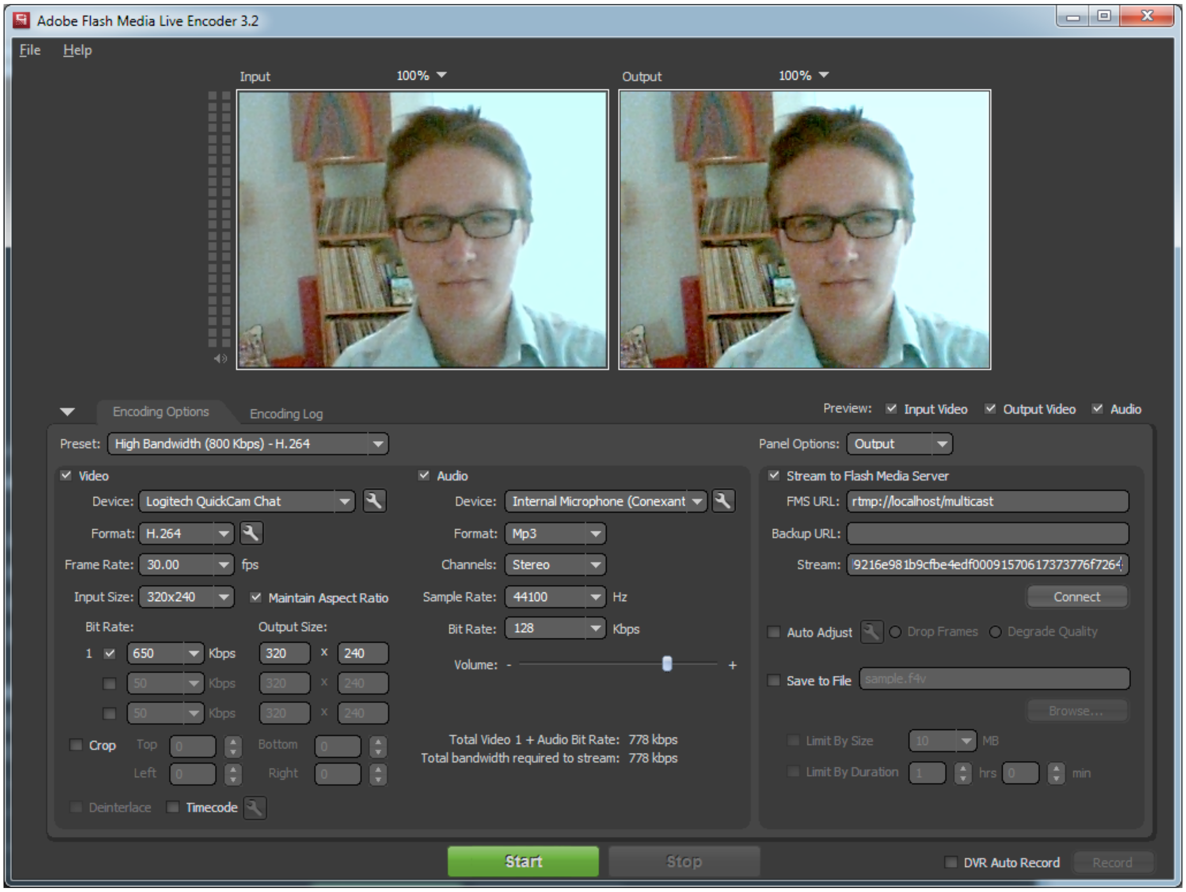
1 Launch Flash Media Live Encoder 3.1 or later and do the following:

2 From the Preset menu, select a single stream preset. The multicast solution does not support multi-bitrate streaming.

3 For AMS URL, enter the URL of the multicast service. If you're testing on the same computer that Adobe Media Server is running on, enter **rtmp://localhost/multicast**.

***Note:** Flash Media Live Encoder connects to Adobe Media Server over the RTMP protocol, not over the RTMFP protocol.*

4 Paste the Publisher Stream Name you copied from the Multicast Config Tool to the Stream field.



Preview: ☑ Input Video ☑ Output Video ☑ Audio

Preset: High Bandwidth (800 Kbps) - H.264

Panel Options: Output

☑ Video

Device: Logitech QuickCam Chat

Format: H.264

Frame Rate: 30.00 fps

Input Size: 320x240 ☑ Maintain Aspect Ratio

Bit Rate: Output Size:

1 ☑ 650 Kbps 320 x 240

□ 50 Kbps 320 x 240

□ 50 Kbps 320 x 240

□ Crop Top 0 Bottom 0

Left 0 Right 0

□ Deinterlace □ Timecode

☑ Audio

Device: Internal Microphone (Conexant

Format: Mp3

Channels: Stereo

Sample Rate: 44100 Hz

Bit Rate: 128 Kbps

Volume: - +

Total Video 1 + Audio Bit Rate: 778 kbps

Total bandwidth required to stream: 778 kbps

☑ Stream to Flash Media Server

FMS URL: rtmp://localhost/multicast

Backup URL:

Stream: 9216e981b9cfbe4edf000915706173737776f7264

□ Auto Adjust ○ Drop Frames ○ Degrade Quality

□ Save to File sample.f4v

□ Limit By Size 10 MB

□ Limit By Duration 1 hrs 0 min

□ DVR Auto Record

Flash Media Live Encoder configured to stream to the multicast application

5 Click Start to connect to the multicast service and start streaming.

6 Launch the Adobe Media Server Administration Console and click on View Applications > Clients. The RTMP client is the connection from FMLE to the server. The RTMFP client is a server-side peer established by the multicast service to republish the live stream into the target RTMFP Group.

Play a multicast stream

Play a stream in the multicast sample player:

1 Open *rootinstall*/tools/multicast/multicastplayer/multicastplayer.html in a browser.

2 To run Flash Player from the local file system, right-click on the screen, choose Global Settings, and do one of the following:

- On the Flash Player Help page, from the list in the top left, choose Global Security Settings panel. Click Edit locations > Add location > Browse for folder. Select folder containing the multicastplayer.swf file (*rootinstall*/tools/multicast/multicastplayer).
- In the Flash Player Settings Manager, choose Advanced and click Trusted Location Settings. Click Add, browse to the folder containing the multicastplayer.swf file (*rootinstall*/tools/multicast/multicastplayer). Click Confirm.

3 Reload the multicastplayer.html file in the browser. The Adobe Flash Player Settings manager displays a Peer Assisted Networking dialog. Click Allow to allow the peer-to-peer connection.

The multicastplayer plays the stream that Flash Media Live Encoder is publishing.

4 Open the Adobe Media Administration Console to see the new client connection.

Play a stream in the Adobe Media Server sample video player

1 Double-click rootinstall/samples/videoPlayer/videoplayer.html to open the sample video player in a browser.

The Adobe Media Server sample video player is built on OSMF Strobe Media Player 1.6.

2 Copy the manifest.f4m file you generated using the Multicast Config Tool from the *rootinstall*/tools/multicast/configurator folder to the *rootinstall*/samples/videoPlayer folder. Both the multicast sample player and the Adobe Media Server sample player must use the same manifest file.

3 In the sample video player, in the Stream URL text box, enter manifest.f4m. Click Play.

4 Open the Adobe Media Administration Console to see the new client connection. There are now 4 connections: 1 from Flash Media Live Encoder (RTMP), 1 from the multicast application publishing into the group (RTMFP), 1 from the multicast sample player, (RMTFP), and 1 from the Adobe Media Server sample video player (RTMFP).

More Help topics

"Building peer-assisted networking applications" on page 257

"Troubleshoot issues with streaming media" on page 111

Configure closed captioning

Closed captioning (CC) allows the content provider to display text overlayed on the video content thus providing additional information about the video content to the consumers. Closed captioning is primarily used as an accessibilty tool to show a transcription of the audio portion of the video content as it occurs. The term "closed" in closed captioning indicates that the captions are not made part of the video by default. Hence, the captions have to be decoded/extracted by an external tool. Adobe Media Server (AMS) 5.0.1 supports injecting closed caption data into the video stream supporting the following streaming techniques:

- HDS (Live and VOD)
- HLS (Live and VOD)
- RTMP

Closed captioning in AMS is implemented by injecting/embedding caption data within the H264 SEI NALU as specified in ANSI/SCTE 128 2010 Section 8. Adobe Media Server supports industry standard 608/708 closed captioning for video streaming to the following output channels/devices:

- Adobe Flash Player/AIR (through the OSMF framework)
- iOS devices (through HLS)

This document defines the input and output workflows, use cases, and configurations for supporting the 608/708 closed caption data in AMS. The closed caption data will be embedded inside the VOD stream and can be viewed through HDS, HLS, or RTMP based subscribers.

Closed captioning workflows

There are many techniques for implementing the caption data within the video stream. Few of these techniques are:

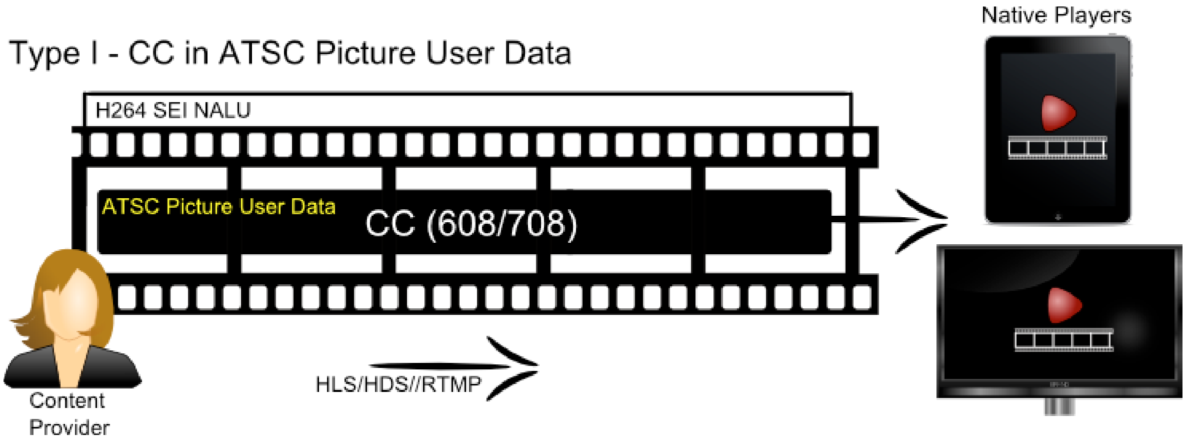
- Embedding the caption data in the video content. For instance, media stream or storage format can hold 608/708 caption data embedded inside the H264 SEI NALU as part of ATSC picture data.
- MP4 video content can contain a timed track, which can hold the caption data.

The following closed captioning workflows are supported in AMS:

- **Type 1** – Streaming captioning data stored in ATSC Picture User Data.
- **Type 2** – Extracting the caption data from a timed track and embed the data in a ATSC Picture User Data inside H264 SEI NALU (Network Abstraction layer Units).
- **Type 3** – Embedding captioning data inside an AMS-defined onCaptionInfo AMS message.

Type 1 – CC in ATSC Picture User Data

In this type, the CC input is natively supported for HLS. The Type 1 input can be just-in-time converted to an `OnCaptionInfo` message to support caption display on HDS and RTMP. The content providers can embed closed captioning data as part of the ATSC Picture User Data inside an H264 SEI NALU. However, this method is only suitable for video streams supporting H264 codec types.



Closed caption in ATSC picture user data

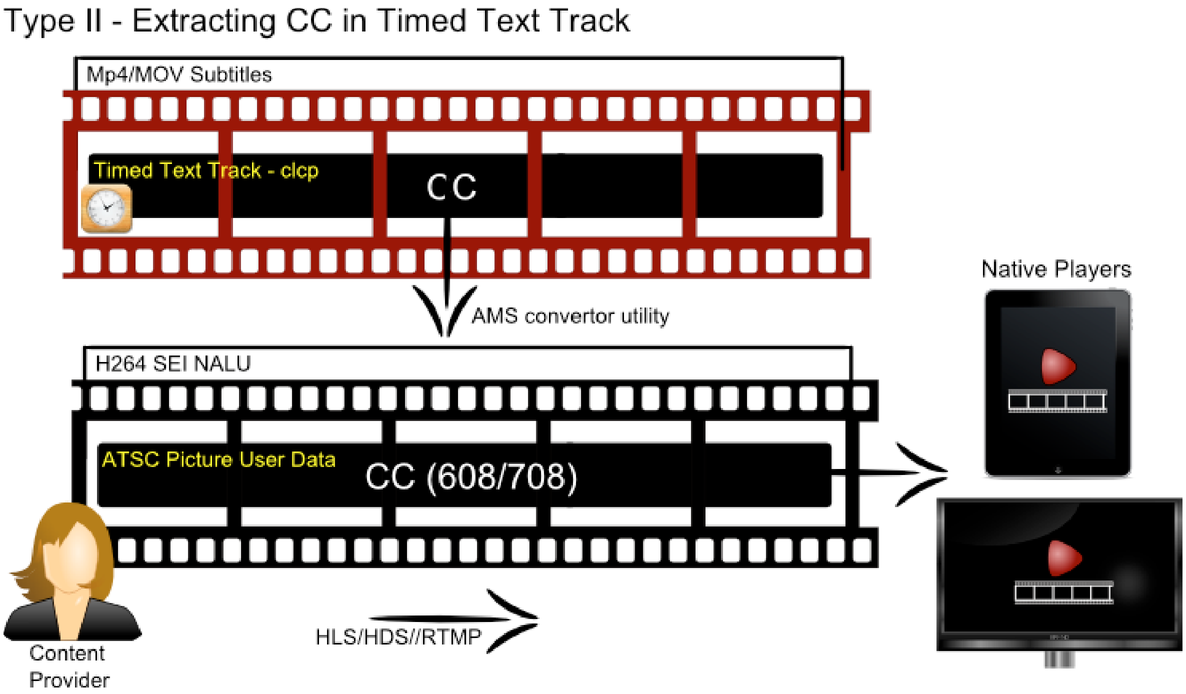
***Note:** This type is widely used for MPEG2 TS streams.*

HLS players can natively support such modified stream. AMS can be used to just-in-time package this embedded format into AMS defined `onCaptionInfo` messages to display captions over HDS and RTMP. Note that this is the ideal way of embedding the caption data. To prepare/package video content in this form, you can:

- Use AMS for packaging the caption data.
- Convert a natively available MPEG2 TS stream to this type.

Type 2 – Extracting CC in Timed Track

The closed caption data can also reside inside a separate timed track. Apple has defined a timed track format for QuickTime movie files.You can use the tool provided by AMS to convert this type of timed tracks containing caption data into an ATSC Picture User Data of Type I.



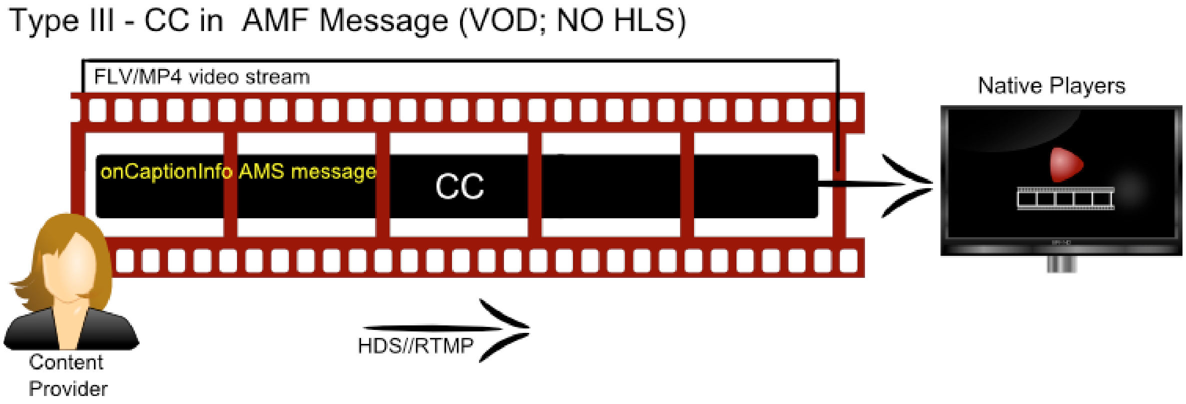
Closed captions in timed track

Type 3 – CC in AMF Message

The closed caption data can be just-in-time extracted from Type 1 data and stored inside the AMS-defined `onCaptionInfo` AMF message. The following file types are supported:

- MP4
- F4V

You can use AMS to package such files capitalizing on the `onCaptionInfo` message as a way of sending caption data as part of the live/VOD stream.

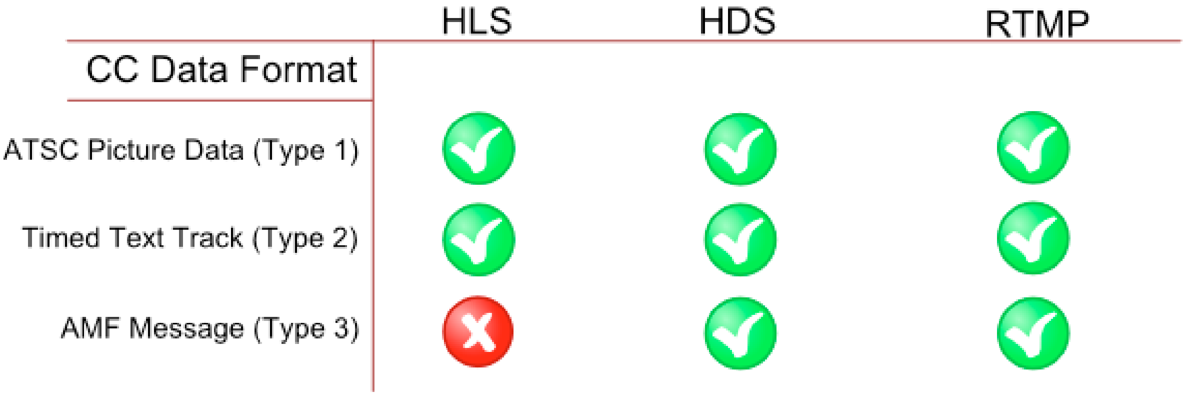

Type III - CC in AMF Message (VOD; NO HLS)
FLV/MP4 video stream
onCaptionInfo AMS message
CC
Native Players
HDS//RTMP
Content
Provider

Closed captions in AMF messages

***Note:** If content is recorded in this format, it can be played through RTMP and HDS without any additional packaging/streaming support from AMS.*

Note that HLS is not supported for this type.

The following table shows the summary of supported types for HLS, HDS, and RTMP:



CC Data Format	HLS	HDS	RTMP
ATSC Picture Data (Type 1)	✓	✓	✓
Timed Text Track (Type 2)	✓	✓	✓
AMF Message (Type 3)	✗	✓	✓

Closed caption types

Packaging captions

The HLS protocol demands packaging the caption data as `user_data_registered_itu_t_t35`, embedded as H264 SEI NALU in video access unit within the MPEG2 TS stream. Therefore, input video frames will be exactly translated into the MPEG2 transport stream. No repackaging in terms of caption data will be required. In case of HDS and RTMP, H264 SEI NALU will be converted to the onCaptionInfo data messages. In the incoming video stream, a NALU of type 6 with a payload of type 4 (user_data_registered_itu_t_t35) will be parsed for caption data. If there are multiple NALU in one access unit of the video frame, they will be packaged into a single data message. In this case, original caption data and corresponding NALU inside H264 access unit will remain intact. This will be done at the time of content packaging.

Configurations

AMS provides easy configurations to turn on and off the processing of the caption data on the server side. However, these configurations will not extract out the caption data encapsulated in any of the types defined in the previous section.

VOD use case

Configuring httpd.conf at the Server level (Only for HDS-VOD)

You can configure closed captioning at the server level by modifying the `JitGenerateCaptionInfo` element in Apache Server's `httpd.conf` file as follows:

```
<Location /hds-vod>
    JitGenerateCaptionInfo true
    ...
</Location>
```

Configuring jit.conf at the content level (Only for HDS-VOD)

You can configure closed captioning at the content level by modifying the `cc-info-enabled` element in `jit.conf` file as follows:

```
<hds>
    <cc-info-enabled>
        true
    </cc-info-enabled>
</hds>
```

Configuring server.xml (only for RTMP)

For RTMP, you can configure closed captioning by adding the following code snippet in the `server.xml` file:

```
<Mp4>
    <Playback>
    <GenerateCCInfo>true</GenerateCCInfo >
    </Playback>
</Mp4>
```

***Note:** In case of VOD, closed captioning data is supported only for H264-encoded MP4/F4V content.*

Live use case

Application level (HDS and RTMP)

You can configure closed captioning at the application level by modifying the `GenerateCCInfo` element in `Application.xml` file as follows:

```
<StreamManager>
    <Live>
        <GenerateCCInfo>true</GenerateCCInfo >
    </Live>
</StreamManager >
```

***Note:** If any of the tags is not present, it will be considered as "false" by AMS. Closed captioning will also be turned off in the Application.xml file.*

Stream level (HDS and RTMP)

You can also override the application level configuration by modifying the application's main.asc file by setting the Stream function `generateCCInfo`:

```
Stream.generateCCInfo = true;
```

***Note:** For both the VOD and live use cases, you must ensure that if the captions are already being supplied from the encoder or the VOD file in the Type 3 format, then these configuration must be turned off to avoid duplicates of the `onCaptionInfo` messages.*

***Note:** For the live use case, the `onCaptionInfo` message generation can be turned on and off through configuration only on the stream ingest server. If the stream is being relayed from another server, then these configuration won't be effective and by default will be off.*

Support for OSMF

Closed captioning can be enabled for OSMF players through the `OSMFCCDecoder` class. The `OSMFCCDecoder` class uses the `displayObject`, updates the video size, and resets the decoder. You can perform the following actions using the `OSMFCCDecoder`:

- Enable or disable the display of captions.
- Select caption channels.

- Modify various caption display attributes like font, text, and background colors.

The following sections describe the steps needed to get started with closed captioning in OSMF:

Step 1 - Preparing the assets

Before you begin, you need to prepare your assets to hold the closed captionining data.

Enable `GenerateCCInfo` in `Server.xml` for VOD playback as follows:

```
<Root>
    <Server>
        <Streams>
            <Mp4>
                <Playback>
                      <GenerateCCInfo>true</GenerateCCInfo>
                </Playback>
            </Mp4>
        </Streams>
    </Server>
</Root>
```

When the `GenerateCCInfo` option is enabled, AMS will process the video packets for closed captions and will generate the onCaptionInfo message, which will be sent to the client.

Enable `GenerateCCInfo` in `Application.xml` for live playback as follows:

```
 <Application>
    <StreamManager>
        <Live>
            <GenerateCCInfo>true</GenerateCCInfo>
        </Live>
    </StreamManager>
</Application>
```

Step 2 - Writing the ActionScript code

You need to initialize the `OSMFCCDecoder` and expose any customisable attributes as follows:

```
    var player:MediaPlayer = ...;

    // Initialize the player.
    // Listen for MediaPlayerStateChangeEvent on player and
    // when the state is MediaPlayerState.READY, then initialize the
    // decoder as shown below:
    var ccDecoder:OSMFCCDeco d er;

     // Create a new instance
    ccDecoder = new OSMFCCDecoder();
     // Bind the instance to your media player

    ccDecoder.mediaPlayer = player;
    // Bind the instance to the media player view area
    // that can be used to render.
    ccDecoder.mediaContainer = mediaContainer;
    // Enable the instance
    ccDecoder.enabled = true;
    // Optionally, select a type and service
    // At present, only one type is supported; CEA-708 wrapped over 608,
    // which is identified by the value CCType.CEA708.
    ccDecoder.type = CCType.CEA708;
    ccDecoder.service = CEA708Service.CC1;
```

***Note:** You must have atleast one instance of `OSMFCCDecoder` for a video player.*

The `OSMFCCDecoder` contains a number of properties that can be exposed through the UI controls. Most of these properties enable overriding the font and color attributes and are required under the FCC guidelines. You are responsible for adding the UI controls to expose these properties. Also, you should add the necessary code to save and restore the settings between sessions. UI controls must be provided in the video player to enable/disable closed captioning display and to select the desired caption service/channel. UI controls may be provided to allow the end user to select values for the displayed font size, background color, and opacity.

The following code snippet describes the steps involved in exposing custom attributes through ActionScript:

```
 /**
    * The type of closed captioning data to decode and render.
    * A value of null indicates no selected type.
    * Valid values are values of the CC Type enumeration class.
    * The default value is null .
    *
    * @throws ArgumentError if the specified type is not supported.
    * /
    public function gettype():String;
    public function settype(value:String):void;
    / **
    * The MediaPlayer this OSMFCCDecoder is associated with.
    * Assig ning any value disassociates the OSMFCCDecoder from a
    * currently assigned MediaPlayer (if any).
    * If the assigned value is non-null, the OSMFCCDecoder will
    * automatically register itself as an event listener on
    * the provided MediaPlayer and access its view property
    * to use as the video surface to render captions over.
    */
    public function get mediaPlayer():MediaPlayer
    public function set mediaPlayer(value:MediaPlayer):void
    /**
```

```
* Whether caption data decoding and rendering is enabled or disabled.
* The default value is true.
* When this property transitions from true to false a side-effect
* is a reset of internal decoder state and
* any currently rendered captions are cleared.
*/
public function get enabled():Boolean
public function set enabled(value:Boolean):void
/**
* The caption "service" to decode and render.
* For CEA-708 wrapped over 608 closed captioning, supported values
* are values of the CEA708Service enumeration class.
* CEA708Service.CC1, CEA708Service.CC2, CEA708Service.CC3, CEA708Service.CC4
* @throws ArgumentError if the assigned service value is not supported
* or available for the closed captioning
* type specified at construction time.
 */
public function get service():String
public function set service(value:String):void
/**
* Font override to apply to rendered text.
* Default value is null, indicating no override, and text will
* assume its intrinsic font according to caption
* authoring.
* To override the font assign an embedded or system font
* name and to disable an override assign null.
*/
public function get font():String
public function set font(value:String):void
/**
* Text color override to apply to rendered text.
* Default value is -1, indicating no override, and text
* will assume its intrisic color which may vary according
* to caption authoring.
* To override the text color assign a RGB hex value (e.g. 0xFF00FF)
* and to disable an override assign -1.
* @throws ArgumentError if the value is not -1 and not within the RGB hex range.
*/
public function get textColor():int
public function set textColor(value:int):void
/**
* Background color override to apply to the background of text character cells.
* Default value is -1, indicating no override, and cell backgrounds
* will assume their intrinsic color which
* may vary according to caption authoring.
* To override the background color assign a RGB hex value (e.g. 0xFF00FF)
* and to disable an override assign -1.
* @throws ArgumentError if the value is not -1 and not within the RGB hex range.
*/
public function get backgroundColor():int
public function set backgroundColor(value:int):void
/**
* Text opacity override to apply to rendered text.
* Default value is -1, indicating no override, and text
* will assume its intrinsic opacity which may vary
* according to caption authoring.
* To override text opacity assign a value between 0 (fully transparent)
* to 1 (fully opaque) and to disable
* an override assign -1.
* @throws ArgumentError if the value is not -1 and not within the range of [0, 1].
*/
public function get textOpacity():Number
public function set textOpacity(value:Number):void
/**
* Background opacity override to apply to the background of text character cells.
* Default value is -1, indicating no override, and backgrounds will
* assume their intrinsic opacity which may
* vary according to caption authoring.
* To override background opacity assign a value between 0 (fully transparent)
* to 1 (fully opaque) and to
* disable an override assign -1.
* @throws ArgumentError if the value is not -1 and not within the range of [0, 1].
*/
public function get backgroundOpacity():Number
public function set backgroundOpacity(value:Number):void
/**
* Resets the OSMFCCDecoder's internal state, clearing any
* currently rendered captions.
* The currently selected caption service and display overrides are not changed.
*/
public function reset():void
/**
* The default font to use when no custom font is specified.
* This is set to a reasonable default for the device. When using custom devices,
* you may alter this property to change the default font.
*/
public function get defaultFont():String
public function set defaultFont(value:String):void
/**
* Edge type override to apply to rendered text.
*
* Valid values are indicated by the EdgeType enumeration, or null to
* indicate no override. Valid Values:
* EdgeType.NONE: Indicates that no edge decoration should be used.
* EdgeType.DEPRESSED: Indicates that the edge will appear as depressed.
* EdgeType.UNIFORM: Indicates that the edge will with a uniform border.
* EdgeType.LEFT_DROP_SHADOW: Indicates that the edge will with a
* shadow falling to the left.
* EdgeType.RIGHT_DROP_SHADOW: Indicates that the edge will with a
* shadow falling to the right.
*
```

```
* @default null
*
* @throws ArgumentError if the value is not null and not a memory of
* the EdgeType enumeration.
**/
public function get edgeType():String
public function set edgeType(value:String):void
/**
* Edge color override to apply to rendered text.
* The value should be an RGB hex value (e.g. 0xFF00FF).
*
* A value of -1 indicates no override. A reasonable
* default edge color will be selected.
*
* @default -1
*
* @throws ArgumentError if the value is not -1 and not within the RGB hex range.
**/
public function get edgeColor():int
public function set edgeColor(value:int):void
```

Note: *The server side closed captioning support is only available for H.264 videos. The DXP and TTML specifications are supported in the side-car files to playback closed captions (for VOD) in OSMF, HDS, or RTMP.*

Using the ccConvertor tool

The ccConvertor tool is a command line utility (bundled with Adobe Media Server) that can be used to convert any MOV files with Closed Caption content (clcp tracks) into an MP4 file with embedded caption NALUs in 708 format (having 608 wrapped in it).

Usage

The usage of the ccConvertor tool on Windows is as follows:

```
ccConvertor.exe --input-file=<file> --[setting]=[arg]...
```

Parameters

The following table describes the supported input parameters:

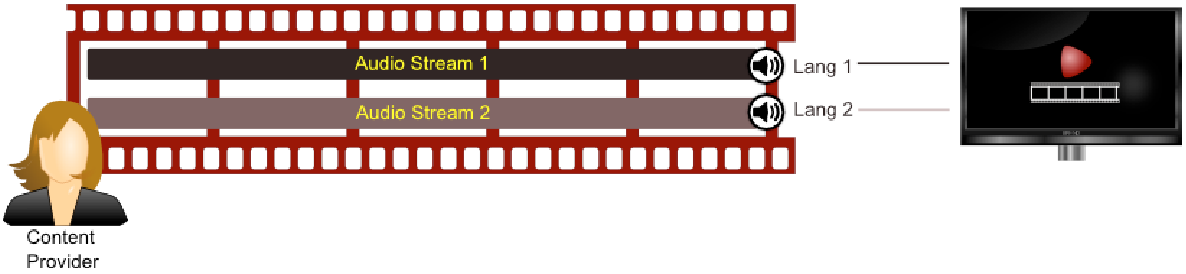
Parameter	Description
`--input-file arg`	Signifies the path to the source container file. The path can be an absolute path or a relative path.
`--output-file arg`	The path to the destination file. Note that this path should not be the same as the input file. The path can be an absolute path or a relative path.
`--conf-file arg`	The configuration file that contains the settings for the packaging process.

Note that if a relative path is passed through the configuration file, the path is relative to the directory containing the configuration file.

To understand the configuration file format, see the sample configuration file available at <*AMS_INSTALL_DIR*>\tools\ccConvertor\sample_cfg.xml.

Configure alternate audio

Adobe Media Server has support for including multiple language tracks for HTTP video streams, without requiring duplication and repackaging of the video for each audio track. This feature, called as "late" binding of audio tracks allow content providers to easily provide multiple language tracks for a given video asset, at any time before or after the asset's initial packaging. The initial packaging of the video asset can include an audio track, and the publisher can still choose to provide one or more additional audio tracks for the video. Video players can provide support for allowing the viewers to switch between audio tracks either before or during playback.



Clients can pick the appropriate audio

The primary use case of this feature is that content providers can publish video content in multiple languages. Viewers can subscribe to such videos using the OSMF player, which allows audio track switching before or during playback.

Playback workflow for HDS-VOD

The following steps provide an overview of the content publishing workflow in case of HDS-VOD:

1 The content providers will load videos in the 'vod' directory along with its set level manifest F4M file. This set level manifest file must contain information on which video to play by default and which audio track to be used as an alternate audio. For instance, if you want to publish a video stream, video1.mp4 along with the language tracks video1_french.mp4 and video1_spanish.mp4, providing the users a choice to switch audio, you can create a set level manifest F4M video1_master.f4m and place it in the vod sub directory. The set level manaifest file can have the following content:

```
<?xml version="1.0" encoding="utf-8"?>
    <manifest xmlns="http://ns.adobe.com/f4m/2.0">
        <media href="../hds-vod/video1.mp4.f4m" bitrate="1000" />
        <media href="../hds-vod/audio-only/video1_french.mp4.f4m"
            bitrate="200" type="audio" label="french" lang="fr" alternate="true" />
        <media href="../hds-vod/audio-only/video1_spanish.mp4.f4m" bitrate="200"
            type="audio" label="spanish" lang="es" alternate="true" />
    </manifest>
```

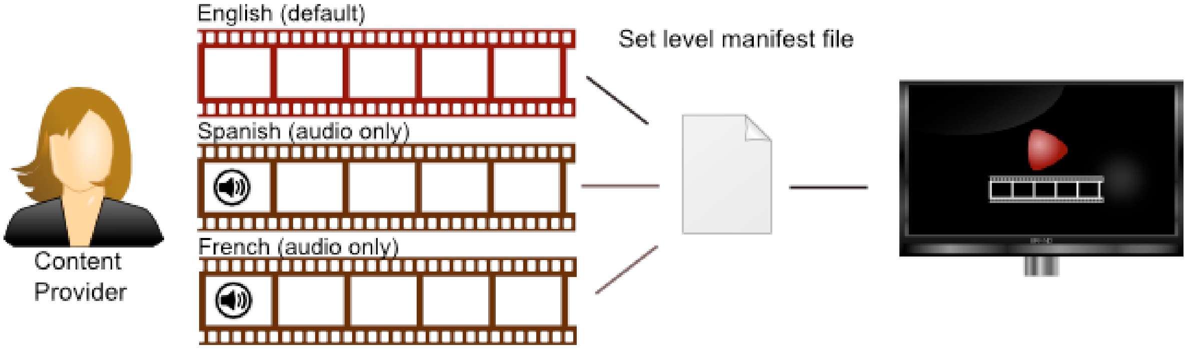
2 When a viewer plays the stream, the player will load this set level manifest file and will play video1.mp4 by default with its embedded audio data. The player will request the manifest file video1.mp4.f4m from the server. The Apache module for on demand HDS will create the manifest file and will send it to the player. The player will then request for fragments of the video1.mp4 file and will play the fragments.

3 When the viewer switches to another audio label (in this case, french or spanish), the player will request for the manifest file for the selected option. For instance, if the viewer has opted for the french track, the player requests for the manifest video1_french.mp4.f4m file to get the audio-only stream.

4 The Adobe Media Server will know that the request for F4M is coming with the 'audio-only' element, hence will create the manifest file just in time almost similar to step 2.

5 Now, the player will request for fragments from both the media files (video1.mp4 and video1_french.mp4) using their respective manifest F4M files. For the first media file, the player will continue fetching fragments. But for the second media file, video1_french.mp4, the player will try to get fragments for the media URL with 'audio-only' value in the URL as supplied in its manifest F4M file. The request for fragment for this media file will be of this format '/hds-vod/audio-only/video1_french.mp4Seg1-Frag1'

6 While serving the fragment request, Adobe Media Server understands the 'audio-only' request and sends only the audio data for the requested fragment of file 'video1_french.mp4'. These fragments are made just in time and served. If the 'audio-only' element is not present, video data for the fragment is served.



Streaming alternate audio on demand

Playback workflow for HLS-VOD

The following steps provide an overview of the content publishing workflow in case of HLS-VOD:

1 The content providers will load videos in the 'vod' directory along with its variant playlist M3U8 file. This variant playlist file must contain information on which video to play by default and which audio track to be used as an alternate audio. For instance, if you want to publish a video stream, video1.mp4 along with the language tracks video1_french.mp4 and video1_spanish.mp4, providing the users a choice to switch audio, you can create a variant playlist M3U8 video1_master.m3u8 and place it in the vod sub directory. The variant playlist file can have the following content:

```
#EXTM3U
#EXT-X-MEDIA:TYPE=AUDIO,GROUP-ID="aac",NAME="English",
DEFAULT=YES,AUTOSELECT=YES,LANGUAGE="en", URI="../hls-vod/audio-only/video1.mp4.m3u8"
#EXT-X-MEDIA:TYPE=AUDIO,GROUP-ID="aac",NAME="French",
DEFAULT=NO,AUTOSELECT=NO,LANGUAGE="fr", URI="../hls-vod/audio-only/video1_french.mp4.m3u8"
#EXT-X-MEDIA:TYPE=AUDIO,GROUP-ID="aac",NAME="Spanish",
DEFAULT=NO,AUTOSELECT=NO,LANGUAGE="es", URI="../hls-vod/audio-
only/video1_spanish.mp4.m3u8"
#EXT-X-STREAM-INF:BANDWIDTH=1280000,CODECS="mp4a.40.5",AUDIO="aac"
../hls-vod/video1.mp4.m3u8
```

2 When a viewer plays the stream, the player will load this variant playlist file and will play video1.mp4 by default with its embedded audio data. The player will request the playlist file video1.mp4.m3u8 from the server. The Apache module for on demand HLS will create the playlist file and will send it to the player. The player will then request for fragments of the video1.mp4 file and will play the fragments.

3 When the viewer switches to another audio label (in this case, french or spanish), the player will request for the playlist file for the selected option. For instance, if the viewer has opted for the french track, the player requests for the playlist video1_french.mp4.m3u8 file to get the audio-only stream.

4 The Adobe Media Server will know that the request for M3U8 is coming with the 'audio-only' element, hence will create the playlist file just in time almost similar to step 2.

5 Now, the player will request for fragments from both the media files (video1.mp4 and video1_french.mp4) using their respective playlist M3U8 files. For the first media file, the player will continue fetching fragments. But for the second media file, video1_french.mp4, the player will try to get fragments for the media URL with 'audio-only' value in the URL as supplied in its playlist M3U8 file. The request for fragment for this media file will be of this format '/hds-vod/audio-only/video1_french.mp4Seg1-Frag1'

6 While serving the fragment request, Adobe Media Server understands the 'audio-only' request and sends only the audio data for the requested fragment of file 'video1_french.mp4'. These fragments are made just in time and served. If the 'audio-only' element is not present, video data for the fragment is served.

***Note:** To package an audio-only stream for HLS, see "Packaging an audio-only stream (HLS)" on page 11. To publish an audio-only stream for HLS, see "Publish an audio-only stream (HLS)" on page 11*

Configure content protection

See "Content protection" on page 115.

Configure HTTP Dynamic Streaming and HTTP Live Streaming

Overview of HTTP Dynamic Streaming and HTTP Live Streaming

Streaming media over HTTP

Delivering content over HTTP is usually called "progressive download". The content must transfer from the server to the client in a progression from the beginning to the end of a file. A client cannot seek to a forward location until that location and all the data before it has downloaded.

Delivering content over RTMP is called "streaming". The client creates a socket connection to the server (such as Adobe Media Server) over which the content is sent in a continuous stream. The client can seek to any point in the content instantly, regardless of how much data has been transferred.

Adobe HTTP Dynamic Streaming combines these approaches to introduce HTTP streaming to the Flash Platform. HTTP Dynamic Streaming packages media files into fragments that Flash Player clients can access instantly without downloading the entire file. Adobe HTTP Dynamic Streaming contains several components that work together to package media and stream it over HTTP to Flash Player and AIR. HTTP Dynamic Streaming supports multi-bitrate streaming, DVR, and Adobe® Flash® Access™ protection.

In Adobe Media Server 5, Adobe HTTP Dynamic Streaming supports Apple HTTP Live Streaming. Use the same HTTP Dynamic Streaming workflow to package and stream live and on-demand content to all devices that support Apple HTTP Live Streaming.

***Note:** Adobe Media Server does not support adding Timed Metadata for HTTP Live Streaming.*

HTTP Dynamic Streaming components

The Adobe HTTP Streaming solution contains the following components:

Live Packager Live Packager is a Adobe Media Server application installed to *rootinstall*/applications/livepkgr. The server ingests a live stream over RTMP and the Live Packager translates it into F4F files and MPEG-2 TS files in real-time.

F4F HTTP Module An Apache HTTP Server module that serves files to Flash Player and AIR. The F4F module (also known as the HTTP Origin Module), serves live content. It also serves on-demand content that was packaged offline with the File Packager tool.

The F4F HTTP Module installs with Flash Media Server 4 and later to *rootinstall*/Apache2.2/modules/mod_f4fhttp.so. The module is also available from adobe.com as a stand-alone Apache module.

HLS HTTP Module An Apache HTTP Server module that serves files to iOS devices and Mac OS.

The HTTP HLS Module installs with Adobe Media Server 5 and later to *rootinstall*/Apache2.2/modules/mod_hlshttp.so. It is not available as a stand-alone Apache module.

JIT HTTP Module An Apache HTTP Server module that packages on-demand files just-in-time and serves them to Flash Player and AIR. "Just-in-time" means that media files are packaged in real-time when clients request them.

The JIT HTTP Module installs with Adobe Media Server 5 and later to *rootinstall*/Apache2.2/modules/mod_jithttp.so. It is not available as a stand-alone Apache module.

File Packager A command-line, offline tool that translates on-demand media into F4F fragments. The File Packager is located in the *rootinstall*/tools/f4fpackager folder and is available from adobe.com.

OSMF media players There are several media players built on the Open Source Media Framework (OSMF) that support HTTP Dynamic Streaming to Flash Player and AIR. See "Pre-built media players" on page 3.

F4F File Format Specification The F4F file format describes how to divide media content into segments and fragments. The Live Packager, the File Packager, and the JIT HTTP module output content based on this specification. Each fragment has its own bootstrap information that provides cache management and fast seeking. For more information, see F4F File Format Specification.

F4M File Format Specification The Adobe Media Manifest file format contains information about a package of files that the HTTP Origin Module can serve. Manifest information includes codecs, resolutions, and the availability of files encoded at multiple bit rates. Manifest information also includes DRM data. The media player uses the F4M file to play a piece of media. For more information, see F4M File Format Specification.

Adobe Access Adobe Access delivers protected media to Flash Player. To use HTTP Dynamic Streaming with Adobe Access, use the File Packager and Adobe Media Server to both package and encrypt content. For more information, see Protecting content with Adobe Access.

Adobe Media Server 5 adds support for Adobe Access level protection without using a Adobe Access server.

Workflow for streaming live media over HTTP

The best way to experience the workflow for live HTTP streaming is to complete a tutorial. See "Stream live media (HTTP)" on page 5.

The following is the workflow for all Adobe HTTP Dynamic Streaming and Apple HTTP Live Streaming use cases:

1 Use the default live event or create a live event.

  The default live event is *rootinstall*/applications/livepkgr/events/_definst_/liveevent. Each live event contains configuration settings for one set of content. For more information, see "Create and configure live events" on page 54.

2 (Optional) To configure multi-bitrate streaming, use the Set-level F4M/M3U8 File Generator tool to generate an set-level manifest file. The set-level manifest file contains information about each stream. See "Publish and play live multi-bitrate streams over HTTP" on page 8.

3 (Optional) To configure DVR for Adobe HTTP Dynamic Streaming, create a set-level manifest file.

To configure a Sliding Window for Apple HTTP Live Streaming, configure the httpd.conf, Application.xml, or Event.xml file.

See "Configure DVR (HDS)" on page 13 and "Configure a sliding window (HLS)" on page 14.

4 (Optional—Adobe HTTP Dynamic Streaming) To encrypt content for protected HDS, see "Configuring content protection for HDS" on page 115.

To encrypt content for use with Adobe Access, edit the Event.xml file. See "Encrypt content for Adobe Access protection" on page 72.

5 (Optional—Apple HTTP Live Streaming) To encrypt content for Apple HTTP Live Streaming, see Configuring content protection for HLS.

6 Create a crossdomain.xml file and copy it to the *rootinstall*\webroot folder.

A crossdomain.xml file allows Flash Player clients hosted on other domains to access data from this domain. For more information, see Website controls (policy files) in the ActionScript 3.0 Developer's Guide.

7 Publish a stream to the server.

8 (Adobe HTTP Dynamic Streaming) Use Strobe Media Playback to play media. Strobe Media Playback is installed with Adobe Media Server.

9 (Apple HTTP Live Streaming) Play the content on iOS or MacOS.

For supported devices, see Apple HTTP Live Streaming documentation.

Workflow for streaming on-demand media over HTTP

The best way to experience the workflow for on-demand HTTP streaming is to complete a tutorial. See "Stream on-demand media (HTTP)" on page 19.

The following is the workflow for Adobe HTTP Dynamic Streaming and Apple HTTP Live Streaming use cases:

1 Encode media files and copy them to the *rootinstall*/webroot/vod directory on Adobe Media Server.

To configure this location, see Configure the location of content (HDS and HLS).

2 (Optional) To configure multi-bitrate streaming, use the Set-level F4M/M3U8 File Generator tool to generate a set-level manifest file. See "Play on-demand multi-bitrate media files over HTTP" on page 22.

3 (Optional—Adobe HTTP Dynamic Streaming) To encrypt content for protected HDS (without Adobe Access), see "Configuring content protection for HDS" on page 115.

To encrypt content for use with Adobe Access, edit the jit.conf file. See "Encrypt content for Adobe Access protection" on page 72.

4 (Optional—Apple HTTP Live Streaming) To encrypt content for Apple HTTP Live Streaming, configure the jit.conf file or the Apache httpd.conf file. See Protect content for Apple HLS.

5 Create a crossdomain.xml file and copy it to the *rootinstall*\webroot folder.

A crossdomain.xml file allows Flash Player clients hosted on other domains to access data from this domain. For more information, see Website controls (policy files) in the ActionScript 3.0 Developer's Guide.

6 (Adobe HTTP Dynamic Streaming) Use Strobe Media Playback to play media. Strobe Media Playback is installed with Adobe Media Server.

7 (Apple HTTP Live Streaming) Play the content on an iOS device or on MacOS.

For supported devices, see Apple HTTP Live Streaming documentation.

Differences in HTTP Dynamic Streaming between Flash Media Server 4.0 and 4.5

The following are differences in HTTP Dynamic Streaming (HDS) between Flash Media Server 4.0 and 4.5:

- Simplified request URLs for live streaming.

  See "Differences in HTTP live URLs from Flash Media Server 4.0 to Adobe Media Server 5" on page 13.

- Support for just-in-time on-demand packaging.

  When a media player requests an on-demand media files from the *rootinstall*/webroot/vod folder using HDS, Apache packages the stream in real-time. You do not need to pre-package the media files for HTTP streaming as you did with Flash Media Server 4.0. See "Stream on-demand media (HTTP)" on page 19.

- Use a jit.conf file to configure on-demand streaming at the stream-level. Copy the jit.conf file to the same folder as the media. See "Configure on-demand HTTP streaming" on page 88.

- Use set-level F4M and M3U8 files to configure a set of streams for adaptive bitrate streaming.

  The set-level files describe the bitrates of the media. The live and just-in-time packagers generate stream-level F4M and M3U8 files to describe the individual pieces of media. The set-level files can live on any webserver. See "Publish and play live multi-bitrate streams over HTTP" on page 8 and "Play on-demand multi-bitrate media files over HTTP" on page 22.

Create and configure live events

You can use the livepkgr application to serve an unlimited number of live streaming events over HTTP (for example, a debate, a sporting event, and a town hall meeting). Each live streaming event requires its own configuration settings for multi-bitrate streaming, DVR, and content protection.

A *live event* is a configuration level within the livepkgr application. Like a Adobe Media Server application, a live event is a folder on the server. The folder contains two configuration files: Manifest.xml and Event.xml. Use these configuration files to configure a set of streams.

Also like a Adobe Media Server application, the name of the live event folder is the name of the live event. The Adobe Media Server application that packages live content for HTTP streaming is called "livepkgr". The livepkgr application contains a live event called "liveevent":

***rootinstall*/applications/livepkgr/events/_definst_/liveevent**

Note: *You can configure the location of the "applications" folder. You can also configure virtual directory mappings for the "streams" folder. However, you cannot configure virtual directory mappings for the "events" folders.*

Streams in a live event are packaged as fragments and written to disk. A live event (the stream content and the metadata) exists until you delete it. A media player can access the content after the source stream has stopped publishing.

Create a live event

For each live streaming event, create a live event folder.

1 Create the following "events" folder structure in the livepkgr application folder:

   rootinstall*/applications/livepkgr/events/*applicationinstancename*/*liveeventname

   The following is a new live event called "liveevent2":

   ***rootinstall*/applications/livepkgr/events/_definst_/liveevent2**

   You can also create a live event for a different instance of the livepkgr application, as in the following:

   ***rootinstall*/applications/livepkgr/events/anotherappinstance/liveevent**

2 Copy the Events.xml file from the default liveevent folder to the new folder:

   ***rootinstall*/applications/livepkgr/events/_definst_/liveevent2/Events.xml**

3 Copy the Manifest.xml file from the default liveevent folder to the new folder:

   ***rootinstall*/applications/livepkgr/events/_definst_/liveevent2/Manifest.xml**

4 Open the Events.xml file in a text editor and change the event name:

```
<Event>
    <EventID>liveevent2</EventID>
    <Recording>
        <FragmentDuration>4000</FragmentDuration>
        <SegmentDuration>400000</SegmentDuration>
        <DiskManagementDuration>3</DiskManagementDuration>
    <Recording>
</Event>
```

Configure a live event

For information about the features you can configure in the Event.xml file, see "Configure live HTTP streaming at the event level (Event.xml)" on page 83.

Write server-side code to assign a stream to a live event

Important: *You do not need to write server-side code for HTTP Dynamic Streaming. The livepkgr application has a Server-Side ActionScript file that assigns streams to a live event. To see the code, open rootinstall/applications/livepkgr/main.asc in a text editor. Read this section to understand the code.*

The Live Packager processes streams that are prefixed with `f4f:`. Flash Media Live Encoder doesn't support the `f4f:` prefix so you must add it to the stream name in a server-side script or in a Adobe Media Server Authorization Plug-in.

An application can contain more than one live event. For that reason, a stream published to an application is not associated with a live event by default. Use a server-side script or the Adobe Media Server Authorization Plug-in (not both) to associate a live stream with a live event. You can associate a live stream with only one live event.

Associate a stream during the publish event. When the stream is published and associated, the server creates a stream record file (.stream) for the stream in the event directory:

applications/*appname*/events/*appinstancename*/*liveeventname*/*livestream*.stream

The stream record file contains information about the location of the packaged stream files. The HTTP Origin Module uses this information to generate an .f4m manifest file. For more information, see "Understanding the application flow for live HTTP Dynamic Streaming" on page 94.

Use Server-Side ActionScript

To see an example of this code, open rootinstall/applications/livepkgr/main.asc in a text editor.

❖ In the Server-Side ActionScript code, use the `application.onPublish` event to do the following:

- Use the `Stream.liveEvent` property to associate a live stream with a live event before the server starts recording.
- Add the `f4f:` prefix to the stream name. This tells the server to package the stream.
- Call `Stream.record()` to record the stream.

Use an Authorization Plug-in

To perform these tasks with an Authorization plug-in, you must write the plug-in code.

❖ In the Authorization Plug-in code, use the `E_PUBLISH` event to do the following:

- Set the `F_STREAM_TYPE` field to `"f4f"`.
- Set the `F_STREAM_LIVE_EVENT` field to `"`*`liveeventname`*`"`. This example uses "liveevent".
- Set the `F_STREAM_PUBLISH_TYPE` field to `0` which means "record".

The following is sample C++ code that handles the `E_PUBLISH` event:

```
case IFmsAuthEvent::E_PUBLISH:
{
    // The name of the FMS app to which the live stream is published.
    char* pLiveApp = "livepkgr";
    // The stream type used for HTTP Dynamic Streaming.
    char* pStreamType = "f4f";
    // The name of the live event defined in the livepkgr app.
    char* pLiveEvent = "liveevent";
    // The Auth Plug-in affects all apps on the server.
    // We only want to process streams published to the livehttp app.
    char* pAppName = getStringField(m_pAev, IFmsAuthEvent::F_APP_NAME);
    if (pAppName && !strncmp(pAppName, pLiveApp, strlen(pLiveApp)))
    {
        // Set the stream type.
        setStringField(m_pAev, IFmsAuthEvent::F_STREAM_TYPE, pStreamType);
        // Set the publish type to record.
        //0 record, 1 append, -1 live
        setI32Field(m_pAev, IFmsAuthEvent::F_STREAM_PUBLISH_TYPE, 0);
        // Associate the stream with a live event.
        setStringField(m_pAev, IFmsAuthEvent::F_STREAM_LIVE_EVENT, pLiveEvent);
    }
}
```

Content storage (HDS and HLS)

Configure the beginning of the request URL

When a media player requests content from the server, it passes the server a request URL. The section of the request URL following the server name (and optional port number) is defined in the `Location` directive path in the Apache httpd.conf file. For example, the following is the Location directive for HDS on-demand streaming:

```
<IfModule jithttp_module>
<Location /hds-vod>
    HttpStreamingJITPEnabled true
    HttpStreamingContentPath "../webroot/vod"
    JitFmsDirPath ".."
    Options -Indexes FollowSymLinks
</Location>
</IfModule>
```

When Apache receives a request with the path /hds-vod, it uses the jithttp_module (just-in-time) to process the request. The jithttp_module uses the values of the directives nested within the `Location` directive.

To configure HTTP streaming for multiple tenants, or for multiple applications, add `Location` directives with different paths. The following table lists the syntax for request URLs, the `Location` directive paths are in bold:

Streaming type	Request URL syntax
Adobe HDS live	http://<fms-dns-or-ip>/**hds-live**/livepkgr/<appinstname>/<eventname>/<streamname>.f4m
Apple HLS live	http://<fms-dns-or-ip>/**hls-live**/livepkgr/<appinstname>/<eventname>/<streamname>.m3u8
Adobe HDS on-demand with just-in-time packaging	http://<fms-dns-or-ip>/**hds-vod**/<streamname>.<fileextension>.f4m
Apple HLS on-demand	http://<fms-dns-or-ip>/**hls-vod**/<streamname>.<fileextension>.m3u8

Add a Location directive

Note: *This task uses HDS on demand, but the steps apply to all types of HTTP streaming.*

1 Open *rootinstall*/Apache2.2/conf/httpd.conf in a text editor.

2 Location the `Location` directive for the streaming type you want to edit, copy it, and paste it.

3 For example, to create a `Location` for HDS on-demand streaming, copy the /hds-vod section and paste it underneath the existing section.

```
<IfModule jithttp_module>
<Location /hds-vod>
    HttpStreamingJITPEnabled true
    HttpStreamingContentPath "../webroot/vod"
    JitFmsDirPath ".."
    Options -Indexes FollowSymLinks

# Uncomment the following directives to enable encryption
# for this location.
#   EncryptionScope server
#   ProtectionScheme phds
</Location>
</IfModule>
```

4 In the copied section, change the `Location` path to /flash-vod. You can also edit the `HttpStreamingContentPath` directive to change the location of the on-demand media on disk.

```
<IfModule jithttp_module>
<Location /flash-vod>
    HttpStreamingJITPEnabled true
    HttpStreamingContentPath "../webroot/vod2"
    JitFmsDirPath ".."
    Options -Indexes FollowSymLinks

# Uncomment the following directives to enable encryption
# for this location.
#   EncryptionScope server
#   ProtectionScheme phds
</Location>
</IfModule>
```

***Note:** To change the location of content for live streaming, see "Configure the location of live HDS and HLS content" on page 59.*

5 Create the *rootinstall*/webroot/vod2 folder and copy the sample1_1500kbps.f4v file to it from the /webroot/vod folder.

6 Restart Apache HTTP Server. The service name is FMSHttpd.

7 Double-click *rootinstall*/samples/videoPlayer/videoplayer.html to open the Adobe Media Server sample video player.

8 Enter the following in the Stream URL text box:

http://localhost/flash-vod/sample1_1500kbps.f4v.f4m

You can play any file from the /webroot/vod2 folder.

Configure the location of on-demand HDS and HLS content

To configure the location of on-demand content on disk, edit the Apache httpd.conf file. Use the following directives:

Directive	Description
`HttpStreamingContentPath`	The root location of the streams. This value can be absolute or relative to the Apache root folder. By default, the location for on-demand HDS and HLS is `"../webroot/vod"`.

1 Open *rootinstall*/Apache2.2/conf/httpd.conf

2 To change the location on the server where media is stored for on-demand streaming, edit the `HttpStreamingContentPath` directive. For example, the following changes the location to c:\hds_vod_content:

```
<IfModule jithttp_module>
<Location /hds-vod>
    HttpStreamingJITPEnabled true
#   HttpStreamingContentPath "../webroot/vod"
    HttpStreamingContentPath "C:\hds_vod_content"
    JitFmsDirPath ".."
    Options -Indexes FollowSymLinks

# Uncomment the following directives to enable encryption
# for this location.
#   EncryptionScope server
#   ProtectionScheme phds
</Location>
</IfModule>

<Location /hls-vod>
    HLSHttpStreamingEnabled true
    HLSMediaFileDuration 8000
#   HttpStreamingContentPath "../webroot/vod"
    HttpStreamingContentPath "C:\hds_vod_content"
    HLSFmsDirPath ".."

# Uncomment the following directives to enable encryption
# for this location.
#    HLSEncryptionScope server
#    HLSEncryptCipherKeyFile "../creds/vodkey.bin"
#    HLSEncryptKeyURI        "https://<ServerName>/hls-key/vodkey.bin"

    Options -Indexes FollowSymLinks</Location>
```

3. Restart the Apache HTTP Server. The service name is FMSHttpd.
4. Copy a sample file from /webroot/vod to c:\hds_vod_content. This task uses sample1_1500kbps.f4v.
5. (HDS) Open *rootinstall*/samples/videoPlayer/videoplayer.html in a browser and in the Stream URL text box, enter:

   http://localhost/hds-vod/sample1_1500kbps.f4v.f4m
6. (HLS) On a device running iOS, open Safari and enter the following:

   http://localhost/hls-vod/sample1_1500kbps.f4v.m3u8

The request URL is the same as it was before the Apache configuration change, but the content is in a different location on the disk. When Apache receives a request that start with /hds-vod, it resolves the rest of the path based on the value of `HttpStreamingContentPath`.

To verify that the Apache configuration took effect, try to stream a sample file that isn't in the c:\hds_vod_content folder. For example, try to stream http://localhost/hds-vod/sample1_1000kbps.f4v.f4m. The sample video player displays an error, "We are unable to connect to the content you've requested. We apologize for the inconvenience."

Configure the location of live HDS and HLS content

To configure the location of live HDS and HLS content on disk, edit the Apache httpd.conf file. Use the following directives:

Directive	Description
`HttpStreamingContentPath`	The root location of the streams on disk (livepkgr/streams, by default). By default, the location is `"../applications"`. This location must be a subfolder of the livepkgr application. This value can be absolute or relative to the Apache root folder.
`HttpStreamingLiveEventPath`	The root location of live events on disk (livepkgr/events, by default). The default value is "../applications". This location must be a subfolder of the livepkgr application. This value can be absolute or relative to the Apache root folder. For HLS, this value is dependant on the value of the `HttpStreamingURLSandboxLevel` directive.

`HttpStreamingURLSandboxLevel`	Defines the scope at which `HttpStreamingLiveEventPath` is configured. Possible values are `"App"`, `"Inst"`, and `"Server"`. The default value is `"Server"`.
	If `HttpStreamingLiveEventPath` is configured to a particular application, use the value `"App"`. In this case, the client request URL can omit the application name, as in the following:
	`HttpStreamingLiveEventPath "..application/livepkgr"`
	`HttpStreamingURLSandboxLevel "App"`
	The request URL is:
	`http://<fms-dns-or-ip>/hls-live/<app-instance>/<event-name>/<stream-name>.m3u8`
	If `HttpStreamingLiveEventPath` is configured to an application instance, use the value `"Inst"`. In this case, the client request URL can omit the application name and the application instance, as in the following:
	`HttpStreamingLiveEventPath "..application/livepkgr/events/_definst_"`
	`HttpStreamingURLSandboxLevel "Inst"`
	The request URL is:
	`http://<fms-dns-or-ip>/hls-live/<event-name>/<stream-name>.m3u8`

Configure the location of the live content on disk

1 Open rootinstall/conf/ams.ini in a text editor. Set the `VHOST.APPSDIR` in parameter to one of the following:

- (Windows) C:/applications
- (Linux) /opt/applications

2 Restart the server.

3 Create a folder at the location in step 1 and copy the livepkgr application to it.

4 Open *rootinstall*/Apache2.2/conf/httpd.conf in a text editor.

5 To change the location of the HDS live content, edit the directives in bold:

```
<Location /hds-live>
    HttpStreamingEnabled true
#   HttpStreamingLiveEventPath "../applications"
#   HttpStreamingContentPath "../applications"
   HttpStreamingLiveEventPath "C:\applications"HttpStreamingContentPath "C:\applications"
    HttpStreamingF4MMaxAge 2
    HttpStreamingBootstrapMaxAge 2
    HttpStreamingFragMaxAge -1
    HttpStreamingDrmmetaMaxAge 3600
    Options -Indexes FollowSymLinks
</Location>
</IfModule>
```

***Note:** On Linux, use* `"/opt/applications"`.

6 To change the location of HLS live content, edit the directives in bold:

```
<IfModule hlshttp_module>
<Location /hls-live>
    HLSHttpStreamingEnabled true
#   HttpStreamingLiveEventPath "../applications"
#   HttpStreamingContentPath "../applications"
   HttpStreamingLiveEventPath "C:\applications"HttpStreamingContentPath "C:\applications"
    HLSMediaFileDuration 8000
    HLSSlidingWindowLength 6
    HLSFmsDirPath ".."
    HLSM3U8MaxAge 2
    HLSTSSegmentMaxAge -1

# Uncomment the following directives to enable encryption
# for this location.
#    HLSEncryptionScope server
#    HLSEncryptCipherKeyFile "../creds/liveeventkey.bin"
#    HLSEncryptKeyURI        "https://<ServerName>/hls-key/liveeventkey.bin"

    Options -Indexes FollowSymLinks
</Location>
```

7 Restart Apache HTTP Server. The service name is FMSHttpd.

8 Open Flash Media Live Encoder and publish a stream with the following settings:

- Video codec—H.264
- Audio codec—AAC
- Keyframe Frequency—4 seconds
- AMS URL—rtmp://localhost/livepkgr
- Stream—livestream?adbe-live-event=liveevent

9 (HDS) Open *rootinstall*/samples/videoPlayer/videoplayer.html in a browser and in the Stream URL text box, enter:

http://localhost/hds-live/livepkgr/_definst_/liveevent/livestream.f4m

10 (HLS) On a device running iOS, open Safari and enter:

http://localhost/hls-live/livepkgr/_definst_/liveevent/livestream.m3u8

***Note:** To play content in Safari 4.0 on Mac OS, create an HTML document that uses the* `<video>` *tag.*

Configure the scope of the request URL

1 Open *rootinstall*/Apache2.2/conf/httpd.conf in a text editor.

2 To change the scope of the request URL (also called the URL sandbox level) for HDS, edit the directives in bold:

```
<Location /hds-live>
    HttpStreamingEnabled true
#   HttpStreamingLiveEventPath "../applications"
#   HttpStreamingContentPath "../applications"
    HttpStreamingLiveEventPath "c:\applications\livepkgr"
    HttpStreamingContentPath "c:\applications"
    HttpStreamingURLSandboxLevel "App"
    HttpStreamingF4MMaxAge 2
    HttpStreamingBootstrapMaxAge 2
    HttpStreamingFragMaxAge -1
    Options -Indexes FollowSymLinks
</Location></IfModule>
```

To change the scope of the request URL for HLS, edit the directives in bold:

```
<IfModule hlshttp_module>
<Location /hls-live>
    HLSHttpStreamingEnabled true
#   HttpStreamingLiveEventPath "../applications"
#   HttpStreamingContentPath "../applications"
    HttpStreamingLiveEventPath "C:\applications\livepkgr"
    HttpStreamingContentPath "C:\applications"
    HttpStreamingURLSandboxLevel "App"
    HLSMediaFileDuration 8000
    HLSSlidingWindowLength 6
    HLSFmsDirPath ".."
    HLSM3U8MaxAge 2
    HLSTSSegmentMaxAge -1

# Uncomment the following directives to enable encryption
# for this location.
#    HLSEncryptionScope server
#    HLSEncryptCipherKeyFile "../creds/liveeventkey.bin"
#    HLSEncryptKeyURI        "https://<ServerName>/hls-key/liveeventkey.bin"

    Options -Indexes FollowSymLinks
</Location>
```

3 Restart the Apache HTTP Server. The service name is FMSHttpd.

4 Open Flash Media Live Encoder and publish a stream with the following settings:

- Video codec—H.264
- Audio codec—AAC
- Keyframe Frequency—4 seconds
- AMS URL—rtmp://localhost/livepkgr
- Stream—livestream?adbe-live-event=liveevent

5 (HDS) Double-click rootinstall/samples/videoPlayer/videoplayer.html to open the Adobe Media Server sample video player. Enter the following for Stream URL and click play:

http://<host>/hds-live/_definst_/liveevent/livestream.f4m

6 On a device running iOS, open Safari and enter:

http://<host>/hls-live/_definst_/liveevent/livestream.m3u8

Because the URL sandbox level is configured to "App" instead of the default "Server", the URL does not include the application name (livepkgr).

Configure set-level F4M/M3U8 files for multi-bitrate streaming

Adobe Media Server 5

Configuration information about multi-bitrate streams has been divided into multiple levels: set-level F4M and M3U8 files and stream-level F4M and M3U8 files. For HDS, the files are F4M manifest files. For HLS, the files are M3U8 variant playlists. (The documentation sometimes uses the generic term "manifest" to refer to both.)

Set-level F4M and M3U8 files Contain the URL to the stream-level manifest file and the bit rate information for each stream in a multi-bitrate set. For HDS, the set-level F4M file can also contain information about a DVR rolling window. For HLS, the set-level M3U8 file can also contain codec information.

Adobe Media Server 5 includes a Set-level F4M/M3U8 File Generator tool. This tool is installed to *rootinstall*/tools/f4mconfig/configurator/f4mconfig.html. Use the tool to generate set-level files. Copy the set-level files to a web server. The media player requests set-level files to play multi-bitrate content.

The following is a set-level F4M file:

```
<manifest xmlns="http://ns.adobe.com/f4m/2.0">
    <baseURL>http://localhost/hds-live/livepkgr/_definst_/liveevent/</baseURL>
    <media href="livestream1.f4m" bitrate="150"/>
    <media href="livestream2.f4m" bitrate="500"/>
    <media href="livestream3.f4m" bitrate="700"/>
</manifest>
```

The following is a set-level M3U8 file:

```
#EXTM3U
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=150000
http://10.0.1.11/hls-live/livepkgr/_definst_/liveevent/livestream1.m3u8
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=500000
http://10.0.1.11/hls-live/livepkgr/_definst_/liveevent/livestream2.m3u8
#EXT-X-STREAM-INF:PROGRAM-ID=1,BANDWIDTH=700000
http://10.0.1.11/hls-live/livepkgr/_definst_/liveevent/livestream3.m3u8
```

Stream-level F4M and M3U8 files contain bootstrap information and DRM metadata. The live packager and the HTTP modules generate stream-level F4M and M3U8 files in real time when a media player requests the content.

Flash/AIR media players that support set-level files To play set-level F4M files, use a media player that uses OSMF 1.6 or later. Strobe Media Playback that installs to *rootinstall*/samples/videoPlayer supports set-level F4M files.

Apple media players that support set-level files To play set-level M3U8 files, use any device that supports Apple HTTP Live Streaming. See the Apple HTTP Live Streaming documentation.

F4M specification For detailed information about the F4M file format, see the F4M File Format Specification.

M3U8 specification For detailed information about the M3U8 file format, see HTTP Live Streaming Internet-Draft.

Tutorials that use set-level files The following tutorials use set-level files for multi-bitrate streaming:

- "Publish and play live multi-bitrate streams over HTTP" on page 8
- "Play on-demand multi-bitrate media files over HTTP" on page 22

- "Publish an audio-only stream (HLS)" on page 11
- "Configure DVR (HDS)" on page 13

Configure absolute time on the server

***Note:** The livepkgr application is configured by default to use absolute time. Use this section to understand HTTP streaming and to duplicate the livepkgr application.*

The livepkgr ingest application is configured by default to use absolute time. To see the configuration, open *rootinstall*/applications/livepkgr/Application.xml in a text editor.

To publish live, multi-bitrate streams, configure the live encoder to publish the streams using absolute time. See "Stream live media (HTTP)" on page 5.

1 Copy an Application.xml file to the application folder.

   For example, if you're publishing streams to the application "livepkgr2", copy an Application.xml file from the conf_defaultRoot__defaultVHost_ folder to the applications/livepkgr2 folder.

2 Open the Application.xml file in a text editor.

3 Set the tag //Application/StreamManager/Live/AssumeAbsoluteTime to true:

```
<Application>
    ...
    <StreamManager>
        ...
        <Live>
            ...
            <AssumeAbsoluteTime>true</AssumeAbsoluteTime>
```

4 Save the file.

5 Restart the server. See Starting and stopping the server.

***Important:** Set `<AssumeAbsoluteTime>` to true only when all streams published to the application use absolute time. If this value is set to true and the application ingests a stream that does not use absolute time, you may see warnings in the server logs about time going backwards when streams are stopped and re-published.*

Configure F4F and TS fragment duration

The server records ingested (live) streams into fragments. It records on-demand files into fragments when a client requests the files.

Adobe HDS fragments are F4F files. Apple HLS fragments are TS files.

Specify the size of content fragments based on frames or based on time. The frame-based configuration overrides the time-based configuration.

Use frame-based configuration when the source media contains video encoded at a constant frame rate. Use frame-based configuration to match the fragment size to the video's keyframe interval. Use time-based configuration for media that contains audio or data but not video.

The server's fragment duration must be a multiple of the encoder's keyframe interval. The value of `KeyframeIntervalsPerFragment` defines the multiple.

Configure fragment size for live HDS and HLS

Server-level configurations for live HDS

Set the following directive in the Apache httpd.conf to configure fragment size for all live HDS events on the server:

Directive	Description	Default
`HttpStreamingFragmentDuration`	The default fragment duration, in milliseconds. This is the time-based configuration, and can be overridden by selecting frame-based values.	4000
`HttpStreamingFrameRate`	For frame-based fragment duration configuration, this is the frame rate of the source content.	None
`HttpStreamingFramesPerKeyframeInterval`	For frame-based fragment duration configuration, this is the number of frames in each keyframe interval (group of pictures).	None
`HttpStreamingKeyframeIntervalsPerFragment`	For frame-based fragment duration configuration, this is the number of keyframe intervals (GOPs) per fragment.	1

Server-level configurations for live HLS

Set the following directive in the Apache httpd.conf to configure a time-based fragment size for all live HLS events on the server:

Directive	Description	Default
`HLSMediaFileDuration`	The duration of TS files, in milliseconds. Use a value that is a multiple of the keyframe intervals for the media.	8000

Application-level and event-level configurations for live HDS

To configure fragment size at the application-level, use the Application.xml file in the directory of the Live Packager application (livepkgr, by default).

To configure fragment size at the event-level, use the Event.xml file in the live event directory (applications/livepkgr/events/_definst_/liveevent, by default).

Use the elements in the `<Recording>` container to configure how the server writes files to disk. In the Application.xml file, the elements are located at `//Application/HDS`. In the Event.xml file, the elements are located at `//Event`.

Element	Description	Default
`/Recording`	The section that configures how the file is written to disk.	None
`/Recording/FragmentDuration`	The length of each fragment, in milliseconds. Each segment can contain one or more fragments. Time-based configuration.	4000
`/Recording/FramePrecision`	The rounding precision for the fragment run table, in frame units. Frame units are derived from the specified frame rate (1/rate). Frame-based configuration.	1
`/Recording/FrameRate`	The frame rate of the original content, in frames per second (fps). The value is floating point; for NTSC, use the value 29.97. Frame-based configuration	None

Element	Description	Default
`/Recording/FramesPerKeyframeInterval`	The number of frames between each keyframe. For example, 30 fps video with a keyframe every 2 seconds contains 60 frames per keyframe interval. Frame-based configuration.	None
`/Recording/KeyframeIntervalsPerFragment`	The number of keyframe intervals per fragment. The default value is 1, which means that the fragment size is the same as the keyframe interval. Frame-based configuration.	1
`/Recording/SegmentDuration`	The length of each segment, in milliseconds. Each .f4f file contains one segment. Time-based configuration. Frame-based configurations override time-based configurations	400000

Application-level and event-level configurations for live HLS

To configure fragment size at the application-level, use the Application.xml file in the directory of the Live Packager application (livepkgr, by default).

To configure fragment size at the event-level, use the Event.xml file in the live event directory (applications/livepkgr/events/_definst_/liveevent, by default).

Use the elements in the `<Recording>` and `<HLS>` containers to configure how the server writes files to disk. In the Application.xml file, the elements are located at `//Application/HDS`. In the Event.xml file, the elements are located at `//Event/`.

Element	Description	Default
`/Recording/FrameRate`	The frame rate of the original content, in frames per second (fps). The value is floating point; for NTSC, use the value 29.97. Frame-based configuration.	None
`/Recording/FramesPerKeyframeInterval`	The number of frames between each keyframe. For example, 30 fps video with a keyframe every 2 seconds contains 60 frames per keyframe interval. Frame-based configuration.	None
`/HLS/KeyframeIntervalPerMediaFile`	The number of keyframe intervals per media file (TS). The default value is 1, which means that the fragment size is the same as the keyframe interval. Frame-based configuration.	1
`/HLS/MediaFileDuration`	The length of each media file (TS), in milliseconds. Time-based configuration. Frame-based configurations override time-based configurations	8000

Specify fragment size for live HDS and HLS based on frames

Use the following elements for HDS: `KeyframeIntervalsPerFragment`, `FrameRate`, `FramesPerKeyframeInterval`, `FramePrecision`.

Use the following elements for HLS: `FrameRate`, `FramesPerKeyframeInterval`, and `KeyframeIntervalPerMediaFile`.

```
 <Event>
    <EventID>liveevent</EventID>
    <Recording>
        <FrameRate>29.97</FrameRate>
        <FramesPerKeyframeInterval>120</FramesPerKeyframeInterval>
        <KeyframeIntervalsPerFragment>1</KeyframeIntervalsPerFragment>
        <FramePrecision>1</FramePrecision>
    </Recording>
    <HLS>
        <KeyframeIntervalPerMediaFile>3</KeyframeIntervalPerMediaFile>
    </HLS>
</Event>
```

The server calculates the HLS media file duration according to the following formula:

```
MediaFileDurationInSeconds = FramesPerKeyframeInterval * KeyframeIntervalPerMediaFile /
FrameRate
```

Specify fragment size for live HDS and HLS based on time

Use the following elements for HDS: `FragmentDuration`, `SegmentDuration`.

Use the following element for HLS: `MediaFileDuration`.

Use the following elements: `FragmentDuration`, `MediaFileDuration`.

```
<Event>
    <EventID>liveevent</EventID>
    <Recording>
        <FragmentDuration>4000</FragmentDuration>
        <SegmentDuration>400000</SegmentDuration>
    </Recording>
<HLS>
    <MediaFileDuration>8000<MediaFileDuration>
</HLS>
</Event>
```

Configure fragment size for on-demand HDS and HLS

Server-level configurations for on-demand HDS

Set the following directive in the Apache httpd.conf to configure fragment size for all on-demand HDS on the server:

Directive	Description	Default
`HttpStreamingFragmentDuration`	The default fragment duration, in milliseconds. This is the time-based configuration. Frame-based configurations override time-based configurations.	4000
`HttpStreamingFrameRate`	The frame rate of the source content. Frame-based configuration.	None
`HttpStreamingFramesPerKeyframeInterval`	The number of frames in each keyframe interval (group of pictures. Frame-based configuration.	None
`HttpStreamingKeyframeIntervalsPerFragment`	The number of keyframe intervals (GOPs) per fragment. Frame-based configuration.	1

Server-level configurations for on-demand HLS

Set the following directive in the Apache httpd.conf to configure fragment size for all on-demand HLS on the server:

Directive	Description	Default
`HLSMediaFileDuration`	The duration of TS files, in milliseconds. Use a value that is a multiple of the keyframe intervals for the media.	8000
`HttpStreamingFrameRate`	The frame rate of the source content. This is a frame-based configuration.	None
`HLSKeyframeIntervalsPerMediaFile`	The number of keyframe intervals (group of pictures) per media file. This is a frame-based configuration.	1
`HttpStreamingFramesPerKeyframeInterval`	The number of frames in each keyframe interval (group of pictures). This is a frame-based configuration.	None

Stream-level configurations for on-demand HDS

To configure a set of on-demand media files (or a single file), copy a jit.conf file to the same directory as the media files. All media files in a directory use the same jit.conf file.

Use a jit.conf file with the following elements to configure fragment size for HDS at the stream-level:

Element	Description	Default
`//manifest/hds:fragment-duration`	The fragment duration for the set of content, in seconds. The value can be fractional (for example, to specify 2002 milliseconds, use the value 2.002). Time-based configuration. Frame-based configurations override time-based configurations	None
`//manifest/hds:frame-rate`	The frame rate of the set of content. Units are frames/second, and can be fractional (i.e. 29.97 for NTSC) . Frame-based configuration.	None
`//manifest/hds:frames-per-keyframe-interval`	The number of frames per keyframe interval (GOP). Units are frames, and should be a whole integer value. Frame-based configuration.	None
`//manifest/hds:keyframe-intervals-per-fragment`	The number of keyframe intervals per fragment. Units are keyframe intervals per fragment, and should be a whole integer value. Frame-based configuration.	1

Stream-level configurations for on-demand HLS

To configure a set of on-demand media files (or a single file), copy a jit.conf file to the same directory as the media files. All media files in a directory use the same jit.conf file. Use a jit.conf file with the following elements to configure fragment size for HLS at the stream-level:

Element	Description	Default
`//manifest/hds:frame-rate`	The frame rate of the set of content. Units are frames/second, and can be fractional (i.e. 29.97 for NTSC) . Frame-based configuration.	None
`//manifest/hds:frames-per-keyframe-interval`	The number of frames per keyframe interval (GOP). Units are frames, and should be a whole integer value. Frame-based configuration.	None
`//manifest/hds:hls`	Container for Apple HTTP Live Streaming configurations.	None
`//manifest/hds:hls/hds:keyframe-intervals-per-media-file`	The number of keyframe intervals per TS file duration. The value must be a whole integer. Frame-based configuration.	None
`//manifest/hds:hls/hds:media-file-duration`	The TS file duration, in milliseconds, for the set of content associated with this file. Use a value that is a multiple of the fragment duration. Time-based configuration. Frame-based configurations override time-based configurations	None

Specify fragment size for on-demand HDS and HLS based on frames

Use the following elements for HDS: `hds:frame-rate`, `hds:frames-per-keyframe-interval`, and `hds:keyframe-intervals-per-fragment`.

Use the following elements for HLS: `hds:frame-rate`, `hds:frames-per-keyframe-interval`, and `hds:keyframe-intervals-per-media-file`.

```
<?xml version="1.0" encoding="utf-8"?>
<manifest xmlns="http://ns.adobe.com/f4m/1.0" xmlns:hds="http://ns.adobe.com/hds-
package/1.0">
    <hds:frame-rate>29.97</hds:frame-rate>
    <hds:frames-per-keyframe-interval>60</hds:frames-per-keyframe-interval>
    <hds:keyframe-intervals-per-fragment>2</hds:keyframe-intervals-per-fragment>
    <hds:hls>
        <hds:keyframe-intervals-per-media-file>2</hds:keyframe-intervals-per-media-file>
    </hds:hls>
</manifest>
```

Specify fragment size for on-demand HDS and HLS based on time

Use the following element for HDS: `hds:fragment-duration`.

Use the following element for HLS: `hds:media-file-duration.`

```
<?xml version="1.0" encoding="utf-8"?>
<manifest xmlns="http://ns.adobe.com/f4m/1.0" xmlns:hds="http://ns.adobe.com/hds-
package/1.0">
    <hds:fragment-duration>8000</hds:fragment-duration>
    <hds:hls>
        <hds:media-file-duration>8000</hds:media-file-duration>
    </hds:hls>
</manifest>
```

Known issue

Note that when the normal recording is on, the HLS module returns an HTTP 503 (Service Unavailable) error intermittenly for TS requests, if the TS duration is less than the fragment duration.

Configure the size of the IO buffer

Adobe Media Server 5

The IO buffer improves the read and write performance of the Apache f4f module. The IO buffer loads the disk file into an in-memory buffer. It reads and writes to the in-memory buffer instead of making system calls.

Note: *This feature impacts both HTTP Dynamic Streaming and HTTP Live Streaming.*

When the buffer is full, or if the read/write head needs to move beyond the buffer boundaries, the server commits the contents of the buffer to the disk file and pre-loads new content into the buffer from the disk file.

The default value of the IO buffer is 4096 bytes. To increase or decrease the value, edit the Event.xml file. Use a multiple of 4096 bytes.

1 Open the Event.xml file in a text editor.

   For live HTTP streaming, the Event.xml file is located in the live event folder. For example, rootinstall/applications/livepkgr/events/_definst_/liveevent/Event.xml.

2 Set the `IOBufferSize` to a value that is a multiple of the cluster size of the underlying operating system. `IOBufferSize` is the size of the IO buffer in bytes.

```
<Event>
    <EventID>liveevent</EventID>
        <Recording>
            <FragmentDuration>4000</FragmentDuration>
            <IOBufferSize>4096</IOBufferSize>
        </Recording>
</Event>
```

3 Save the file.

Disk management

Adobe Media Server 5

Set the `DiskManagementDuration` configuration parameter in the Application.xml file or the Event.xml file to specify how much of the latest live content is on disk. The limit set at the application level cannot be overriden at the event level. If the limit has been set in the Application.xml file, the Event.xml file can set a duration that is equal to or lower than that limit.

All live HTTP streaming uses disk management to prevent the disk from filling. Disk management allows you to create 24/7 live streaming events. In addition, when using DVR, configure disk management in relation to the DVR settings. See "Configure DVR (HDS)" on page 13.

By default, disk management keeps 3 hours of a live stream. The server constantly checks the content and if the content duration is above the value of `DiskManagmentDuration`, the server deletes the .f4f files and the .f4x files for segments beyond the limit.

If `SegmentDuration` or `DiskManagmentDuration` have values (both do by default), the following formula must be true for the server to start recording:

```
FragmentDuration < SegmentDuration < DiskManagementDuration
```

Also, the value of `SegmentDuration` must be greater than 0. (The value 0 creates 1 segment.) The value of `SegmentDuration` specifies how rapidly segments are removed. The smaller the value of `SegmentDuration`, the faster segments are deleted.

The server maintains the total duration of all the content available. After a segment is created, the server checks if the segment does not have any fragments that are in the disk management window. If so, the server removes the segment from the bootstrap entries, then removes the segment from disk.

Note: *This behavior is a change from Flash Media Server 4.0 which allowed* `FragmentDuration` *to be greater than* `SegmentDuration`*.*

Element	Default	Description
`DiskManagementDuration`	3	The maximum duration of the content on the server, in hours. The default value is 3. Use a fractional value to specify minutes.

The value of `DiskManagementDuration` in the Application.xml file cannot be overridden by the value in the Event.xml file. If a value is not set in Application.xml, you can set a value in Event.xml.

The following sets disk management duration to 4 hours in the Application.xml file. This value impacts all live events running in this application.

```
<Application>
    ...
    <HDS>
        <Recording>
            <FragmentDuration>4000</FragmentDuration>
            <SegmentDuration>16000</SegmentDuration>
            <DiskManagementDuration>4</DiskManagementDuration>
        </Recording>
    </HDS>
</Application>
```

The following sets the disk management duration to 1 hour in the Event.xml file. This value limits the content duration on the server for the event called "liveevent."

```
<Event>
    <EventID>liveevent</EventID>
    <Recording>
        ...
        <DiskManagementDuration>1</DiskManagementDuration>
    </Recording>
</Events>
```

For more information, see "Configure live HTTP streaming" on page 81.

Configure content caching (HDS)

To improve the performance of live and on-demand HTTP Dynamic Streaming, enable Apache content caching on the origin server. When caching is enabled, the server caches content that it packages for HDS. When the server receives a request, it checks the cache before it serves content. Content caching is disabled by default.

1 Run the htcacheclean tool.

To limit the amount of storage the disk cache uses, run the Apache htcacheclean tool when content caching is enabled. The tool can run manually or as a daemon. The htcacheclean tool is located at *rootinstall*/Apache2.2/bin/. For information about how to run the tool, see the Apache documentation.

2 In the Apache httpd.conf file, uncomment the caching configuration section.

```
# Uncomment this to enable caching
LoadModule cache_module modules/mod_cache.so
<IfModule mod_cache.c>
    LoadModule disk_cache_module modules/mod_disk_cache.so
    IfModule mod_disk_cache.c>
        CacheEnable disk /hds-vod
        CacheEnable disk /hls-vod
        CacheRoot cacheroot
        CacheMaxFileSize 10000000
        CacheLock On
    </IfModule>
</IfModule>
```

Verify that the directory specified by the `CacheRoot` directive exists. The default cacheroot directory is located at *rootinstall*/Apache2.2/cacheroot.

3 Restart Apache.

For more information about content caching, see the Apache documentation.

Encrypt content for Adobe Access protection

Important: *Use the HTTP Dynamic Streaming packagers to both encrypt and fragment content. Do not use the Adobe Access packaging tools to encrypt content. The HTTP Dynamic Streaming packagers cannot fragment content that has already been encrypted.*

To deliver live or on-demand content with HTTP Dynamic Streaming and protect it with Adobe Access, use Adobe Access Server for Protected Streaming. This server is a Adobe Access license server implementation optimized for use with HTTP Dynamic Streaming. See *Adobe Access Protecting Content*.

Note: *The Adobe Access SDK and the Adobe Access license server reference implementation can also issue licenses for HTTP Dynamic Streaming.*

After you've deployed Adobe Access Server for Protected Streaming, configure Adobe Media Server to package and encrypt the content in real-time.

Configure Adobe Media Server to encrypt live content

You can configure live content encryption with Adobe Access at the application level and at the event level.

To configure Adobe Access at the application level, use the Application.xml file located here:

***rootinstall*/applications/livepkgr/Application.xml**

In the Application.xml file, the `FlashAccessV2` container element is located here:

```
//Application/HDS/Recording/ContentProtection/FlashAccessV2
```

To configure Adobe Access at the event level, use the Event.xml file located here:

***rootinstall*/applications/livepkgr/events/_definst_/liveevent/Event.xml**

In the Event.xml file, the `FlashAccessV2` container element is located here:

```
//Event/Recording/ContentProtection/FlashAccessV2
```

1 Open the file *rootinstall*/applications/livepkgr/events/_definst_/liveevent/Event.xml in a text editor.

2 Add the XML tags required for encrypting the streams. The following is a sample Event.xml file:

```
<Event>
    <EventID>liveevent</EventID>
    <Recording>
        <FragmentDuration>4000</FragmentDuration>
        <SegmentDuration>10000</SegmentDuration>
         <ContentProtection enabled="true">
            <ProtectionScheme>FlashAccessV2</ProtectionScheme>
            <FlashAccessV2>
                <ContentID>foo</ContentID>
                <CommonKeyFile>common-key.bin</CommonKeyFile>
                <LicenseServerURL>http://dill.corp.adobe.com:8090</LicenseServerURL>
                <TransportCertFile>production_transport.der</TransportCertFile>
                <LicenseServerCertFile>license_server.der</LicenseServerCertFile>
                <PackagerCredentialFile>production_packager.pfx</PackagerCredentialFile>
                <PackagerCredentialPassword>hbXX5omIhzI=</PackagerCredentialPassword>
                <PolicyFile>policy01.pol</PolicyFile>
            </FlashAccessV2>
        </ContentProtection>
    </Recording>
</Event>
```

***Note:** The certificates that you use with the Live Packager must match the license server.*

Element	Description	Default value
`FlashAccessV2`	A container for settings used by Adobe Access to protect content.	None
`FlashAccessV2/CommonKeyFile`	A path to a common key file used to generate a content encryption key. The path should be absolute or relative to the configuration file. The base key used with the content ID to generate the content encryption key. This is a binary file containing a 16-byte/128-bit binary key. For adaptive bitrate streaming, use the same common key and content ID for an entire set of content. Using the same key and id allows a single license to decrypt a set of content.	None
`FlashAccessV2/ContentID`	The content ID used with the common key to generate the content encryption key.	None
`FlashAccessV2/LicenseServerCertFile`	The DER encoded license server certificate file used for content protection.	None
`FlashAccessV2/LicenseServerURL`	The URL of the license server that handles license acquisition for this content.	None

Element	Description	Default value
`FlashAccessV2/PackagerCredentialFile`	The PFX file containing the packager's protection credentials.	None
`FlashAccessV2/PackagerCredentialPassword`	The password string used to secure the packager credentials.	None
`FlashAccessV2/PolicyFile`	The file containing the policy for this content. Currently only a single policy can be applied to content packaged with this tool.	None
`FlashAccessV2/TransportCertFile`	The DER encoded transport certificate file.	None

The Live Packager outputs the fragmented and protected files needed for HTTP Dynamic Streaming. Each fragment is persistently protected in both the CDN cache and the browser cache.

Configure Adobe Media Server to encrypt on-demand content:

1 Create a file called jit.conf and copy it to the same folder as the on-demand media files. The settings in the jit.conf file apply to all the files in the folder.

2 Add the XML tags required for encrypting the streams. The following is a sample jit.conf file:

```
<?xml version="1.0" encoding="utf-8"?>
<manifest xmlns="http://ns.adobe.com/f4m/1.0" xmlns:hds="http://ns.adobe.com/hds-
package/1.0">

    <hds:frame-rate>29.97</hds:frame-rate>
    <hds:frames-per-keyframe-interval>60</hds:frames-per-keyframe-interval>
    <hds:content-protection enabled="true">
        <hds:protection-scheme>FlashAccessV2</hds:protection-scheme>
        <hds:FlashAccessV2>
            <hds:content-id>myfile.manifest</hds:content-id>
            <hds:common-key-file>common.bin</hds:common-key-file>
            <hds:license-server-url>http://mylicenseserver.myhost.com</hds:license-server-
url>
            <hds:transport-cert-file>transport.der</hds:transport-cert-file>
            <hds:license-server-cert-file>server.der</hds:license-server-cert-file>
            <hds:packager-credential-file>packager.pfx</hds:packager-credential-file>
            <hds:packager-credential-password>??????</hds:packager-credential-password>
            <hds:policy-file>policy.pol</hds:policy-file>
        </hds:FlashAccessV2>
    </hds:content-protection>
</manifest>
```

Element	Description	Default
`//manifest/hds:content-protection/hds:flash-access/hds:common-key-file`	The path to common key file. File contains 16-byte/128-bit random key. The path must be absolute or relative to the jit.conf file.	None
`//manifest/hds:content-protection/hds:flash-access/hds:content-id`	The Content ID to be used for content protection. If not specified, the salt is the filename. If specified, the salt is shared with all content in the directory.	None
`//manifest/hds:content-protection/hds:flash-access/hds:license-server-url`	The License Server URL.	None
`//manifest/hds:content-protection/hds:flash-access/hds:transport-cert-file`	The path to transport cert file. The file is in DER format. The path should be absolute or relative to the jit.conf file.	None
`//manifest/hds:content-protection/hds:flash-access/hds:license-server-cert-file`	The path to transport cert file. File is in DER format. The path should be absolute or relative to the jit.conf file.	None
`//manifest/hds:content-protection/hds:flash-access/hds:packager-credential-file`	The path to packager credential cert file. File is in PFX format. The path should be absolute or relative to the jit.conf file.	None
`//manifest/hds:content-protection/hds:flash-access/hds:packager-credential-password`	The packager credential password.	None
`//manifest/hds:content-protection/hds:flash-access/hds:policy-file`	The path to a policy file. File is in FAXS policy format. The path should be absolute or relative to the jit.conf file.	None

Publish live streams

❖ Publish a live stream or multi-bitrate streams from a supported encoder to the livepkgr service. Adobe Access doesn't require any special settings.

Play encrypted streams

❖ OSMF media players do the following:

a Request the file fragments and play them back seamlessly.

b Obtain a content license from Adobe Access Server for Protected Streaming. The content license contains the key required to play the content to legitimate clients.

c Decrypt the content in real-time.

HTTP streaming configuration file reference

Configure live and on-demand HTTP Streaming at the server level (httpd.conf)

Apache HTTP Server that installs with Adobe Media Server is configured by default for Adobe HTTP Dynamic Streaming (to Flash Player and AIR) and Apple HTTP Live Streaming (to iOS and Mac OS). The Adobe Media Server Apache installation includes four custom modules that handle HTTP streaming.

The following table describes the types of streaming that each modules handles, and where the module is configured in the httpd.conf file:

Streaming type	Module	Location directive in httpd.conf
Live Adobe HTTP Dynamic Streaming	f4fhttp_module	`<Location /hds-live>`
On-demand Adobe HTTP Dynamic Streaming packaged offline by the File Packager tool	f4fhttp_module	`<Location /vod>`
On-demand Adobe HTTP Dynamic Streaming packaged in real-time	jithttp_module	`<Location /hds-vod>`
Live Apple HTTP Live Streaming	hlshttp_module	`<Location /hls-live>`
On-demand Apple HTTP Live Streaming	hlshttp_module	`<Location /hls-vod>`
HTTP Streaming failover	ctrlplane_module	`<Location /ctrlplane`

***Note:** Despite the name, Apple HTTP Live Streaming supports live and on-demand content.*

The configuration of the modules determines the format of the URL that the player requests from the server, the location of the content, the content fragmentation settings, and additional settings. Configure the settings at the server-level in the Apache httpd.conf file:

***rootinstall*/Apache2.2/conf/httpd.conf**

To see the default configurations, open the httpd.conf file.

Directives for all the HTTP streaming modules

Use the following directives to configure all the HTTP streaming modules (jithttp_module, f4fhttp_module, and hlshttp_module):

Directive	Required	Description
`HttpStreamingContentPath`	Yes	The physical location of the content path. For live streaming, the location of livepkgr/streams. This value can be absolute or relative to the Apache root folder. See "Content storage (HDS and HLS)" on page 56.
`HttpStreamingLiveEventPath`	Yes For live streaming only.	The location of the live event directory (by default livepkgr/events). The default value is "../applications". This value can be absolute or relative to the Apache root folder. See "Content storage (HDS and HLS)" on page 56. HTTP streaming file operations are routed through the File plug-in. You can optionally use a File plug-in to manage content. For more information, see "Use the File plug-in to manage content for live HTTP streaming" on page 333.

Directive	**Required**	**Description**
`HttpStreamingURLSandboxLevel`	No For live streaming only	Defines the scope at which `HttpStreamingLiveEventPath` is configured. Possible values are `"App"`, `"Inst"`, and `"Server"`. The default value is `"Server"`.
`Location`	Yes	The section of the request URL after the server name (and optional port number). This path tells Apache which module to use to process the request.

For more information, see Configure HTTP Streaming failover.

Directives for Adobe HTTP Dynamic Streaming

Use the following directives to configure the Adobe HTTP Dynamic Streaming modules (jithttp_module for on-demand and f4fhttp_module for live):

Parameter	**Module**	**Required**	**Description**
`EncryptionScope`	jithttp_module, f4fhttp_module	No	The scope at which protected HTTP Dynamic Streaming (PHDS) is configured. Possible values are: `server`—The server uses configuration settings in the httpd.conf file. `content`—The server uses configuration settings in the Event.xml or Application.xml file (live), or in the jit.conf file (on-demand). See Protected HTTP Dynamic Streaming (PHDS).
`HttpStreamingBootstrapMaxAge`	f4fhttp_module	No	The time to live for bootstrap files, in seconds. The default value is 2.
`HttpStreamingEnabled`	f4fhttp_module	Yes	Setting `HttpStreamingEnabled` to true enables the Origin module to process requests matching the Location directive. Possible values are `true` and `false`.
`HttpStreamingFragMaxAge`	f4fhttp_module jithttp_module	No	The time to live for fragment files, in seconds. The default value is -1 which is unlimited.
`HttpStreamingDrmmetaMaxAge`	f4fhttp_module	No	The time to live for DRM MetaData, in seconds. The default value is 3600 (1 hour).
`HttpStreamingFragmentDuration`	jithttp_module	No	The length of an F4F file, in milliseconds. The default value is 4000. See Configure F4M and TS duration.
`HttpStreamingFrameRate`	jithttp_module hlshttp_module	No	For frame-based fragment duration configuration, this is the frame rate of the source content. This value is overridden if a jit.conf file is provided. See Configure F4M and TS duration.

Parameter	Module	Required	Description
`HttpStreamingFramesPerKeyframeInterval`	jithttp_module hlshttp_module	No	For frame-based fragment duration configuration, this is the number of frames in each keyframe interval (group of pictures). This value is overridden if a jit.conf file is provided. See Configure F4M and TS duration.
`HttpStreamingF4MMaxAge`	f4fhttp_module	No	The time to live for F4M files, in seconds. The default value is 2.
`HttpStreamingJITConfAllowed`	jithttp_module	No	Indicates whether jit.conf processing is allowed. The default is `true`. Setting this to `false` means that jit.conf will not be processed, even if present, and the server will not even attempt to load a jit.conf file.
`HttpStreamingJITPEnabled`	jithttp_module	Yes	Indicates whether just-in-time packaging is enabled (`true`) or not (`false`).
`HttpStreamingKeyframeIntervalsPerFragment`	jithttp_module	No	For frame-based fragment duration configuration, this is the number of keyframe intervals (GOPs) per fragment. The default value is 1. This value is overridden if a jit.conf file is provided. See Configure F4M and TS duration.
`HttpStreamingMaxFragmentDuration`	jithttp_module, f4fhttp_module	No	This is the maximum allowed size of a fragment (in milliseconds). Set this directive to a value that prevents user-supplied overrides from creating large fragments that can clog the network.
`HttpStreamingMinFragmentDuration`	jithttp_module, f4fhttp_module	No	This is the minimum allowed size of a fragment (in milliseconds).Set this directive to a value that prevents user-supplied overrides from creating many small fragments. When media is packaged into too many fragments, the client has to make too many requests.
`JitFmsDirPath`	jithttp_module	Yes	The location of the Adobe Media Server installation.
`JitGenerateCaptionInfo`	jithttp_module	No	You can configure closed captioning at the server level by providing `true` as a value for this element.
`cc-info-enabled`	jithttp_module	No	You can configure closed captioning at the server level by providing `true` as a value for this element.
`PHDSSWFVerification`	jithttp_module, f4fhttp_module	No	Enables SWF verification for PHDS (`true`) or not (`false`). If `EncryptionScope` is set to `content`, this value can be set in the jit.conf file.

Parameter	Module	Required	Description
PHDSSWFWhiteListFolder	jithttp_module, f4fhttp_module	No	A path to the folder containing the whitelist for on-demand SWF verification. The folder can contain more than one whitelist file. The default value is the content folder (which contains both media and the jit.conf file). If EncryptionScope is set to content, this value can be set in the jit.conf file..
PHDSSWFMaxTimeToVerify	jithttp_module, f4fhttp_module	No	Specifies the maximum time for SWF verification for PHDS. The default value is 100s.
ProtectionScheme	jithttp_module, f4fhttp_module	No	The type of protection to apply to content. Possible values are phds and FlashAccessV2.

For more HDS directives that can be used for content protection, see "Configuring content protection for HDS" on page 115

Directives for Apple HTTP Live Streaming

Use the following directives to configure the Apple HTTP Live Streaming module for live and on-demand streaming (hlshttp_module):

Directive	Required	Description
HLSEncryptCipherKeyFile	No	The path of the default cipher key used to encrypt the content for PHLS.
HLSEncryptionScope	No	The scope at which protected HTTP Live Streaming (PHLS) is configured. Possible values are: server—The server uses configuration settings in the httpd.conf file. content—The server uses configuration settings in the Event.xml or Application.xml file (live), or in the jit.conf file (on-demand). See Protected HTTP Live Streaming (PHLS).
HLSEncryptKeyURI	No	The URI that the client uses to fetch the encryption key for PHLS.
HLSFmsDirPath	Yes	The location of the Adobe Media Server installation. The default value is "..". This value can be absolute or relative to the Apache root folder. See "Content storage (HDS and HLS)" on page 56.
HLSHttpStreamingEnabled	Yes	Indicates whether or not HLS is enabled (true) or not (false). The default value is false.
HLSJITConfAllowed	No	Indicates whether jit.conf processing is allowed. The default is true. Setting this to false means that jit.conf will not be processed, even if present, and the server will not even attempt to load a jit.conf file.

Directive	**Required**	**Description**
`HLSMediaFileDuration`	No	The duration of the TS file to be served, in milliseconds. This number must be a multiple of the fragment duration specified in Flash Media Live Encoder. The default value is 8000. See Configure F4M and TS duration. In the case of HLS-VOD, `HLSMediaFileDuration` can be in between 500 ms to 30000ms. These bound values can be modified using the following directives: • `HLSMaxMediaFileDuration` • `HLSMinMediaFileDuration`
`HLSM3U8MaxAge`	No	Specifies the max-age to set in the cache-control header for M3U8 responses, in seconds. The default value for HLS Live is 2. The default value for HLS on-demand is 86400 (1 day). The value -1 does not set a cache-control header.
`HLSSlidingWindowLength`	No	The number of TS files in a playlist and available for seeking within a sliding window. The time within the window is the value of `HLSSlidingWindowLength * HLSMediaFileDuration`. HTTP Live Streaming clients use the sliding window to configure the seek bar. The default value is 6. See Configure a sliding window (HLS).
`HLSTSSegmentMaxAge`	No	Specifes the max-age to set in the cache-control header for TS segment responses, in seconds. The default value for HLS Live is -1 which doesn't set a cache-control header. The default value for HLS on-demand is 86400 (1 day).
`HttpStreamingUnavailableResponseCode`	No	The HTTP response code that FMS returns for unavailable fragments (end-of-program, gaps, etc.) The default is 503.
`HLSMaxEventAge`	No	The maximum number of seconds FMS allows the bootstrap to age before marking it as stale. If the bootstrap age is greater than this value, the stream is considered to be down. The default is 300.
`HLSAddPadPacketsForCont`	No	Specifies whether or not to allow adding padding TS packets to maintain continuity across media files.
`HLSAllowExtractAudioOnly`	No	Specifies whether or not to allow audio-only extraction from audio-video sources.
`HLSEncryptKeyURIPrefix`	No	A text string prefixed to the event ID to generate the encryption key.
`HLSEncryptKeyURISuffix`	No	A string suffixed to the event ID to generate the encryption key.
`HLSFAXSUseUniqueContentID`	No	Set the value to true to use a unique content ID.
`HLSKeyframeIntervalsPerMediaFile`	No	Specifies the total number of keyframes per HLS media file in the VOD content.
`HLSHttpUnavailableMaxAge`	No	Specifies the maximum duration (in seconds) to set the Cache-Control header for unavailable fragment responses.
`HttpStreamingFrameRate`	No	Specifies the frame rate in the VOD content.
`HttpStreamingFramesPerKeyframeInterval`	No	Specifies the number of frames per KFI in the VOD content.

For more HLS directives that can be used for content protection, see "Configuring content protection for HLS" on page 157.

Directives for HTTP Streaming failover

Use the following directives to configure HTTP Streaming failover (ctrlplane_module):

Directive	Required	Description
`HdsHttpStreamingLiveEventPath`	Yes	The physical location of the root of the event path for HDS. This can be relative to the Apache installation root. For example, "C:\Program Files\Adobe\Adobe Media Server 5\applications"or "..\applications."
`HlsHttpStreamingLiveEventPath`	Yes	The physical location of the root of the event path for HLS. This can be relative to the Apache installation root. For example, "C:\Program Files\Adobe\Adobe Media Server 5\applications"or "..\applications."
`MaxBootstrapAge`	No	The maximum number of seconds the Control Plane module lets the bootstrap age before marking it as stale. If the bootstrap age is greater than this value, the stream is considered to be down and the Control Plane module sets the `up` status to false. The default is 300.
`HttpStreamingURLSandboxLevel`	No	The Level at which `HttpStreamingLiveEventPath` sandboxes the access. You should set this to match corresponding settings in HDS and HLS configurations. Valid values include App, Application, Inst, Instance, and Server. The default is Server. For example, if `HdsHttpStreamingLiveEventPath` is set to `../applications/livepkgr/events/_definst_`, set `HttpStreamingURLSandboxLevel` to Inst. if `HdsHttpStreamingLiveEventPath` is set to `../applications/livepkgr/`, set `HttpStreamingURLSandboxLevel` to App. For information on similar HDS and HLS settings, see "Configure live and on-demand HTTP Streaming at the server level (httpd.conf)" on page 76

Configure live HTTP streaming

You can configure live HTTP Dynamic Streaming and live HTTP Live Streaming at the following levels:

Level	Configuration file
Server	*rootinstall*/Apache2.2/conf/httpd.conf
Application	*rootinstall*/applications/livepkgr/Application.xml
Event	*rootinstall*/applications/livepkgr/events/_definst_/liveevent/Event.xml

Configure live HTTP streaming at the application level (Application.xml)

To configure HTTP streaming for all live events in an application, use the Application.xml file. The Application.xml file overrides settings in the httpd.conf file for a single application.

Place the Application.xml file in the application folder. For example, *rootinstall*/applications/livepkgr/Application.xml.

Note: *Configuring HTTP streaming in the Application.xml file at the vhost level is not supported.*

The following is an Application.xml file with all possible configuration elements for HTTP streaming:

```
<Application>
    <StreamManager>
        <Live>
            <AssumeAbsoluteTime></AssumeAbsoluteTime>
        </Live>
    </StreamManager>
    <HDS>
        <HLS>
            <KeyframeIntervalPerMediaFile></KeyframeIntervalPerMediaFile>
            <MediaFileDuration></MediaFileDuration>
            <SlidingWindowLength></SlidingWindowLength>
            <Encryption enabled="true">
                <KeyFile></KeyFile>
                <KeyURI></KeyURI>
            </Encryption>
        </HLS>
        <Recording>
            <FragmentDuration></FragmentDuration>
            <SegmentDuration></SegmentDuration>
            <FrameRate></FrameRate>
            <FramesPerKeyframeInterval></FramesPerKeyframeInterval>
            <KeyframeIntervalsPerFragment></KeyframeIntervalPerFragment>
            <FramePrecision></FramePrecision>
            <DiskManagementDuration></DiskManagementDuration>
             <ContentProtection enabled="true">
                <ProtectionScheme></ProtectionScheme>
                <PHDS>
                    <CommonKeyFile></CommonKeyFile>
                    <VideoEncryptionLevel></VideoEncryptionLevel>
                    <UpdateInterval></UpdateInterval>
                    <SWFVerification enabled="true">
                        <WhiteListFolder></WhiteListFolder>
                        <UpdateInterval></UpdateInterval>
                    </SWFVerfication>
                </PHDS>
                <FlashAccessV2>
                    <ContentID></ContentID>
                    <CommonKeyFile></CommonKeyFile>
                    <LicenseServerURL></LicenseServerURL>
                    <TransportCertFile></TransportCertFile>
                    <LicenseServerCertFile></LicenseServerCertFile>
                    <PackagerCredentialFile></PackagerCredentialFile>
                    <PackagerCredentialPassword></PackagerCredentialPassword>
                    <PolicyFile></PolicyFile>
                </FlashAccessV2>
            </ContentProtection>
        </Recording>
    </HDS>
</Application>
```

For information about each element, see "Application.xml and Event.xml" on page 84.

Configure live HTTP streaming at the event level (Event.xml)

To configure a single event, use an Event.xml file. Copy the Event.xml file to a live event folder. The default live event is *rootinstall*/applications/livepkgr/events/_definst_/liveevent. The Event.xml file overrides the Application.xml file and the Apache httpd.conf file for a single live event. Every live event folder must have a single Event.xml file.

Note: *There are a few exceptions in which the Event.xml configuration does not override the Application.xml configuration. These exceptions are noted with the configuration parameter.*

The following is an Event.xml file with all possible configuration elements for HTTP streaming:

```
<Event>
    <EventID></EventID>
    <HLS>
        <KeyframeIntervalPerMediaFile></KeyframeIntervalPerMediaFile>
        <MediaFileDuration></MediaFileDuration>
        <SlidingWindowLength></SlidingWindowLength>
        <Encryption enabled="true">
            <KeyFile></KeyFile>
            <KeyURI></KeyURI>
        </Encryption>
    </HLS>
    <Recording>
        <FragmentDuration></FragmentDuration>
        <SegmentDuration></SegmentDuration>
        <FrameRate></FrameRate>
        <FramesPerKeyframeInterval></FramesPerKeyframeInterval>
        <KeyframeIntervalsPerFragment></KeyframeIntervalPerFragment>
        <FramePrecision></FramePrecision>
        <DiskManagementDuration></DiskManagementDuration>
        <ContentProtection enabled="true">
            <ProtectionScheme></ProtectionScheme>
            <PHDS>
                <CommonKeyFile></CommonKeyFile>
                <VideoEncryptionLevel></VideoEncryptionLevel>
                <PlaybackExpiration></PlaybackExpiration>
                <UpdateInterval></UpdateInterval>
                <SWFVerification enabled="true">
                    <WhiteListFolder></WhiteListFolder>
                    <UpdateInterval></UpdateInterval>
                </SWFVerfication>
            </PHDS>
            <FlashAccessV2>
                <ContentID></ContentID>
                <CommonKeyFile></CommonKeyFile>
                <LicenseServerURL></LicenseServerURL>
                <TransportCertFile></TransportCertFile>
                <LicenseServerCertFile></LicenseServerCertFile>
                <PackagerCredentialFile></PackagerCredentialFile>
                <PackagerCredentialPassword></PackagerCredentialPassword>
                <PolicyFile></PolicyFile>
            </FlashAccessV2>
        </ContentProtection>
    </Recording>
</Event>
```

Application.xml and Event.xml

Root elements

Element	Description	Default
//Application	The root element for an Application.xml configuration file.	N/A
//Event	The root element for an Event.xml configuration file.	N/A
//Event/EventID	The name of a live event. This name is the same as the name of the event folder and the value of the Server-Side ActionScript Stream.liveEvent property.	liveevent

Configurations for HTTP Live Streaming

Use the elements in the `<HLS>` container to configure HTTP Live Streaming. In the Application.xml file, the elements are located at `//Application/HDS/HLS`. In the Event.xml file, the elements are located at `//Event/HLS`.

Element	Description	Default
/HLS/KeyframeIntervalPerMediaFile	Sets frame-based file duration. See "Configure F4F and TS fragment duration" on page 64.	None
/HLS/MediaFileDuration	The duration of the TS file to be served, in milliseconds. This number must be a multiple of the FragmentDuration. See "Configure F4F and TS fragment duration" on page 64.	None
/HLS/SlidingWindowLength	The seekable portion of the stream for Apple HTTP Live Streaming. The time within the window is the value of SlidingWindowLength * MediaFileDuration. HTTP Live Streaming clients use the sliding window to configure the seek bar. The sliding window is relative to the current position of the live stream. See "Configure a sliding window (HLS)" on page 14.	6
/HLS/Encryption	Set the enabled attribute to "allow" to allow PHLS configurations in the Event.xml file to override settings in the Application.xml file. Set the enabled attribute to "true" to configure PHLS in the Application.xml file. These configurations apply to all live events in the application.	None
/HLS/Encryption/KeyFile	The path of the default cipher key used to encrypt the content.	None
/HLS/Encryption/KeyURI	The URI that the client uses to fetch the decryption key.	None

Configurations for recording

Use the elements in the `<Recording>` container to configure how the server writes files to disk. In the Application.xml file, the elements are located at `//Application/HDS/Recording`. In the Event.xml file, the elements are located at `//Event/Recording`.

Element	Description	Default
`/Recording`	The section that configures how the file is written to disk.	None
`/Recording/DiskManagementDuration`	The maximum duration of live content on disk, in hours. Use a fractional value to specify minutes. When this value is set in Application.xml, it cannot be overridden in Event.xml. See "Disk management" on page 70.	3
`/Recording/FragmentDuration`	The length of each fragment, in milliseconds. Each segment can contain one or more fragments. See "Configure F4F and TS fragment duration" on page 64.	4000
`/Recording/FramePrecision`	The rounding precision for the fragment run table, in frame units. Frame units are derived from the specified frame rate (1/rate). See "Configure F4F and TS fragment duration" on page 64.	1
`/Recording/FrameRate`	The frame rate of the original content, in frames per second (fps). The value is floating point; for NTSC, use the value 29.97. See "Configure F4F and TS fragment duration" on page 64.	None
`/Recording/FramesPerKeyframeInterval`	The number of frames between each keyframe. For example, 30 fps video with a keyframe every 2 seconds contains 60 frames per keyframe interval. See "Configure F4F and TS fragment duration" on page 64.	None
`/Recording/IOBufferSize`	The size of the IO buffer for the recording, in bytes. The IO buffer loads the disk file into a buffer in memory. It reads and writes to the buffer in the memory instead of making system calls. See "Configure the size of the IO buffer" on page 70.	4096
`/Recording/KeyframeIntervalsPerFragment`	The number of keyframe intervals per fragment. The default value is 1, which means that the fragment size is the same as the keyframe interval. See "Configure F4F and TS fragment duration" on page 64.	1
`/Recording/SegmentDuration`	The length of each segment, in milliseconds. Each .f4f file contains one segment. See "Configure F4F and TS fragment duration" on page 64.	400000

Configurations for content protection

Use the `ContentProtection` and `ProtectionScheme` elements to enable content protection with Protected HTTP Dynamic Streaming or Adobe Access.

Element	Description	Default
`/ContentProtection`	Whether to enable content for protection with Protected HTTP Dynamic Streaming or Adobe Access. To enable content protection, set the `enabled` attribute to `true`.	None
`/ContentProtection/ProtectionScheme`	The type of protection. Possible values are `phds` and `FlashAccessV2`.	None

Configurations for Protected HTTP Dynamic Streaming (PHDS)

Use the elements in the PHDS container to configure Protected HTTP Dynamic Streaming. In the Application.xml file, the elements are located at //Application/HDS/Recording/ContentProtection/PHDS. In the Event.xml file, the elements are located at //Event/Recording/ContentProtection/PHDS.

Configurations for Adobe Access

Use the elements in the FlashAccessV2 container to configure content protection with Adobe Access. In the Application.xml file, the elements are located at //Application/HDS/Recording/ContentProtection/FlashAccessV2. In the Event.xml file, the elements are located at //Event/Recording/ContentProtection/FlashAccessV2.

For information about the elements in the FlashAccessV2 container, see "Encrypt content for Adobe Access protection" on page 72.

Manifest.xml

Adobe Media Server 5 introduces *set-level manifest files* that replace Manifest.xml files. However, Adobe Media Server 5 supports Manifest.xml files and, in some cases, you may want to use them.

For example, Flash Media Playback does not support set-level manifest files. To use Flash Media Playback for DVR or multi-bitrate streaming, configure a Manifest.xml file.

The following is the default Manifest.xml file with an added <dvrInfo> element:

```
<manifest xmlns="http://ns.adobe.com/f4m/1.0">
   <dvrInfo beginOffset="0"></dvrInfo>
   <media streamId="livestream1" bitrate="100">
   </media>
   <media streamId="livestream2" bitrate="200">
   </media>
   <media streamId="livestream3" bitrate="350">
   </media>
</manifest>
```

Element	Attribute	Description
dvrInfo		
	beginOffset	An offset, in seconds, from the beginning of the recorded stream. Clients can begin viewing the stream at this location. The default value is 0.
	endOffset	An offset, in seconds, before the current duration of the recorded stream. Clients cannot view the stream before this location. The default value is 0. Negative values are treated as 0. If neither endOffset nor beginOffset is set, the start time is the beginning of the content.
media		Represents one stream.
	streamID	The name of the publishing stream.
	bitrate	The bitrate at which the stream was encoded.

Element	Attribute	Description
`bestEffortFetchInfo`		Parent element for best effort fetchconfiguration. If thiselement is present, AMS enables best effort fetch. The `bestEffortFetchInfo` element is expected to be a part of F4M 2.0. To specify your Manifest.xml as F4M 2.0, you must change the `xmlns` attribute to "http://ns.adobe.com/f4m/2.0". AMS currently honors this syntax. However, since the official F4M specification 2.0 is not yet complete, this element may change in future releases.
	maxForwardFetches	Maximum total number of forward fetches that the OSMF player performs when it encounters a liveness or dropout error that is encountered. If the forward fetches fail, the OSMF player reverts to non-best-effort behavior. The default is 2.
	maxBackwardFetches	Maximum number of consecutive failed backward fetches that OSMF player performs before reverting to non-best-effort behavior. A value of 0 indicates that no backward fetches will be performed. The default is 2.
	fragmentDuration	Corresponds to the fragment interval (in seconds) that is currently in use on the packaging server. You can specify up to 3 decimal points. Ensure that this value matches the configured `FragmentInterval` in your Event.xml file. Differences of as little as a millisecond can result in incorrect best effort fetch behavior. This attribute is required.
	segmentDuration	Corresponds to the segment interval (in seconds) that is currently in use on the packaging server. You can specify up to 3 decimal points. Ensure that this value matches the configured `SegmentInterval` in your Event.xml file. Differences of as little as a millisecond can result in incorrect best effort fetch behavior. This attribute is required. For more on `fragmentDuration` and `segmentDuration`, see "Configure live HTTP streaming" on page 81.

***Note:** Manifest.xml files support F4M v1.0 only (if not using best-effort fetch).*

***Note:** The `bestEffortFetchInfo` element is expected to be a part of F4M 2.0. To use best-effort fetch, specify your Manifest.xml as F4M 2.0 by changing the xmlns attribute to "http://ns.adobe.com/f4m/2.0". AMS currently honors this syntax. However, since the official F4M specification 2.0 is not yet complete, this element may change in future releases.*

Specify the `bestEffortFetchInfo` element in the set-level manifest. AMS ignores `bestEffortFetchInfo` elements specified in the stream-level manifest.

Configure on-demand HTTP streaming

You can configure on-demand HTTP Dynamic Streaming and HTTP Live Streaming at the following levels:

Level	Configuration file
Server	*rootinstall*/Apache2.2/conf/httpd.conf
Stream	jit.conf

Configure on-demand HTTP streaming at the stream level (jit.conf)

Configure stream-level configuration live in a jit.conf file. Create a jit.conf file and copy it to the same directory as the on-demand stream or streams. A directory can contain only 1 jit.conf file. The settings in the file apply to all content in the directory.

The following is a jit.conf file with all possible configurations:

```
<?xml version="1.0" encoding="utf-8"?>
<manifest xmlns="http://ns.adobe.com/f4m/1.0" xmlns:hds="http://ns.adobe.com/hds-
package/1.0">

    <hds:frame-rate></hds:frame-rate>
    <hds:frames-per-keyframe-interval></hds:frames-per-keyframe-interval>
    <hds:fragment-duration></hds:fragment-duration>
    <hds:keyframe-intervals-per-fragment><hds:keyframe-intervals-per-fragment>

    <hds:hls>
        <hds:keyframe-intervals-per-media-file></hds:keyframe-intervals-per-media-file>
        <hds:media-file-duration></hds:media-file-duration>
        <hds:encryption enabled="true">
            <hds:keyfile></hds:keyfile>
            <hds:keyuri></hds:keyuri>
        </hds:encryption>
    </hds:hls>

    <hds:content-protection enabled="true">
        <hds:protection-scheme>phds</hds:protection-scheme>
            <hds:phds>
                <hds:common-key-file></hds:common-key-file>
                <hds:video-encryption-level></hds:video-encryption-level>
                <hds:playback-expiration></hds:playback-expiration>
            </hds:phds>
            <hds:FlashAccessV2>
                <hds:content-id></hds:content-id>
                <hds:common-key-file></hds:common-key-file>
                <hds:license-server-url></hds:license-server-url>
                <hds:transport-cert-file></hds:transport-cert-file>
                <hds:license-server-cert-file></hds:license-server-cert-file>
                <hds:packager-credential-file></hds:packager-credential-file>
                <hds:packager-credential-password></hds:packager-credential-password>
                <hds:policy-file></hds:policy-file>
        </hds:FlashAccessV2>
    </hds:content-protection>

</manifest>
```

Configurations for recording to disk

For more information about elements in this table, see "Configure F4F and TS fragment duration" on page 64.

Element	**Description**	**Default**
`//manifest/hds:fragment-duration`	Use this element to configure fragment duration by time. The fragment duration for the set of content, in seconds. The value can be fractional (for example, to specify 2002 milliseconds, use the value 2.002). If this value is not specified, the value is taken from the module configurations in httpd.conf. See "Configure F4F and TS fragment duration" on page 64.	None
`//manifest/hds:frame-rate`	Use this element to configure fragment duration by frame rate. The frame rate that determines the fragment duration for the set of content. Units are frames/second, and can be fractional (i.e. 29.97 for NTSC) . See "Configure F4F and TS fragment duration" on page 64.	None
`//manifest/hds:frames-per-keyframe-interval`	Use this element to configure fragment duration by frame rate. The number of frames per keyframe interval (GOP). Units are frames, and should be a whole integer value. See "Configure F4F and TS fragment duration" on page 64.	None
`//manifest/hds:keyframe-intervals-per-fragment`	Use this element to configure fragment duration by frame rate. The number of keyframe intervals per fragment. Units are keyframe intervals per fragment, and should be a whole integer value. See "Configure F4F and TS fragment duration" on page 64.	1
`//manifest/hds:hls`	Container for Apple HTTP Live Streaming configurations.	N/A
`//manifest/hds:hls/hds:keyframe-intervals-per-media-file`	Use this element to configure a frame-based file duration. The number of keyframe intervals per TS file duration. The value must be a whole integer. See "Configure F4F and TS fragment duration" on page 64.	None
`//manifest/hds:hls/hds:media-file-duration`	Use this element to configure a time-based file duration. The TS file duration, in milliseconds, for the set of content associated with this file. Use a value that is a multiple of the fragment duration. See "Configure F4F and TS fragment duration" on page 64.	None

Configurations for content protection

Use the `ContentProtection` and `ProtectionScheme` elements to enable content protection with Protected HTTP Dynamic Streaming or Adobe Access.

Element	Description	Default
`//manifest/hds:content-protection`	Whether to enable content for protection with Protected HTTP Dynamic Streaming or Adobe Access. To enable content protection, set the `enabled` attribute to `"true"`.	None
`//manifest/hds:content-protection/hds:protection-scheme`	The type of protection. Possible values are `phds` and `FlashAccessV2`.	None

Configurations for protected HTTP Dynamic Streaming (PHDS)

Use the elements in the `<hds:phds>` container to configure content protection with PHLS.

Configurations for protected HTTP Live Streaming (PHLS)

Use the elements in the `<hds:hls>` container to configure content protection with PHLS.

Configurations for Adobe Access

Use the elements in the `<hds:FlashAccessV2>` container to configure content protection with Adobe Access.

For more information, see "Encrypt content for Adobe Access protection" on page 72.

Build custom media players

Build media players for the live and vod services

Adobe has developed the Open Source Media Framework (OSMF) to make it easier to build media players. OSMF is an open software framework for building robust, feature rich video players and applications based on the Flash Platform. For information about building a media player with the OSMF library, see Building streaming video players in Flash with the Adobe Open Source Media Framework.

You can also use ActionScript to build a media player from scratch. For more information, see the following:

- "Developing streaming media applications" on page 188
- "Build an HTTP Dynamic Streaming media player" on page 94

Connect to a streaming service

Like all Adobe Media Server applications, streaming services expect a `NetConnection.connect()` URI in the following format:

```
rtmp://ams-ip-or-dna/serviceName/[formatType:][instanceName/]fileOrStreamName
```

`hostName` The Adobe Media Server domain name.

`serviceName` Either `live` or `vod`.

`instanceName` If the client is connecting to the default instance, you can either omit the instance name or use `_definst_`. If the client is connecting to an instance you have created, such as *room1*, use that name.

`formatType` For mp3 files, `mp3:`. For MP4/F4V files, `mp4:`. Not required for FLV files.

`fileOrStreamName` Either a filename (for example, my_video.mp4) or a path (for example, subdir/subdir2/my_video.mp4), for example, `rtmp://www.examplemediaserver.com/vod/mp4:ClassicFilms/AnOldMovie.mp4`. For MPEG-4-based files, if the file on the server uses a filename extension (.mp4, .f4v, and so on), specify it. If the stream is live and the publisher specified a filename extension, specify it. For instance, `rtmp://www.examplemediaserver.com/live/livestream`.

Unsupported ActionScript APIs

Clients for the vod and live services can use any ActionScript APIs except remote shared objects (`SharedObject.getRemote()`).

You cannot edit the server-side code for streaming services. However, the services do have a custom API that lets you access information from the server. Call the `NetConnection.call()` method from client-side code and pass it the name of the API you want to call.

More Help topics

"Streaming services API" on page 93

Allow connections from specific domains

By default, clients can connect to the live and vod services from any domain. You can limit the domains from which clients can connect.

❖ Navigate to the *rootinstall*/applications/live or *rootinstall*/applications/vod folder and do one of the following:

- To add a domain for SWF clients, edit the allowedSWFdomains.txt file.
- To add a domain for HTML clients, edit the allowedHTMLdomains.txt file.

The TXT files contain detailed information about adding domains.

Access raw audio and video data

***Note:** Adobe Media Server Standard supports this feature through the Access plug-in only.*

Beginning with Flash Media Server 3 and Flash Player 9.0.115, you can access raw audio and video data in streams. Use this data to create snapshots in your applications. To access the data, call the ActionScript 3.0 `BitmapData.draw()` and `SoundMixer.computeSpectrum()` methods. For more information, see ActionScript 3.0 Reference for Flash Platform.

By default, Adobe Media Server prevents you from accessing streams. To allow stream access, do the following:

1 Move the main.far file from *rootinstall*/applications/live or *rootinstall*/applications/vod to *rootinstall*/samples/applications/live or *rootinstall*/samples/applications/vod.

You cannot edit FAR files, so you must replace main.far with main.asc.

2 Copy the main.asc file from *rootinstall*/samples/applications/live or *rootinstall*/samples/applications/vod to *rootinstall*/applications/live or *rootinstall*/applications/vod.

3 Open the main.asc file in a text editor.

4 Uncomment the following code to allow all clients to access all streams:

```
//p_client.audioSampleAccess = "/";
//p_client.videoSampleAccess = "/";
```

5 Save the main.asc file.

***Note:** The application instance requires restarting in order for the changes in the main.asc to take affect.*

Streaming services API

getStreamLength()

`getStreamLength(streamObj)`

Returns the length of a stream, in seconds. Call this method from a client-side script and specify a response object to receive the returned value.

Availability

Flash Media Server 3, vod streaming service

Parameters

`streamObj` A Stream object.

Returns

A number.

Example

The following client-side code gets the length of the `sample_video` stream and returns the value to `returnObj`:

```
nc.call("getStreamLength", returnObj, "sample_video");
```

getPageUrl()

`getPageUrl()`

Returns the URL of the web page in which the client SWF file is embedded. If the SWF file isn't embedded in a web page, the value is the location of the SWF file. The following code shows the two examples:

```
// trace.swf file is embedded in trace.html.
getPageUrl returns: http://www.example.com/trace.html

// trace.swf is not embedded in an HTML file.
getPageUrl returns: http://www.example.com/trace.swf
```

The value must be an HTTP address. For security reasons, local file address (for example, file:///C:/Adobe Media Server applications/example.html) are not displayed.

Availability

Flash Media Server 3, vod streaming service, live streaming service

Example

The following example calls the `getPageUrl()` method on the server:

```
nc.call("getPageUrl", returnObj);
```

getReferrer()

`getReferrer()`

Returns the URL of the SWF file or the server where the connection originated.

Availability

Flash Media Server 3, vod streaming service, live streaming service

Example

The following code calls the `getReferrer()` method on the server:

```
myNetConnection.call("getReferrer", returnObj);
```

Build an HTTP Dynamic Streaming media player

Use Open Source Media Framework (OSMF)

You don't need to build an HTTP Dynamic Streaming media player—Strobe Media Playback and Flash Media Playback support HDS by default.

If you do decide to build a custom player, Adobe strongly recommends using OSMF. OSMF is a robust framework designed to deliver high-quality video. Use the OSMF Sample Player for HTTP Dynamic Streaming as a reference implementation. The OSMF Sample Player uses the ActionScript 3.0 `NetStream.appendBytes()` API to deliver bytes to Flash Player and AIR.

Understanding the application flow for live HTTP Dynamic Streaming

To play a stream, a media player requests a .f4m file. The player contains logic that requests additional files when a user seeks, pauses, and plays media. The Adobe Media Server Live Packager (livepkgr application) creates these files when the server ingests a live stream or when a client requests an on-demand media file.

The HTTP Origin Module generates the .f4m file when it is requested. The file is not physically present on the disk. However, you can request the file even when a live stream has stopped publishing.

Streaming media using Adobe HDS requires Adobe Media Manifest files (.f4m).

The manifest file contains information about a media asset or information about each stream in a multiple stream event. This information can include the location of the media, DRM additional header data, media bootstrap information, adaptive streaming bitrates, and so on.

The Apache HTTP modules create the F4M and M3U8 files as the streams are packaged and written to disk. To create these files, the HTTP modules use metadata from the Event.xml file (if it exists) and data from the .f4f, .f4x, .meta, .bootstrap, .stream, and .drmmeta files.

Important: *Do not move the files from their original locations.*

To generate the file, the HTTP modules follow these steps:

1. Combine the request URL and the `HttpStreamingLiveEventPath` directive to locate the live event.
2. Retrieve metadata about the event from the Event.xml file and the multi-level manifest file.
3. Scan the event directory for stream record files (.stream).
4. Retrieve the path to the corresponding content from each .stream file. Each .stream file becomes a `<media>` element in the manifest file.
5. Retrieve metadata from the .meta file.
6. Create links to the bootstrap information and DRM additional header data (if the content is protected).
7. Return the generated manifest document (.f4m or .m3u8).

For more information, see the F4M File Format Specification at adobe.com.

For more information, sabout the M3U8 file, see the the HTTP Live Streaming Internet-Draft.

In the *rootinstall*\applications\livepkgr\streams_definst_ folder, the Live Packager creates a folder with the name of each stream: livestream1, livestream2, and livestream3. The Live Packager creates the following files in each folder:

- livestream#.bootstrap
- livestream#.control
- livestream#.meta
- livestream#Seg#.f4f
- livestream#Seg#.f4x

These are the files that an HTTP Dynamic Streaming media player requests to handle pausing, fast forwarding, rewinding, and so on. The following table describes each file type:

File	**Description**
livestream#Seg#.f4f	A segment. The Live Packager outputs one or more F4F files. Each file contains a segment of the source file. Each segment contains the fragmented (and optionally protected) content.
livestream#Seg#.f4x	An index file listing the fragment offsets in each .f4f file. The Live Packager outputs one or more F4X files. The HTTP Origin Module uses this file to deliver fragments.
livestream#.meta	Contains the stream metadata (bitrate, screen size, and so on).
livestream#.bootstrap	Contains the bootstrap information for the stream.
livestream#.control	Contains internal metadata that the Live Packager uses to manage stream state.
livestream#.drmmeta	Contains additional header information when a stream is encrypted for use with Adobe Access.

The following are the URL request formats for these files when Apache is configured for HTTP Dynamic Streaming:

Request type	**URL Form**
Fragment	http://<host>/<location-tag-alias>/streams/<app-name>/streams/<app-instance>/<stream name>Seg<segment #>-Frag<fragment #> **Note:***You cannot request an F4F file directly. Request a Fragment.*
Bootstrap (.bootstrap)	http://<host>/<location-tag-alias>/streams/<app-name>/streams/<app-instance>/<stream name>.bootstrap

For more information about configuring Apache for HTTP Dynamic Streaming, see Apache configurations for HTTP Dynamic Streaming.

Stream record files

A stream record file (.stream) is an XML document that contains the physical location of the stream. The server creates the stream record file when an incoming stream is associated with a live event. The server creates the file with an encoded name in the following location:

applications/*appname*/events/*appinstancename*/*eventname*/*MTg1ODAyNjgwNg=*.stream

The HTTP Origin Module reads the stream record file to locate the content for the stream. The following is the format of the .stream file:

```
<?xml version="1.0" encoding="UTF-8"?>
<stream xmlns="http://www.adobe.com/liveevent/1.0">
    <type>
        f4f
    </type>
    <name>
        livestream
    </name>
    <path>
        C:\Program Files\Adobe\Adobe Media Server 
5\applications\myapp\streams\_definst_\livestream
    </path>
</stream>
```

Sample .f4m manifest file

The player uses the manifest file to request a content fragment. In the following example, there are two streams associated with the live event, "livestream" and "livestream1". The `<media>` elements provide the absolute URL path to the content location with the prefix "/live/streams".

If .bootstrap and .drmmeta are found in the same location as the .f4f file, `<bootstrap>` and `<drmAdditionalHeader>` elements are included in the manifest file. The .drmmeta file can be shared across multiple streams, therefore there is only one `<drmAdditionalHeader>` element in the sample manifest file. Both myStream and myStream1 refer to the same .drmmeta file through the `drmAdditionalHeaderId` attribute.

The `metadata` element contains the metadata for one piece of media in Base64 encoding. The metadata is the same information dispatched in the ActionScript `NetStream.onMetaData()` event.

```
<?xml version="1.0" encoding="UTF-8" ?>
<manifest xmlns="http://ns.adobe.com/f4m/1.0">
    <id>live_mbr_event</id>
    <streamType>live</streamType>
    <duration>0</duration>
    <bootstrapInfo
        profile="named"
        url="/live/streams/myStream.bootstrap"
        id="bootstrap2267" />
    <drmAdditionalHeader
        url="/live/streams/myStream.drmmeta"
        drmContentId="live_mbr_event"
        id="drmMetadata9996" />
    <media>
        url="/live/streams/myStream"
        bitrate="408"
        bootstrapInfoId="bootstrap2267"
        drmAdditionalHeaderId="drmMetadata9996">
        <metadata>
AgAKb25NZXRhRGF0YQgAAAAAAAhkdXJhdGlvbgBARo9cKPXCjwAFd2lkdGgAQJQAAAAAAAAABmhlaWdodABAho
AAAAAAAAAMdmlkZW9jb2RlY2lkAgAEYXZjMQAMYXVkaW9jb2RlY2lkAgAEbXA0YQAKYXZjcHJvZmlsZQBAWQAA
AAAAAAAIYXZjbGV2ZWwAQEAAAAAAAAAAAABmFhY2FvdAAAAAAAAAAAAAAAOdmlkZW9mcmFtZXJhdGUAQD34U+JVay
gAD2F1ZGlvc2FtcGxlcmF0ZQBA53AAAAAAAAANYXVkaW9jaGFubmVscwBAAAAAAAAAAAAJdHJhY2tpbmZvCgAA
AAIDAAZsZW5ndGgAQTSinwAAAAAACXRpbWVzY2FsZQBA3UwAAAAAAAAIbGFuZ3VhZ2UCAANlbmcAAAkDAAZsZW
5ndGgAQUCGAAAAAAAACXRpbWVzY2FsZQBA53AAAAAAAAAIbGFuZ3VhZ2UCAANlbmcAAAkAAAk=
        </metadata>
    </media>
    <bootstrapInfo
        profile="named"
        url="/live/streams/myStream1.bootstrap"
        id="bootstrap7975" />
    <media>
        url="/live/streams/myStream1"
        bitrate="1108"
        bootstrapInfoId="bootstrap7975"
        drmAdditionalHeaderId="drmMetadata9996">
        <metadata>
AgAKb25NZXRhRGF0YQgAAAAAAAhkdXJhdGlvbgBARo9cKPXCjwAFd2lkdGgAQJQAAAAAAAAABmhlaWdodABAho
AAAAAAAAAMdmlkZW9jb2RlY2lkAgAEYXZjMQAMYXVkaW9jb2RlY2lkAgAEbXA0YQAKYXZjcHJvZmlsZQBAWQAA
AAAAAAAIYXZjbGV2ZWwAQEAAAAAAAAAAABmFhY2FvdAAAAAAAAAAAAAAOdmlkZW9mcmFtZXJhdGUAQD34U+JVay
gAD2F1ZGlvc2FtcGxlcmF0ZQBA53AAAAAAAAAANYXVkaW9jaGFubmVscwBAAAAAAAAAAAAAJdHJhY2tpbmZvCgAA
AAIDAAZsZW5ndGgAQTSinwAAAAAACXRpbWVzY2FsZQBA3UwAAAAAAAAAIbGFuZ3VhZ2UCAANlbmcAAAkDAAZsZW
5ndGgAQUCGAAAAAAAACXRpbWVzY2FsZQBA53AAAAAAAAAAIbGFuZ3VhZ2UCAANlbmcAAAkAAAk=
        </metadata>
    </media>
</manifest>
```

Offline packaging

Offline packaging for HDS

The File Packager is a command-line tool that packages on-demand media files into a format that can stream over HTTP. The File Packager can also encrypt files for protection with Adobe Access.

***Note:** To package on-demand content for HTTP streaming in real-time when a client requests the content, see Stream on-demand media (HTTP) in the Adobe Media Server Developer's Guide.*

Packaging on-demand media

File Packager location

The File Packager is available from the following locations:

- Adobe Media Server *rootinstall*/tools/f4fpackager folder

The File Packager requires several libraries to run. These libraries are located in the f4fpackager folder. To move the File Packager, move the f4fpackager folder and all its contents.

Input file formats and output file formats

To stream on-demand media over HTTP, process the file with the File Packager. The File Packager parses the content, translates it into segments and fragments, and writes the segments and fragments to the F4F format. You can also choose to have the File Packager encrypt the content for use with Adobe Access.

The File Packager supports the following file formats:

- F4V/MP4 compatible files
- FLV

Important: *The file formats can contain any codec the format supports, except Speex audio. You can package a file that includes Speex audio, the File Packager does not throw an error, but the audio will not play. The packager can package video-only and audio-only streams.*

The File Packager outputs the following files:

File type	Description
.f4f	A segment. The tool outputs one or more F4F files. Each file contains a segment of the source file. Each segment contains one or more fragments of content. A player can use a URL to address each fragment.
.f4m	Flash Media Manifest file. Contains information about codec, resolution, and the availability of multi-bitrate files.
.f4x	Index file. Contains the location of specific fragments within a stream.
.bootstrap	Bootstrap file. Contains the bootstrap information for each segment of the file. The tool creates an external .bootstrap file only if you specify the `--external-bootstrap` option. If you don't specify that option, the bootstrap information is included in the manifest file.
.drmmeta	DRM Additional Header file. Contains additional header information about an encrypted file. The tool creates this file only if you specify the `--external-bootstrap` option and encrypt the file. Otherwise, the information is included in the manifest file.

Segments And Fragments

An F4F file contains one segment of a media file. Each segment can contain multiple fragments. Use the `--segment-duration` and `--fragment-duration` options to set the length of each segment and fragment, in seconds. Specifying a segment-duration greater than 0 will create multiple files. A good size fragment is small enough to send over the internet and long enough to prevent multiple calls to Apache HTTP Server.

Run The File Packager

To package a file, pass the File Packager the name of the input file.

There are two ways to specify the name:

- Enter the `--input-file` option at the command line.
- Create a configuration file that specifies the input file and enter the `--conf-file` option at the command line.

Package content on Windows

1 Open a Command Window.

2 Change directories to the directory containing f4fpackager.exe.

3 Enter the name of the tool and any options:

```
f4fpackager --input-file=sample.f4v --output-path=c:\sampleoutput
```

The following files are output: sampleSeg1.f4f, sample.f4x, and sample.f4m.

Package content on Linux

1 Open a terminal window.

2 On a clean installation of Linux, set the `LD_LIBRARY_PATH` to the directory containing the File Packager libraries.

3 At the shell prompt, enter the name of the tool and any options:

```
f4fpackager --input-file=sample.f4v --output-path=/sampleoutput
```

The following files are output: sampleSeg1.f4f, sample.f4x, and sample.f4m.

Packaging multi-bitrate content

To stream multi-bitrate content, encode a piece of media at multiple bitrates, creating multiple files. Use the File Packager to process each media file individually. The media files share a manifest file that lists information about each media file. The OSMF Player uses this information for playback.

About packaging multi-bitrate files

Each time you run the File Packager, it creates a manifest file with the filename of the input file. When you pass the `--manifest-file` option, the File Packager adds the information from the manifest file you specify to the manifest file it creates. For example, suppose you run the following:

```
--input-file=sample1.f4v
```

The tool creates the files: sample1Seg1.f4f, sample1.f4x, and sample1.f4m.

Suppose you then run the following:

```
--input-file=sample2.f4v --manifest-file=sample1.f4m
```

The tool creates the files sample2Seg1.f4f, sample2.f4x, and sample2.f4m. The manifest file sample2.f4m contains information about sample2.f4v and sample1.f4v.

To create a manifest file for multi-bitrate content, pass the `--manifest-file` option and specify the manifest file of the previously processed file. Package the files from lowest bitrate to highest bitrate. Do not pass the `--manifest-file` option for the first file you process.

Package multi-bitrate files

Note: *This example uses Windows. For Linux commands, see "Package content on Linux" on page 98.*

1. Encode a media file at several bitrates, for example, 150 kbps, 700 kbps, and 1500 kbps.

   This example uses three files: sample1_150kbps.f4v, sample1_700kbps.f4v, and sample1_1500kbps.f4v. On Adobe Media Server, these files are installed to *rootinstall*\applications\vod\media.

2. Copy the files to the same folder as the File Packager.

   On Adobe Media Server 5, the File Packager is located in the *rootinstall*\tools\f4fpackager folder.

3. Open a Command Window.

4. Change directories to the directory containing f4fpackager.exe.

5. Enter the following:

   ```
    f4fpackager --input-file=sample1_150kbps.f4v --bitrate=150
   ```

   The following files are output: sample1_150kbpsSeg1.f4f, sample1_150kbps.f4x, and sample1_150kbps.f4m.

6. Enter the following to process the second file:

   ```
    f4fpackager --input-file=sample1_700kbps.f4v --manifest-file=sample1_150kbps.f4m --
   bitrate=700
   ```

   The following files are output: sample1_700kbpsSeg1.f4f, sample1_700kbps.f4x, and sample1_700kbps.f4m. The manifest file sample1_700kbps.f4m includes information from the input file and information from the sample1_150kbps.f4m manifest file.

7. Enter the following to process the third file:

   ```
    f4fpackager --input-file=sample1_1500kbps.f4v --manifest-file=sample1_700kbps.f4m --
   bitrate=1500
   ```

The following files are output: sample1_1500kbpsSeg1.f4f, sample1_1500kbps.f4x, and sample1_1500kbps.f4m. The manifest file sample1_1500kbps.f4m includes information from the input file and information from the sample1_700kbps.f4m manifest file. The sample1_700kbps.f4m file contains information about the sample1_150kbps.f4v file and the sample1_700kbps.f4v file.

8 To play the file, pass the URL of the sample1_1500kbps.f4m file to OSMF Player. Because this manifest file was created last, it contains information about all three files.

Packaging and encrypting content

Use the File Packager to both package and encrypt content for use with Adobe Access.

Adobe Access includes a license server implementation based on the Adobe Access SDK called Adobe® Access™ Server for Protected Streaming. Adobe Access Server for Protected Streaming is designed for use with HTTP Dynamic Streaming. For more information about this Adobe Access, see Adobe Access documentation.

Generate a common key

To use the File Packager to encrypt content, you must generate a common key file. The common key file must contain 128 bytes of randomly generated data. The File Packager uses this key with the content ID to generate the content encryption key.

Note: *Although the base key is 128 bytes, only the first 128 bits (16 bytes) are used to generate the key.*

You can use the open source tool Open SSL or Adobe's Scramble utility to generate random data and output a common key file. Download Open SSL from www.openssl.org.

Generate random data and output a common key file:

```
openssl rand -out commonKey.bin 128
```

Pass commonKey.bin as the `--common-key` command line option or in the `<common-key>` element of a configuration file.

Use a configuration file to package and encrypt content

Many command-line options are required to encrypt a file. It's easier to use a configuration file to set options than it is to set options at the command line.

1 Open the file f4fpackager_config.xml in a text editor.

By default, the tool looks for f4fpackager_config.xml in the same directory as the File Packager. However, you can move it to any directory and give it any name.

2 Enter values for the following required options (sample values are included here):

```
<offline>
    <input-file>someFile.f4v</input-file>
    <content-id>contentId</content-id>
    <common-key>commonKey.bin</common-key>
    <license-server-url>http://server1.com:9999</license-server-url>
    <license-server-cert>licenseServer.der</license-server-cert>
    <transport-cert>transportCert.der</transport-cert>
    <packager-credential>packagerCredential.pfx</packager-credential>
    <credential-pwd>mYpwd</credential-pwd>
    <policy-file>policyFile.pol</policy-file>
</offline>
```

You can add additional elements to change their default values.

3 Save the configuration file.

4 Open a Command Window.

5 Change directories to the directory containing f4fpackager.exe.

6 Enter the following:

```
f4fpackager --conf-file=f4fpackager_config.xml
```

Use the command line to package and encrypt content

1 Open a Command Window.

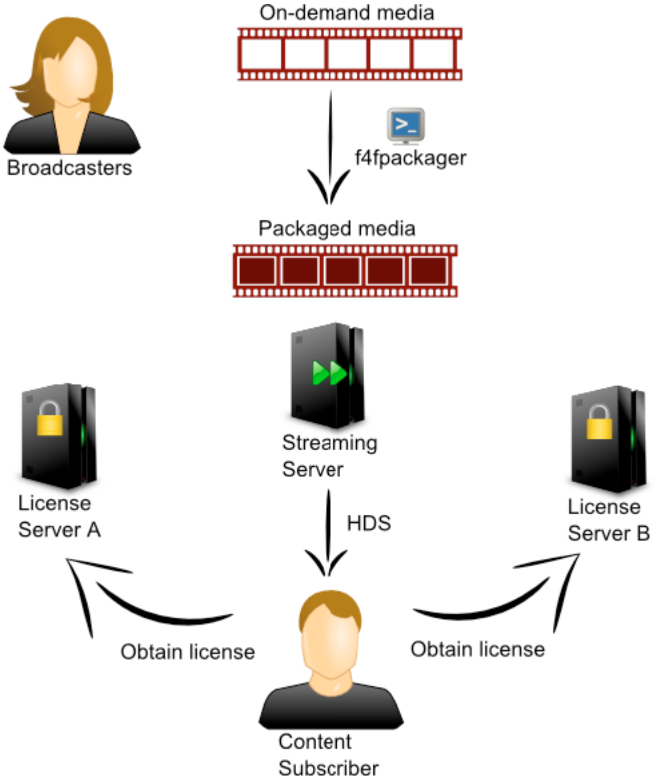
2 Change directories to the directory containing the File Packager.

3 Enter the following:

```
f4fpackager --input-file=someFile.f4v --license-server-url=http://server1.com:9999 --
transport-cert=transportCert.der --license-server-cert=licenseServer.der --packager-
credential=packagerCredential.pfx --credential-pwd=mYpwd --policy-file=policyFile.pol --
common-key=commonKey.bin --content-id=contentId
```

Updating DRM information

Broadcasters who use different license servers for testing and production, package the content individually for each license server causing bottlenecks during testing and publishing. The `f4fpackager` enhancements enable broadcasters to package content only once using multiple Adobe Access license servers.



f4packager package-once model

The packager enhancements include updates to various DRM parameters, such as license server URL, license server certificate, and usage policies using a new command-line option.

You can also use the `update-drm-info` option to generate a new manifest file with updated DRM metadata from the source manifest file. When you pass this option to the packager, repackaging of the original content is not performed. The remaining contents of the generated manifest file are similar to those of the source manifest file.

Update DRM information using the command line

To generate multiple manifest files that contain different license server URLs embedded in the DRM metadata, follow this workflow:

1 Generate the manifest file with single-bitrate or multi-bitrate encrypted streams, using the `f4fpackager` as explained in Package on-demand media offline and stream over HTTP.

2 Generate a manifest file with modified license server URL by running the `f4fpackager` again with the following parameters:

   a Specify the no value parameter `--update-drm-info`.

   b Specify the manifest file generated in Step 1 through the `--manifest-file` option.

   c Specify the encryption parameters with the same values as specified in Step 1, except for the `--license-server-url` parameter, which contains the new value.

   d (Optional) Specify the `--output-manifest-file` option. If you use the packager to update the DRM information without specifying the `--output-manifest-file` option, a name from the input manifest file is generated with a number. For example, <*input-manifest*>_<*n*>.f4m, where *n* denotes a natural number.

   If a file with the generated name exists in the output directory, an incremental number is used in the file name. The `--output-manifest-file` option works while performing the packaging operation. In this case, the manifest file is created with the given name. Other assets (.f4f, .booststrap, and .drmmeta) are created using the base name (without extension) of this parameter. Any existing files are overwritten without any warning.

3 Repeat Step 2 for all the new manifest files to be generated with the various license server URLs.

Update DRM information using configuration parameters

You can use a configuration file to package and encrypt content. The following snippet shows the sample values for the `f4fpackager_config.xml` file:

```
<offline>
    <input-file>someFile.f4v</input-file>
    <output-manifest-file>sample.f4m</
    output-manifest-file>
    <update-drm-info>true<update-drm-info>
    <content-id>contentId</content-id>
    <common-key>commonKey.bin</common-key>
    <license-server-url>http://server1.com:9999</license-server-url>
    <license-server-cert>licenseServer.der</license-server-cert>
    <transport-cert>transportCert.der</transport-cert>
    <packager-credential>packagerCredential.pfx</packager-credential>
    <credential-pwd>mYpwd</credential-pwd>
    <policy-file>policyFile.pol</policy-file>
</offline>
```

Note: *For an MBR manifest file, the DRM metadata of the streams are updated with the encryption parameter values specified in Step 2c. You cannot use different DRM metadata for individual bitrate streams of an MBR manifest file.*

You can use the previously mentioned workflow to update encryption parameters in the DRM metadata of the source manifest file (including license server URL, license server certificate, and usage policies). You cannot use this method to update encryption parameters, such as content ID and common key. You shouldn't update content ID and CEK in the DRM metadata because they are unique.

See the Package on-demand media offline and stream over HTTP for other packaging information.

File Packager reference

Command-line options to fragment files

You can use the command line to set all the options for the File Packager. Options set at the command line override any options set in the configuration file. The only required option is `--input-file`.

Option	Description	Required	Default value
--allowed-drift	As the tool rounds off durations, it drifts from the real time. This option specifies how far the tool can vary from the real time, in seconds.	No	1
--bitrate	Specifies the bitrate of the content to fragment and adds it to the manifest file. Use this option to serve files encoded at multiple bitrates. Use this option or manually edit the manifest file to add the bitrate.	No	0
--config-dump	Dumps the configuration settings. This option does not fragment the file.	No	None
--conf-file	The configuration file that contains settings for the packaging process. You can specify an absolute path or a path relative to the directory containing the tool. You can use the filename f4fpackager_config.xml or any other legal filename. Suppose the tool is in the c:\media directory and the configuration file is in the c:\media\test directory. Run the following from the command line: `f4fpackager --conf-file=test\f4fpackager_config.xml`.	No	f4fpackager_config.xml If the f4fpackager_config.xml file is not in the current directory, the default value is "none".
--duration-precision	How precise the fragment durations are, in seconds. When using a time-based fragment duration, duration precision controls how much rounding is applied to the fragment durations in the fragment run tables. This prevents small variations in the fragment durations from causing the run tables to grow too large.	No	0.1
--edit-list	If you specify this option, the File Packager follows the edit list in the input file. By default, the packager does not follow the edit list.	No	None
--external-bootstrap	Outputs the bootstrap box to a .bootstrap file. If you omit this option, the information in the bootstrap box is placed in the manifest file (F4M). Outputs DRM additional header information to a .drmmeta file when a file is encrypted. If you omit this option, the additional header information is included in the manifest file.	No	None

Option	Description	Required	Default value
--fragment-duration	The target length of each fragment, in seconds. The actual length of each fragment depends on properties of the input file, such as the keyframe interval.	No	4
--frame-precision	This option is the frame-based equivalent to the --duration-precision option. This value is the number of frames to round to in the fragment run tables.	No	1
--frame-rate	The frame rate of the original content, in frames per second (fps). The value is floating point; for NTSC, use the value 29.97.	No	
--frames-per-keyframe-interval	The number of frames between each keyframe. For example, 30 fps video with a keyframe every 2 seconds contains 60 frames per keyframe interval.	No	
--input-file	The path to the source container file. The path can be absolute or relative to the directory containing the File Packager. If you specify the `--conf-file` option and the configuration file contains an <input-file> value, you do not need to use the `--input-file` option.	Yes	None
--inspect-bootstrap	Inspects the bootstrap file. If a .bootstrap file is provided as an input, it inspects the file. If not, it inspects the bootstrap box in an F4F file.	No	None
--inspect-fragments	Inspects the F4F fragments in an F4F file.	No	None
--inspect-index	Inspects the AFRA index file if an AFRA file is provided.	No	None
--keyframe-intervals-per-fragment	The number of keyframe intervals per fragment. The default value is 1 which sets the fragment size to the same size as the keyframe interval.	No	1
--manifest-file	Updates an existing manifest file. Use this option to package a file encoded at multiple bitrates for dynamic streaming. For example, you could package three files (sample_500kpbs.f4v, sample_1000kbps.f4v, and sample_1500kbps.f4v) and add them all to the same manifest file. If you omit this option, the tool creates a Flash Media Manifest file (F4M).	No	None
--output-manifest-file	Creates a manifest file with the name provided.	No	None
--output-path	The path to the directory to which the packaged files are output. If the directory doesn't exist, the tool creates it. The path can be absolute or relative to the directory containing the File Packager.	No	Current directory
--sample-dump	Dumps the samples and the fragments they're set to. This option does not fragment the file.	No	None
--segment-duration	The length, in seconds, of a segment. The default value is 0 which creates 1 segment.	No	0

Setting fragment duration based on frames or time

The File Packager has options that determine the size of fragments. There is a frame-based option, `--keyframe-interval-per-fragment`, and a time-based option, `--fragment-duration`. Each option has additional options that refine how the tool creates fragments.

Use the frame-based option when the source media contains video encoded at a constant frame rate. The frame-based option lets you carefully match the fragment size to the video's keyframe interval. Use the time-based option for media that contains audio or data but not video.

Frame-based options: `--keyframe-interval-per-fragment`, `--frames-per-keyframe-interval`, `--frame-rate`, `--frame-precision`. Time-based options: `--fragment-duration`, `--duration-precision`. The frame-based option overrides the time-based option. If you don't provide a `--keyframe-interval-per-fragment` option, the File Packager uses the `--fragment-duration` option. If you don't provide values for `--keyframe-interval-per-fragment` or `--fragment-duration`, the File Packager uses the default fragment duration, which is 4 seconds.

Command-line options to encrypt files

Use the File Packager to encrypt files for use with Adobe Access. Some of the following options require digital certificates. To obtain digital certificates, see the Adobe Access Certificate Enrollment Site.

Option	Description	Required	Default
--transport-cert	The DER encoded transport certificate file.	Yes	None
--license-server-url	The URL of the license server that handles license acquisition for this content.	Yes	None
--license-server-cert	The DER encoded license server certificate file used for content protection.	Yes	None
--packager-credential	The PFX file containing the packager's protection credentials.	Yes	None
--credential-pwd	The password string used to secure the packager credentials.	Yes	None
--common-key	A file containing the common key. The common key is used with the content ID to generate the content encryption key. The common key must be randomly generated and 128 bytes. Although the common key is 128 bytes, only the first 128 bits (16 bytes) are used to generate the key. For dynamic streaming, use the same common key and content ID for an entire set of content. Using the same key and id allows a single license to decrypt a set of content.	Yes	None
--content-id	The content ID used with the common key to generate the content encryption key.	Yes	None
--policy-file	The file containing the policy for this content. Currently only a single policy can be applied to content packaged with this tool.	Yes	None
--encrypt-audio	Specifies whether to encrypt the audio in the file.	No	true
--encrypt-video	Specifies whether to encrypt the video in the file.	No	true

Option	Description	Required	Default
--encrypt-data	Specifies whether to encrypt the data in the file. This option supports both FLV and F4V/MP4 formats.	No	true
--ms-unencrypted	The number of milliseconds at the beginning of the content that remains unencrypted. Use this option to improve stream start times because it allows the client to acquire the license asynchronously as playback begins.	No	0
--video-encrypt-level	The level of encryption for H.264 content. Possible values are 0 (low), 1(medium), and 2 (high). The default value is 2. **Note:** *Using this option on non-H.264 content won't generate an error, but it won't change the encryption level*. The values "0" and "1" mean "partial encryption"; only important samples like video keframes are encrypted. This setting creates fewer frames to decrypt. Use this setting to improve playback performance on the client. The value "2", encrypts all video samples (after --ms-unencrypted, if specified)..	No	2

File Packager configuration file

The File Packager uses an XML configuration file to read values for command-line options. The default configuration file, f4fpackager_config.xml, is located in the same directory as the File Packager. It is often easier to specify options in a text file that you can reuse than it is to enter options at the command line.

Options you set at the command line override options set in the configuration file. This feature lets you create a configuration file for a set of content but edit a few settings for individual files.

Note: *You do not need to use the configuration file to run the tool.*

Enter the `--conf-file` option at the command line to pass the name and location of a configuration file. If you don't enter the `--conf-file` option, the tool checks for a file called f4fpackager_config.xml file in the same directory as the tool.

You can enter all the options for packaging and encrypting, including the input file, in the configuration file.

The following is the format of the configuration file:

```
<offline>
    <input-file>/media/input-file.f4v</input-file>
    <output-path>/media/http</output-path>
    <fragment-duration></fragment-duration>
   ... You can set as many options as you want ...
</offline>
```

For options that take a path, you can specify absolute paths or paths relative to the directory containing the tool.

Errors

The File Packager outputs the following errors:

Error code	Error message	Description
1	File not found	The input file, the manifest file or the configuration file could not be found.
2	Input File not given	No input file has been specified. To run the File Packager, you must specify a path to a supported file type.
3	Invalid input file	The input file, manifest file, or configuration file is an invalid input file.
4	Can't access file	The input file, manifest file, or configuration file cannot be accessed.
5	Unsupported codec	The input file contains unsupported codecs.
6	No supported tracks	The input file doesn't contain any supported tracks.
7	Write error	A write error occurred. Verify that the input file is structured correctly and that there is enough memory to run the File Packager.
8	XML library error	An error occur ed when parsing the configuration file. Verify that the XML is structured correctly.
9	Program Options error	The program options library failed. Verify that the command line arguments are structured correctly.
10	Offline Encryption info error	The File Packager doesn't have enough information to encrypt the file. Verify that there aren't any missing inputs.
11	Encryption library error	The encryption library threw an error while encrypting the content.
12	Standard error	A Standard Exception has occur ed. Verify that there is enough memory to run the File Packager.
13	Invalid Bootstrap error	The bootstrap file is corrupt and cannot be parsed.
14	FLV Error	The FLV file cannot be read.
999	Internal error	There has been an internal error in the File Packager.
1000	Invalid Input file warning	Invalid input file warning.
2000	Invalid Error code	The error code is invalid.

Offline packaging for HLS

The Adobe Media Server installation includes a command line tool for packaging on-demand media files into fragmented video content that can be streamed over HTTP. This offline packager can also encrypt files for protection with Adobe Access.

The `hlspackager` is the command line tool that takes F4V source and produces fragmented HLS VOD content. You can pass various options to the packager either through the command line or through a configuration file. However, options passed through the command line will override the corresponding options specified in the configuration file.

***Note:** The packager can recognize only F4Vand MP4 files.*

The packager tool generates several types of output files during the packaging process. They are:

- m3u8 file (single-stream playlist)
- TS segments listed in the m3u8 file

The following command shows the syntax of `hlspackager`:

```
hlspackager --input-file=<input> --output-path=<output> --base-url=<baseurl> --audio-only=<bool> --config-dump --help --media-target-duration=<fragmentLength> --conf-file=<confFile>
```

Note: *The HLS Offline Packager does not prefix the names of the assets generated by the stream or filename. Hence, it will overwrite any files in the output directory that has the same file name as the generated content. You must use the output-path option to specify different directory paths for generated content for MBR and batch job use cases.*

The various supported options are described in the following table:

Option	**Description**	**Default**	**Required**
`--input-file`	Path to the source file.	None	Yes
`--output-path`	Path for the HLS files to be created.	Current directory	No
`--base-url`	Base URL to append in the TS file names in the m3u8 file.	No base URL, relative to the m3u8 file	No
`--media-target-duration`	Target duration for each TS segment.	8000 ms	No
`--conf-file`	Specifies the configuration file that contains additional settings for the packaging process.	hlspackager_config .xml in working directory	No
`--config-dump`	Output the finally applied configuration. In this case, packaging does not happen.	No output by default	No
`--audio-only`	Whether to extract audio only stream ("`true`"/"`false`") .	false	No
`--protection-scheme`	Encryption scheme to be used ("`Vanilla`" or "`FlashAccessV4`").	None (No encryption)	Yes
`--transport-cert`	Specifies the DER-encoded transport certificate file to be used for content protection.	None	Yes, if scheme is `FlashAccessV4`
`--license-server-url`	The URL of the license server that will handle license acquisition for this content.	None	Yes, if scheme is `FlashAccessV4`
`--license-server-cert`	Specifies the DER-encoded license server certificate file to be used for content protection.	None	Yes, if scheme is `FlashAccessV4`
`--packager-credential`	Specifies the PFX file containing the packager's protection credentials.	None	Yes, if scheme is `FlashAccessV4`
`--credential-pwd`	Specifies the password string used to secure the packager credentials.	None	Yes, if scheme is `FlashAccessV4`

Option	Description	Default	Required
`--common-key-file`	The base key used (along with the content ID) to generate the final content encryption key. Using the same common key and content ID for an entire set of content allows a single license to decrypt a set on content, which is useful for multi-bitrate use cases. To generate the common key, see "Hosting and key generation" on page 110	None	Yes, if scheme is `FlashAccessV4`
`--content-id`	The content ID for this content. Used along with the common key to generate the final content encryption key.	input-file-name	Optional (only for the `FlashAccessV4` scheme)
`--policy-file`	Specifies the file containing the policy to be used for this content. Currently only a single policy can be applied to content packaged with this tool.	None	Yes, if scheme is `FlashAccessV4`
`--key-rotation-interval`	The Key rotation interval for the "`FlashAccessV4`" protection scheme. Specifies after how many segments, keys should be rotated. 0 , -1, or invalid values mean key rotation will be disabled.	No key rotation	Optional (only for the `FlashAccessV4` scheme)
`--embed-leaf-license`	Whether to embed the leaf license or not.	false	Optional (only for the `FlashAccessV4` scheme)
`--license-server-cred-file`	License server credential used when protecting content for the specified location.	None	Yes, if `embed-leaf-license` is set to true (only for the `FlashAccessV4` scheme)
`--license-server-cred-pwd`	License server credential password for the configured license server credential file.	None	Yes, if `embed-leaf-license` is set to true (only for the `FlashAccessV4` scheme)
`--key-file-path`	Key file path for "`Vanilla`" encryption (Base-64 encoded key). To generate the common key, see "Hosting and key generation" on page 110	None	Yes, if scheme is `Vanilla`
`--key-file-url`	Key file URL for "`Vanilla`" encryption (plain 16-bit key only) . To generate the common key, see "Hosting and key generation" on page 110	None	Yes, if scheme is `Vanilla`

Option	Description	Default	Required
`--faxskey-server-url`	The URL of the Adobe Access key Server that will handle serving keys to the iOS clients.	None	Yes, if scheme is `FlashAccessV4`
`--key-server-cert-file`	Key server certificate file for the remote key serving.	None	Yes, if scheme is `FlashAccessV4`

Instead of passing the options along with the command, you can also pass a configuration file containing the options to the packager. The configuration file is an XML file that can be used to specify settings for the packaging process.

The packager checks whether a configuration file has been passed by the user. If not, it checks for the `hlspackager_config.xml` file in the same directory as the command-line program. The configuration file is not needed to run the packager. The format of the configuration file is:

```
<offline>
    <input-file>/media/input-file.f4v</input-file>
    <output-path>/media/http</output-path>
    <media-target-duration></media-target-duration>

    ... other options...
</offline>
```

Hosting and key generation

You can host the fragmented content directly from the server's *webroot* directory. For instance, if you keep the fragmented content inside the *webroot*/myOfflineContent/ directory, you can playback the content through the URL - `http://server-name/myOfflineContent/prog_index.m3u8.`

Key generation for Vanilla content

The content fragmentation for the Vanilla encryption protection scheme will take base-64 encoded key as input. To generate the base-64 encoded key, run the OpenSSL command as shown below:

```
openssl rand -out keyfile.bin -base64 16
```

The generated `keyfile.bin` file has the base-64 key and can be used for content fragmentation. To generate the key for hosting, run:

```
openssl base64 -d -in keyfile.bin -out plain-keyfile.bin
```

The `plain-keyfile.bin` file can be hosted for key delivery.

Key generation for Adobe Access content

To generate the base-64 encoded key, run the OpenSSL command as shown below:

```
openssl rand -out keyfile.bin -base64 16
```

Error codes

The following table describes the various error codes raised by the packager:

Error Code	Description	Example
1	Input file is not found or invalid	Wrong or invalid input file path is specified.
2	Input file parameter is not mentioned	`--input-file` parameter is missing.

Error Code	Description	Example
3	Input file is not valid or corrupt	Corrupted mp4 file.
4	Cannot access the input file	Unable to open or access the input file .
7	Write error to disk	Unable to write m3u8 or ts segments to disk.
8	Unable to parse input xml file	Parsing issue.
9	Invalid program options	Parsing issues in the command line errors.
10	Invalid encryption parameters	Wrong license file or invalid key file.
12	System standard input or output errors	Inputs casting issue.
16	m3u8 generation error	Not able to generate m3u8 files.
17	ts file generation issues	Not able to generate the ts file.
18	Invalid target duration	Target duration is too less or too large.
19	Protection scheme is not valid	If a scheme other than "FlashAccessV4" or "Vanilla" is specified,
999	Internal processing error	-

Licensing configurations and restrictions (HDS and HLS)

You can add a license path by adding the `license-path` element. This element must contain the path to the license directory in Adobe Media Server. If the value is not explicitly specified, `'../../'` will be assumed as the license path considering that the offline packagers for both HDS and HLS are in the tools directoy.The path specified by the `license-path` element in the configuration file can be absolute or relative to the directory containing the offline packagers. For instance:

```
<license-path>C:\Program Files\Adobe\Adobe Media Server 5\</license-path>
```

***Note:** The license configurations and restrictions are applicable for both `f4fpackager` and `hlspackager`.*

Troubleshoot issues with streaming media

Troubleshoot live streaming (HTTP)

1 Use the Services Control Panel applet (Windows) or the service window (Linux) to verify that the Adobe Media Server (AMS), Adobe Media Administration Server, and FMSHttpd services are running.

2 Verify that the request URL is correct. See "URLs for publishing and playing live streams over HTTP" on page 12.

3 Verify that the server is listening to the port to which the client is trying to connect. The *rootinstall*/logs/edge.xx.log file shows on which ports the server is listening.

   By default, Adobe Media Server listens on port 80 and proxies HTTP requests to Apache HTTP Server on port 8134. Proxying traffic can cause issues with HTTP streaming. If Adobe Media Server is listening on port 80, use port 8134 in the request URL. For example, http://ams.example.com:8134/hds-live/livepkgr/_definst_/liveevent/livestream.f4m.

   Otherwise, configure Apache HTTP Server to listen on port 80 and configure Adobe Media Server not to listen on port 80. See Configure ports for HTTP streaming.

4 To allow a Flash player hosted on another web server to access content from the Adobe Media Server web server, copy a crossdomain.xml file to the *rootinstall*/webroot directory. The crossdomain.xml file grants a web client permission to handle data across multiple domains. For more information, see Cross-domain policy file specification.

5 In Flash Media Live Encoder, select the Encoding Options tab, choose Output from the Panel options menu, and verify the following:

- The value of AMS URL is rtmp://*ams-dns-or-ip*/livepkgr. If you're testing on the same server as Adobe Media Server, you can use the value **localhost** for *ams-dns-or-ip*.
- For a single stream, the value of Stream is **livestream?adbe-live-event=liveevent**.
- For adaptive bitrate streaming, the value of Stream is **livestream%i?adbe-live-event=liveevent**.

  Flash Media Live Encoder uses this value to create unique stream names. To use another encoder, provide your own unique stream names, for example, **livestream1?adbe-live-event=liveevent, livestream2?adbe-live-event=liveevent**.
- Check the logs for errors. Adobe Media Server logs are located in the *rootinstall*/logs folder. The master.xx.log file and the core.xx.log file show startup failures.

  Apache logs are located in the rootinstall/Apache2.2/logs folder.

6 Use the Administration Console to verify that the encoder and the client connected to the livepkgr application.

See Connect to the Administration Console.

Troubleshoot live streaming (RTMP)

1 Use the Services Control Panel applet (Windows) or the service window (Linux) to verify that the Adobe Media Server (AMS) and Adobe Media Administration Server services are running.

If you're using the Apache HTTP server, verify that the AMSHttpd service is running.

2 Use the Administration Console to verify that the encoder and the client connected to the live application.

See Connect to the Administration Console.

3 Verify that the server is listening to the port to which the client is trying to connect. The *rootinstall*/logs/edge.xx.log file shows on which ports the server is listening. By default, the server listens on ports 1935 and 80. If the server is not listening on port 1935, open the rootinstall/ams.ini file, set `ADAPTOR.HOSTPORT` = :1935,80 and restart the server.

In the access.00.log file, the `s-uri` and `cs-uri-stem` fields indicate the port to which the client attempted to connect. Unless you specify a port number in the URL, RTMP and RTMFP clients connect to the server over port 1935 and fall back to port 80.

For more information, see Port requirements.

4 In Flash Media Live Encoder, select the Encoding Options tab, choose Output from the Panel options menu, and verify the following:

- The value of AMS URL is rtmp://*ams-dns-or-ip*/live. If you're testing on the same server as Adobe Media Server, you can use the value localhost for *ams-dns-or-ip*.
- For a single stream, the value of Stream is **livestream**.
- For adaptive bitrate streaming, the value of Stream is **livestream%i**.

5 Verify that the request URL is correc. See "URLs for publishing and playing live streams over RTMP" on page 18.

6 The live service does not support DVR recording.

Use the DVRCast application available from Adobe Media Server Tools. For more information, see the article Using DVRCast with Flash Media Live Encoder in the Adobe Media Server Developer Center.

7 Check the logs for errors. Adobe Media Server logs are located in the *rootinstall*/logs folder. The master.xx.log file and the core.xx.log file show startup failures.

Apache logs are located in the rootinstall/Apache2.2/logs folder.

Troubleshoot on-demand streaming (HTTP)

***Note:** When you play a video over HTTP, the client does not connect to the vod application. Instead, Apache serves the video to the client.*

1 Use the Services Control Panel applet (Windows) or the service window (Linux) to verify that the Adobe Media Server (AMS) and AMSHttpd services are running.

2 Verify that the server is listening to the port to which the client is trying to connect. The *rootinstall*/logs/edge.xx.log file shows on which ports the server is listening.

By default, Adobe Media Server proxies requests on port 80 to Apache HTTP Server on port 8134. Proxying HTTP streaming traffic can cause issues. If Adobe Media Server is listening on port 80, use port 8134 in the request URL. For example, http://ams.example.com:8134/hds-vod/sample1_1500.f4v.f4m. Otherwise, configure Apache HTTP Server to listen on port 80 and configure Adobe Media Server not to listen on port 80. See Configure ports for HTTP streaming.

3 Verify that the request URL is correct. See "URLs for playing on-demand streams over HTTP" on page 23.

4 By default, Apache streams on-demand media from the *rootinstall*/webroot/vod folder. To change this location, see "Content storage (HDS and HLS)" on page 56.

5 Check the logs for errors. Apache logs are located in the rootinstall/Apache2.2/logs folder.

Troubleshoot on-demand streaming (RTMP)

1 Use the Services Control Panel applet (Windows) or the service window (Linux) to verify that the Adobe Media Server (AMS) and Adobe Media Administration Server services are running.

If you're using the Apache HTTP Server, verify that the FMSHttpd service is running.

2 Use the Administration Console to verify that the client is connected to the vod application.

See Connect to the Administration Console.

3 Verify that the server is listening to the port to which the client is trying to connect. The *rootinstall*/logs/edge.xx.log file shows on which ports the server is listening. By default, the server listens on ports 1935 and 80. If the server is not listening on port 1935, open the rootinstall/ams.ini file, set `ADAPTOR.HOSTPORT` = :1935,80 and restart the server.

In the access.00.log file, the `s-uri` and `cs-uri-stem` fields indicate the port to which the client attempted to connect. Unless you specify a port number in the URL, RTMP and RTMFP clients connect to the server over port 1935 and fall back to port 80.

For more information, see Port requirements.

4 Open the Administration Console (*rootinstall*/tools/fms_adminConsole.htm) and choose View Applications to verify that the client is connecting to the vod application. To reset your password, see Reset Administration Console password.

5 Verify that the request URL is correct. See "URLs for playing on-demand media files over RTMP" on page 26

6 Do not include the /media folder in the stream URL to play the file. When you specify a filename, for example, mp4:mymediafile.f4v, the server is configured to look for /applications/vod/media/mymediafile.f4v.

7 Verify that the *rootinstall*/applications/vod directory is installed. If any files are missing, uninstall and reinstall the server.

8 Check the logs. Adobe Media Server logs are located in the *rootinstall*/logs folder. The master.xx.log file and the core.xx.log file show startup failures.

Troubleshoot multicast streaming (RTMFP)

1 Use the Services Control Panel applet (Windows) or the service window (Linux) to verify that the Adobe Media Server (AMS) and Adobe Media Administration Server services are running.

If you're using the Apache HTTP Server, verify that the FMSHttpd service is running.

2 Use the Administration Console to verify that the encoder and the client connected to the multicast application.

See Connect to the Administration Console.

3 Follow the steps in the tutorial "Multicast media (RTMFP)" on page 36.

4 Verify that the correct ports are open. The *rootinstall*/logs/edge.xx.log file shows on which ports the server is listening.

Open UDP 1935 and 19350-65535. If the server is located behind a NAT, specify its public (outside NAT) address in the *rootinstall*/conf/_defaultRoot_/Adaptor.xml file in the Adaptor/RTMFP/Core/HostPortList element. See Configure ports.

5 Check the logs. Adobe Media Server logs are located in the *rootinstall*/logs folder. The master.xx.log file and the core.xx.log file show startup failures.

Contact Support

- Post a question to the Adobe Media Server forum at forums.adobe.com/community/flash/flash_media_server. Many members of the Adobe Media Server engineering and support teams answer questions in the forum.
- Contact Adobe Support at www.adobe.com/support.

Chapter 2: Content protection

Configuring content protection for HDS

You can use Protected HTTP Dynamic Streaming (PHDS) or Adobe Access for protecting content for HDS.

Overview

Configure PHDS/Adobe Access for live streaming at the following levels:

- Server—*rootinstall*/Apache2.2/conf/httpd.conf
- Application—*rootinstall*/applications/livepkgr/Application.xml
- Event—*rootinstall*/applications/livepkgr/events/_definst_/liveevent/Event.xml

PHDS

Use Adobe Media Server 5 to serve live and on-demand protected content to Flash Player and AIR over HTTP without using a DRM License Server. When Adobe Media Server packages the content, it generates the license and embeds it into the DRM metadata of the content stream. This feature is called Protected HTTP Dynamic Streaming (PHDS). In addition to encrypting content, PHDS also supports SWF verification for HTTP Dynamic Streaming.

The F4F packaging process for on-demand and live PHDS generates a license, embeds it in the DRM metadata, and delivers it with the media. Flash Player 11 and AIR 3 clients can retrieve the license from the content stream, which eliminates communication between the client and a License Server.

The Adobe Media Server installer generates credentials, certificates, and policy files to the *rootinstall*/creds directory. The installer also creates a common-key.bin file in the /creds directory. You can change the content of this file or create a new common key file. To create a common key file (common-key.bin), which is used to derive the Content Encryption Key, use the Scramble tool. See the Scramble tool.

Use the following policy files to generate licenses for on-demand and live PHDS (AMS 5 includes four new policy files to support output protection):

Policy name	Description
phds_24hr_policy.pol	24 Hour limited policy `anonymous; 24 hours limited license caching.` This is the default policy. Users can start playback within 24 hours of the time the content was packaged. Users can continue watching the content until the end of the content (users may pause content). The 24 hours window starts when the DRM metadata is generated.
phds_policy.pol	Unlimited policy `anonymous; unlimited license caching; and binding to Protected Streaming is permitted` This policy allows playback at any time.

Policy name	Description
phds-24hr-OPBestEffort.pol	(AMS 5) 24 Hours Limited / Best Effort Output Protection Policy Set in the same way as the 24 Hours Limited / No Output Protection Policy policy with an additional restriction to use hardware content protection, if available. Users are still able to playback media if the client hardware doesn't support Output Protection. If the client hardware supports Output Protection but it is disabled, Flash Player returns DRM Run Time Error: 3342 (NoDigitalProtectionAvail).
phds-OPBestEffort.pol	(AMS 5) Unlimited / Best Effort Protection Policy Set in the same way as the Unlimited / No Output Protection Policy policy with an additional restriction to use hardware content protection, if available. Users arestill able to playback media if the client hardware doesn't support Output Protection. If the client hardware supports Output Protection, but it is disabled, Flash Player returns DRM Run Time Error: 3342 (NoDigitalProtectionAvail).
phds-24hr-OPRequired.pol	(AMS 5) 24 Hours Limited / Required Output Protection Policy Set in the same way as the 24 Hours Limited / No Output Protection Policy policy with an additional restriction to use hardware content protection. Users cannot playback media if the client hardware doesn't support Output Protection. If the client hardware doesn't support Output Protection or if it supports Output Protection, but it is disabled, Flash Player returns DRM Run Time Error: 3342 (NoDigitalProtectionAvail).
phds-OPRequired.pol	(AMS 5) Unlimited / Required Output Protection Policy Set in the same way as the Unlimited / No Output Protection Policy policy with an additional restriction to use hardware content protection. Users cannot playback media if the client hardware doesn't support Output Protection. If the client hardware doesn't support Output Protection or if it supports Output Protection but, it is disabled, Flash Player returns DRM Run Time Error: 3342 (NoDigitalProtectionAvail).

The simple unlimited policy is not intended for a regular use. It is provided as a temporary work around in case there is an issue with the network. When media is cached on network devices between Adobe Media Server and Flash Player, clients may receive expired policy data from the network instead of the expected media from the server. If media that was generated with the 24 hours policy is cached for more than 24 hours the player does not allow playback. Switch to the unlimited PHDS policy as a temporary solution until the network configuration is fixed and the caches are flushed. This solution allows you to distribute media with lower protection instead of not distributing the media. After switching to the Unlimited Policy, flush the caches to allow the unlimited license to propagate to clients.

Adobe Access

To deliver live or on-demand content with HDS, you can enable HDS with Adobe Access for protected streaming. The Adobe Access server for protected streaming is a license server implementation optimized for use with HDS. See the Adobe Access documentation for more details.

Important: Use the HDS packagers to both encrypt and fragment content. Do not use the Adobe Access packaging tools to encrypt content. The HDS packagers cannot fragment encrypted content.

Note: *The Adobe Access SDK and the Adobe Access license server reference implementation can issue licenses for HDS.*

After you have deployed Adobe Access Server for protected streaming, configure Adobe Media Server to package and encrypt the content in real-time.

Live use case

In httpd.conf, `ContentProtection` tag is specified under `<Location hds-live>`.

Whereas, both the Application.xml file and the Event.xml file have a `ContentProtection` container that holds the live PHDS configuration settings. In Application.xml, the container is located under `//Application/HDS/Recording/ContentProtection`. In Event.xml, the container is located under `//Event/Recording/ContentProtection`.

Getting Started

To quickly get started with PHDS, you need to understand the following directives:

Directive	Default Value	Description
`HttpStreamingEncryptionScope`	content	Possible values are off, content, and server. When the value is off, content remains in the unprotected format. When the value is content, configuration settings in the application.xml or event.xml files are used to protect the content. When the value is server, configuration settings in the httpd.conf are used to protect the content.
`HttpStreamingProtectionScheme`	PHDS	Encryption type for the content. It can be FlashAccessV3, FlashAccessV2 or PHDS. HttpStreamingProtectionScheme is applicable if encryption is enabled. Use HttpStreamingEncryptionScope to determine the scope of the encryption.

To configure PHDS with basic settings, perform the following steps:

1 After installing Adobe Media Server, navigate to the <*root-install*>/Apache 2.2/conf/ directory. Edit the httpd.conf file and add the following tags under `<Location hds-live>`:

```
<Location /hds-live>
    HttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications"
    HttpStreamingContentPath "../applications"
    HdsFmsDirPath ".."
    HttpStreamingF4MMaxAge 2
    HttpStreamingBootstrapMaxAge 2
    HttpStreamingDrmmetaMaxAge 3600
    HttpStreamingFragMaxAge -1

    HttpStreamingEncryptionScope serverHttpStreamingProtectionScheme PHDS
</Location>
```

Note: *This configuration change will enable PHDS at the server level.*

2 Publish a live stream called "livestream?adbe-live-event=liveevent" to livepkgr.

3 Playback the stream using the URI http://<*server-ip*>:8134/hds-live/livepkgr/_definst_/liveevent/livetsream.f4m.

Detailed configuration

The following sections provides detailed configurations for both PHDS and Adobe Access schemes.

Server level

Server-level configurations for live PHDS/Adobe Access

When server level configuration is specified, the protection parameters specified are applied server wide. Encryption parameters specified in Application/Event level will be ignored.

Flash Media Server 4.5.3 and higher allows setting the encryption configurations at the server level. These settings will apply to live events recorded on the server. To enable or disable encryption, configure the following directives for the `f4fhttp_module` in the Apache httpd.conf file:

Common configuration:

Directive	Default Value	Description
`HdsFmsDirPath`	None	Relative path of the Adobe Media Server root directory. Use '..' as Relative path.
`HttpStreamingEncryptionScope`	content	Possible values are off, content, and server. When the value is off, content remains in the unprotected format. When the value is content, configuration settings in the application.xml or event.xml files are used to protect the content. When the value is server, configuration settings in the httpd.conf are used to protect the content.
`HttpStreamingProtectionScheme`	PHDS	Encryption type for the content. It can be FlashAccessV3, FlashAccessV2 or PHDS. HttpStreamingProtectionScheme is applicable if encryption is enabled. Use HttpStreamingEncryptionScope to determine the scope of the encryption.

PHDS configuration

Directive	Default Value	Description
`PHDSCommonKeyFile`	`<AMSInstallDir>`/creds/common-key.bin	A common key used to protect content at this location. PHDSCommonKeyFile path is relative to rootinstall/Apache2.2.
`PHDSVideoEncryptionLevel`	2	The level of encryption for the content (0-low,1-medium, 2-high). Lower settings provide partial encryption. A subset of the samples (like video keyframes) are encrypted. Partial encryption can improve playback performance on the client, because there are fewer frames to decrypt.
`PHDSPlaybackExpiration`	24Hours	The duration within which the content playback is available. Possible values are 24Hours and Unlimited.
`PHDSOutputProtection`	none	The required hardware Output Protection of media on the client. Possible values are None, BestEffort, and Required.

Adobe Access configuration

Directive	**Default Value**	**Description**
`HdsDrmCommonKeyFile`	None	A common key used to protect content at this location. HdsDrmCommonKeyFile path is relative to rootinstall/Apache2.2.
`HdsDrmLicenseServerURL`	None	The URL of the license server used for protecting content.
`HdsDrmTransportCertFile`	None	The transport certificate used for protecting content.
`HdsDrmLicenseServerCertFile`	None	The License server certificate used for protecting content.
`HdsDrmPackagerCredentialFile`	None	The Packager credential used for protecting content.
`HdsDrmPackagerCredentialPassword`	None	The Packager credential password for the configured packager credential file.
`HdsDrmPolicyFile`	None	Policy for protecting content.
`HdsDrmUseUniqueContentID`	false	By default, Adobe Media Server uses non-unique content ID if the protection scheme is set. The file path is used as the content ID. Hence, the content IDs for the files present in a directory will be the same. This feature is used in allowing content ID encryption for multi-bitrate stream where all the files with different bit rate are kept in the same directory. If you want to use unique content IDs for each file, add this element.
`HdsDrmContentID`	None	You can manually specify the content ID, which will be used for all the files. Note that this element must not be used when `HdsDrmUseUniqueContentID` is set to true.

The following example enables and configures PHDS in the httpd.conf file. These settings apply to every live event configured for this server.

```
<Location /hds-live>
    HttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications"
    HttpStreamingContentPath "../applications"
    HdsFmsDirPath ".."
    HttpStreamingF4MMaxAge 2
    HttpStreamingBootstrapMaxAge 2
    HttpStreamingDrmmetaMaxAge 3600
    HttpStreamingFragMaxAge -1
    Options -Indexes FollowSymLinks
    HttpStreamingEncryptionScope server
    HttpStreamingProtectionScheme PHDS
    PHDSCommonKeyFile "../creds/common-key.bin"
    PHDSPlaybackExpiration 24Hours
    PHDSOutputProtection  None
</Location>
```

The following example enables and configures Adobe Access (FlashAccessV2) in the httpd.conf file. These settings apply to every live event configured for this server.

```
<Location /hds-live-faxs>
    HttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications"
    HttpStreamingContentPath "../applications"
    HdsFmsDirPath ".."
    HttpStreamingF4MMaxAge 2
    HttpStreamingBootstrapMaxAge 2
    HttpStreamingDrmmetaMaxAge 3600
    HttpStreamingFragMaxAge -1
    HttpStreamingEncryptionScope server
    HttpStreamingProtectionScheme FlashAccessV2
    HdsDrmCommonKeyFile "../creds/common-key.bin"HdsDrmLicenseServerURL http://<aaxs-test-
server>/HdsDrmTransportCertFile "aaxs-test-server-trnsCert.der"
    HdsDrmLicenseServerCertFile "aaxs-test-server-licCert.der"
    HdsDrmPackagerCredentialFile " aaxs-test-server-
pkgrCert.pfx"HdsDrmPackagerCredentialPassword pwd=HdsDrmPolicyFile "sample_policy.pol"
    Options -Indexes FollowSymLinks
</Location>
```

The following example enables and configures Adobe Access (FlashAccessV3) in the httpd.conf file. These settings apply to every live event configured for this server.

```
<Location /hds-live-faxs>
    HttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications"
    HttpStreamingContentPath "../applications"
    HdsFmsDirPath ".."
    HttpStreamingF4MMaxAge 2
    HttpStreamingBootstrapMaxAge 2
    HttpStreamingDrmmetaMaxAge 3600
    HttpStreamingFragMaxAge -1
    HttpStreamingEncryptionScope server
    HttpStreamingProtectionScheme FlashAccessV3
    HdsDrmCommonKeyFile "../creds/common-key.bin"HdsDrmLicenseServerURL http://<aaxs-test-
server>/HdsDrmTransportCertFile "aaxs-test-server-trnsCert.der"
    HdsDrmLicenseServerCertFile "aaxs-test-server-licCert.der"
    HdsDrmPackagerCredentialFile " aaxs-test-server-
pkgrCert.pfx"HdsDrmPackagerCredentialPassword pwd=HdsDrmPolicyFile "sample_policy.pol"
    Options -Indexes FollowSymLinks
</Location>
```

Application level

When Application level configuration is specified, the protection parameters specified are applied to the particular application (to all the events under the application). Encryption parameters specified in Event/Server level will be ignored.

Common configuration

Element	Default	Description
HDS/Recording/Content Protection	"allow" in Application.xml "false" in Event.xml	Container element for content protection configurations. In Application.xml, set the enabled attribute to "true" to enable content protection, "false" to disable content protection, or "allow" to allow settings in the Event.xml file to override the ContentProtection section of the Application.xml file. When enabled="allow", the server uses none of the settings in the ContentProtection section of the Application.xml file. If a ContentProtection section is not specified in Event.xml, content protection is disabled because the default value is "false" in Event.xml. In Event.xml, set the enabled attribute to "true"or "false".
HDS/Recording/Content Protection/Protection Scheme	None	Possible values are phds, FlashAccessV2, and FlashAccessV3. For PHDS, use PHDS.

PHDS configuration

Element	Default	Description
HDS/Recording/Content Protection/PHDS	None	Container for PHDS encryption settings.
HDS/Recording/Content Protection/PHDS/CommonKeyFile	None	A relative path to the common-key.bin file containing a base key used (along with the content ID) to generate the final content encryption key. This file is generated during installation to *rootinstall*/creds/common-key.bin. If you define the CommonKeyFile in the Application.xml file, the server looks for the file relative to the application directory. If you define the CommonKeyFile in the Event.xml file, the server looks for the file relative to the event folder.
HDS/Recording/Content Protection/PHDS/PlaybackExpiration	24Hours	The protection policy. The policy determines the duration within which content playback is available. Possible values are 24Hours and Unlimited.
HDS/Recording/Content Protection/PHDS/VideoEncryptionLevel	2	The level of encryption for the content (0-low,1-medium,2-high). Lower settings mean "partial encryption", where a subset of the samples (like video keyframes) are encrypted. This can improve playback performance on the client, since there will be fewer frame to decrypt.
HDS/Recording/Content Protection/PHDS/UpdateInterval	60	The frequency at which the server generates the drm metadata, in minutes.
HDS/Recording/Content Protection/PHDS/OutputProtection	None	The required hardware Output Protection of media on the client. Possible values are None, BestEffort, and Required.

Adobe Access configuration

Element	Default	Description
`HDS/Recording/ContentProtection/FlashAccessV2`	None	Container for FlashAccessV2 encryption settings.
`HDS/Recording/ContentProtection/FlashAccessV3`	None	Container for FlashAccessV3 encryption settings.
`HDS/Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/ContentID`	None	The content ID used when protecting the streams in the live event
`HDS/Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/CommonKeyFile`	None	The file containing the common key
`HDS/Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/LicenseServerURL`	None	The URL of the license server that will provide licensing services for the protected content
`HDS/Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/TransportCertFile`	None	The file containing the transport certificate, in DER format
`HDS/Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/LicenseServerCertFile`	None	The file containing the license server certificate, in DER format
`HDS/Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/PackagerCredentialFile`	None	The file containing the packager credentials, in PFX format
`HDS/Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/PackagerCredentialPassword`	None	The password for the packager credentials
`HDS/Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/ PolicyFile`	None	The file containing the content protection policy

Configure the httpd.conf as given below to allow protection configurations at the application level.

```
<Location /hds-live>
    HttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications"
    HttpStreamingContentPath "../applications"
    HdsFmsDirPath ".."
    HttpStreamingF4MMaxAge 2
    HttpStreamingBootstrapMaxAge 2
    HttpStreamingDrmmetaMaxAge 3600
    HttpStreamingFragMaxAge -1
    Options -Indexes FollowSymLinks
    HttpStreamingEncryptionScope content
</Location>
```

The following example enables and configures PHDS in the Application.xml file. These settings apply to every live event configured for this application.

```
<Application>
    <StreamManager>
        <Live>
            <AssumeAbsoluteTime>true</AssumeAbsoluteTime>
            <PublishTimeout>0</PublishTimeout>
            <AdjustForZeroTimeStampMessages>2</AdjustForZeroTimeStampMessages>
            <AdjustForRecordingRollover>false</AdjustForRecordingRollover>
         </Live>
    </StreamManager>
    <HDS>
        <Recording >
            <ContentProtection enabled="true" >
                <ProtectionScheme>PHDS</ProtectionScheme>
                    <PHDS>
                        <CommonKeyFile>common-key.bin</CommonKeyFile>
                        <VideoEncryptionLevel>2</VideoEncryptionLevel>
                        <PlaybackExpiration>24Hours</PlaybackExpiration>
                        <OutputProtection>None</OutputProtection>
                    </PHDS>
            </ContentProtection>
        </Recording>
    </HDS>
</Application>
```

The following example enables and configures Adobe Access V2 in the Application.xml file. These settings apply to every live event configured for this application.

```
<Appl ication>
    <S treamManager>
        < Live>
            <Assume Ab soluteTime> true</ AssumeAbsoluteTime>
            <PublishTimeou t >0</Pu blishTimeout>
            <AdjustForZeroTimeStampMessages>2< /AdjustF orZeroTimeStampMessages>
            < AdjustForRe cordingRollover>false</AdjustForRecordingRollover>
        </Live>
    </Str eamManager>
    <HDS>
        <Re cording>
            <ContentProtection enab led="true">
                <ProtectionScheme>FlashAccessV2</ProtectionScheme>
                <FlashAccessV2>
                    <ContentID>liveevent</ContentID>
                    <CommonKeyFile>common-key.bin</CommonKeyFile>
                    <LicenseServerURL>http://<aaxs-test-server>/</LicenseServerURL>
                    <TransportCertFile>
                        aaxs-test-server-trnsCert.der
                    </TransportCertFile>
                    <LicenseServerCertFile>
                        aaxs-test-server-licCert.der
                    </LicenseServerCertFile>
                    <PackagerCredentialFile>
                        aaxs-test-server-pkgrCert.pfx</PackagerCredentialFile>
                    <PackagerCredentialPassword>pwd=</PackagerCredentialPassword>
                    <PolicyFile>sample_policy.pol</PolicyFile>
                </FlashAccessV2>
            </ContentProtection>
        </Recording>
    </HDS>
</Application>
```

The following example enables and configures Adobe Access V3 in the Application.xml file. These settings apply to every live event configured for this application.

```
<Application >
      <StreamManager>
        <Live>
            <AssumeAbsoluteTime>true</AssumeAbsoluteTime>
            <PublishTimeout>0</PublishTimeout>
            <AdjustForZeroTimeStampMessages>2</AdjustForZeroTimeStampMessages>
            <AdjustForRecordingRollover>false</AdjustForRecordingRollover>
        </Live>
    </StreamManager>
    <HDS>
        <Recording>
            <ContentProtection enabled="true">
            <ProtectionScheme>FlashAccessV3</ProtectionScheme>
                <FlashAccessV3>
                    <ContentID>liveevent</ContentID>
                    <CommonKeyFile>common-key.bin</CommonKeyFile>
                    <LicenseServerURL>http://<aaxs-test-server>/</LicenseServerURL>
                    <TransportCertFile>
                        aaxs-test-server-trnsCert.der
                    </TransportCertFile>
                    <LicenseServerCertFile>
                        aaxs-test-server-licCert.der</LicenseServerCertFile>
                    <PackagerCredentialFile>
                        aaxs-test-server-pkgrCert.pfx
                    </PackagerCredentialFile>
                    <PackagerCredentialPassword>pwd=</PackagerCredentialPassword>
                    <PolicyFile>sample_policy.pol</PolicyFile>
                </FlashAccessV3>
            </ContentProtection>
        </Recording>
    </HDS>
</Application>
```

Note: *In this case, copy the common-key.bin file from the rootinstall/creds directory to the root-install/applications/livepkgr/ directory.*

Event level

When Event level configuration is specified, the protection parameters specified are applied to the particular event. Encryption parameters specified in Application/Server level will be ignored.

Common configuration

Element	**Default**	**Description**
`Recording/ContentProtection`	`"allow"` in Application.xml `"false"` in Event.xml	Container element for content protection configurations. In Application.xml, set the `enabled` attribute to `"true"` to enable content protection, `"false"` to disable content protection, or `"allow"` to allow settings in the Event.xml file to override the ContentProtection section of the Application.xml file. When `enabled="allow"`, the server uses none of the settings in the `ContentProtection` section of the Application.xml file. If a `ContentProtection` section is not specified in Event.xml, content protection is disabled because the default value is `"false"` in Event.xml. In Event.xml, set the `enabled` attribute to `"true"`or `"false"`.
`Recording/ContentProtection/ProtectionScheme`	None	Possible values are `phds`, `FlashAccessV2`, `and FlashAccessV3`. For PHDS, use `PHDS`.

PHDS configuration

Element	**Default**	**Description**
`Recording/ContentProtection/PHDS`	None	Container for PHDS encryption settings.
`Recording/ContentProtection/PHDS/CommonKeyFile`	None	A relative path to the common-key.bin file containing a base key used (along with the content ID) to generate the final content encryption key. This file is generated during installation to *rootinstall*/creds/common-key.bin. If you define the `CommonKeyFile` in the Application.xml file, the server looks for the file relative to the application directory. If you define the `CommonKeyFile` in the Event.xml file, the server looks for the file relative to the event folder.
`Recording/ContentProtection/PHDS/PlaybackExpiration`	24Hours	The protection policy. The policy determines the duration within which content playback is available. Possible values are `24Hours` and `Unlimited`.
`Recording/ContentProtection/PHDS/VideoEncryptionLevel`	2	The level of encryption for the content (0-low,1-medium,2-high). Lower settings mean "partial encryption", where a subset of the samples (like video keyframes) are encrypted. This can improve playback performance on the client, since there will be fewer frame to decrypt.
`Recording/ContentProtection/PHDS/UpdateInterval`	60	The frequency at which the server generates the drm metadata, in minutes.
`Recording/ContentProtection/PHDS/OutputProtection`	None	The required hardware Output Protection of media on the client. Possible values are None, BestEffort, and Required.

Adobe Access configuration

Element	**Default**	**Description**
`Recording/ContentProtection/FlashAccessV2`	None	Container for FlashAccessV2 encryption settings.
`Recording/ContentProtection/FlashAccessV3`	None	Container for FlashAccessV3 encryption settings.
`Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/ContentID`	None	The content ID used when protecting the streams in the live event
`Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/CommonKeyFile`	None	The file containing the common key
`Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/LicenseServerURL`	None	The URL of the license server that will provide licensing services for the protected content
`Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/TransportCertFile`	None	The file containing the transport certificate, in DER format
`Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/LicenseServerCertFile`	None	The file containing the license server certificate, in DER format
`Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/PackagerCredentialFile`	None	The file containing the packager credentials, in PFX format
`Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/PackagerCredentialPassword`	None	The password for the packager credentials
`Recording/ContentProtection/FlashAccessV3(or FlashAccessV2)/PolicyFile`	None	The file containing the content protection policy

Configure the httpd.conf as given below to allow protection configurations at the event level:

```
<Location /hds-live>
    HttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications"
    HttpStreamingContentPath "../applications"HdsFmsDirPath ".."
    HttpStreamingF4MMaxAge 2
    HttpStreamingBootstrapMaxAge 2
    HttpStreamingDrmmetaMaxAge 3600
    HttpStreamingFragMaxAge -1
    Options -Indexes FollowSymLinks
    HttpStreamingEncryptionScope content
</Location>
```

The following is an example of an Application.xml file that allows protection configurations at the event level and tells the server to look for configurations in the Event.xml file for each live event:

```
<Application>
    <StreamManager>
        <Live>
            <AssumeAbsoluteTime>true</AssumeAbsoluteTime>
        </Live>
    </StreamManager>

    <HDS>
        <Recording>
            <ContentProtection enabled="allow">
            </ContentProtection>
        </Recording>
    </HDS>

</Application>
```

The following Event.xml file configures PHDS for a single live event:

```
<Event>
    <EventID>liveevent</EventID>
    <Recording>
        <FragmentDuration>4000</FragmentDuration>
        <SegmentDuration>400000</SegmentDuration>
        <DiskManagementDuration>3</DiskManagementDuration>
        <ContentProtection enabled="true">
            <ProtectionScheme>PHDS</ProtectionScheme>
            <PHDS>
                <CommonKeyFile>common-key.bin</CommonKeyFile>
                <VideoEncryptionLevel>2</VideoEncryptionLevel>
                <PlaybackExpiration>24Hours</PlaybackExpiration>
                <OutputProtection>None</OutputProtection>
            </PHDS>
        </ContentProtection>
    </Recording>
</Event>
```

In this case, copy the common-key.bin file from the *rootinstall*/creds directory to the *rootinstall*/applications/livepkgr/events/_definst_/liveevent directory.

The following Event.xml file configures Adobe Access V2 for a single live event:

```
<Event>
    <EventID>liveevent</EventID>
    <Recording>
        <FragmentDuration>4000</FragmentDuration>
        <SegmentDuration>400000</SegmentDuration>
        <DiskManagementDuration>3</DiskManagementDuration>
        <ContentProtection enabled="true">
            <ProtectionScheme>FlashAccessV2</ProtectionScheme>
            <FlashAccessV2>
                <ContentID>liveevent</ContentID>
                <CommonKeyFile>common-key.bin</CommonKeyFile>
                <LicenseServerURL>
                    http://<aaxs-test-server>/
                </LicenseServerURL>
                <TransportCertFile>
                    aaxs-test-server-trnsCert.der
                </TransportCertFile>
                <LicenseServerCertFile>
                    aaxs-test-server-licCert.der
                </LicenseServerCertFile>
                <PackagerCredentialFile>
                    aaxs-test-server-pkgrCert.pfx
                </PackagerCredentialFile>
                <PackagerCredentialPassword>pwd=</PackagerCredentialPassword>
                <PolicyFile>sample_policy.pol</PolicyFile>
            </FlashAccessV2>
        </ContentProtection>
    </Recording>
</Event>
```

The following Event.xml file configures Adobe Access V3 for a single live event:

```
<Event>
    <EventID>liveevent</EventID>
    <Recording>
        <FragmentDuration>4000</FragmentDuration>
        <SegmentDuration>400000</SegmentDuration>
        <DiskManagementDuration>3</DiskManagementDuration>
        <ContentProtection enabled="true">
            <ProtectionScheme>FlashAccessV3</ProtectionScheme>
            <FlashAccessV3>
                <ContentID>liveevent</ContentID>
                <CommonKeyFile>common-key.bin</CommonKeyFile>
                <LicenseServerURL>
                    http://<aaxs-test-server>/
                </LicenseServerURL>
                <TransportCertFile>
                    aaxs-test-server-trnsCert.der
                </TransportCertFile>
                <LicenseServerCertFile>
                    aaxs-test-server-licCert.der
                </LicenseServerCertFile>
                <PackagerCredentialFile>
                    aaxs-test-server-pkgrCert.pfx
                </PackagerCredentialFile>
                <PackagerCredentialPassword>pwd=</PackagerCredentialPassword>
                <PolicyFile>sample_policy.pol</PolicyFile>
            </FlashAccessV3>
        </ContentProtection>
    </Recording>
</Event>
```

Note: *In this case, copy the common-key.bin file from the rootinstall/creds directory to the rootinstall/applications/livepkgr/events/_definst_/liveevent directory.*

License chaining

Adobe Media Server will support embedding leaf licenses in the DRM metadata from the policy generated using a chained license. Adobe Media Server will need the license server credential and the credential password configured so that the root license from the policy can be used to encrypt the CEK contained in the embedded leaf license.

If the configuration for embedding the leaf license is turned off, Adobe Media Server will still support such a policy except that the leaf license will not be embedded in the DRM metadata.

Note: *The support will be limited to a single license server credential and credential-password pair.*

The following table provides the required configuration:

Parameter	Description	Default value
HdsDrmEmbedLeafLicense (Server level) EmbedLeafLicense (Application and event level)	Enables embedding of leaf licenses in DRM metadata. Possible values are "true" or "false". ***Note:*** *The policy file must be created using a chained license.*	false

Parameter	Description	Default value
HdsDrmLicenseServerCredentialFile (Server level) LicenseServerCredentialFile (Application and event level)	Required if HdsDrmEmbedLeafLicense is set to true. The license server credential used when protecting content at this location.	NA
HdsDrmLicenseServerCredentialPassword (Server level) LicenseServerCredentialPassword (Application and event level)	Required if HdsDrmEmbedLeafLicense is set to true. The license server credential password for the configured license server credential file.	NA

The following example shows the license chaining configuration at the application level:

```
<Application>
    <HDS>
        <Recording>
            <ContentProtection enabled="true">
                <ProtectionScheme>FlashAccessV3</ProtectionScheme>
                <FlashAccessV3>
                    <ContentID>liveevent</ContentID>
                    <CommonKeyFile>common-key.bin</CommonKeyFile>
                    <LicenseServerURL>http://<aaxs-test-server>/</LicenseServerURL>
                    <TransportCertFile>
                        aaxs-test-server-trnsCert.der
                    </TransportCertFile>
                    <LicenseServerCertFile>
                        aaxs-test-server-licCert.der
                    </LicenseServerCertFile>
                    <PackagerCredentialFile>
                        aaxs-test-server-pkgrCert.pfx
                    </PackagerCredentialFile>
                    <PackagerCredentialPassword>pwd=</PackagerCredentialPassword>
                    <PolicyFile>sample_policy.pol</PolicyFile>
                    <EmbedLeafLicense>true</EmbedLeafLicense>
                    <LicenseServerCredentialFile>
                        aaxs-test-server-pkgrCertLic.pfx
                    </LicenseServerCredentialFile>
                    <LicenseServerCredentialPassword>
                        pwd_lic=
                    </LicenseServerCredentialPassword>
                </FlashAccessV3>
            </ContentProtection>
        </Recording>
    </HDS>
</Application>
```

The following example shows the license chaining configuration at the event level:

```
<Event>
    <Recording>
        <ContentProtection enabled="true">
                <ProtectionScheme>FlashAccessV3</ProtectionScheme>
                <FlashAccessV3>
                        <ContentID>liveevent</ContentID>
                        <CommonKeyFile>common-key.bin</CommonKeyFile>
                        <LicenseServerURL>
                            http://<aaxs-test-server>/
                        </LicenseServerURL>
                        <TransportCertFile>
                            aaxs-test-server-trnsCert.der
                        </TransportCertFile>
                        <LicenseServerCertFile>
                            aaxs-test-server-licCert.der
                        </LicenseServerCertFile>
                        <PackagerCredentialFile>
                            aaxs-test-server-pkgrCert.pfx
                        </PackagerCredentialFile>
                        <PackagerCredentialPassword>
                            pwd=
                        </PackagerCredentialPassword>
                        <PolicyFile>sample_policy.pol</PolicyFile>
                        <EmbedLeafLicense>true</EmbedLeafLicense>
                        <LicenseServerCredentialFile>
                            aaxs-test-server-pkgrCertLic.pfx
                        </LicenseServerCredentialFile>
                        <LicenseServerCredentialPassword>
                            pwd_lic=
                        </LicenseServerCredentialPassword>
                </FlashAccessV3>
        </ContentProtection>
    </Recording>
</Event>
```

***Note:** License chaining is not supported for the "VOD use case" on page 141.*

Key rotation

Adobe Media Server 5 supports Key Rotation for protected HTTP Dynamic Streaming when used with Adobe Access and PHDS. You can encrypt content packaged with AMS 5 using a set of keys. You can periodically change the encryption key and specify how often the content encryption key is to be changed.

Server level - Adobe Access

Parameter	Description	Default value
`HdsDrmEnableKeyRotation`	Whether to use Key Rotation with AAXS protection scheme	false
`HdsDrmKeyRotationInterval`	Key rotation interval to be used (in seconds), when enabling key rotation.	900 seconds

The following httpd.conf will enable key rotation at server level :

```
<Location /hds-live>
    HttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications"
    HttpStreamingContentPath "../applications" HdsFmsDirPath ".."
    HttpStreamingF4MMaxAge 2
    HttpStreamingBootstrapMaxAge 2
    HttpStreamingDrmmetaMaxAge 3600
    HttpStreamingFragMaxAge -1
    Options -Indexes FollowSymLinks
    HttpStreamingEncryptionScope server
    HttpStreamingProtectionScheme FlashAccessV3
    HdsDrmCommonKeyFile "../creds/common-key.bin"
    HdsDrmLicenseServerURL http://<aaxs-test-server>/
    HdsDrmTransportCertFile aaxs-test-server-trnsCert.der
    HdsDrmLicenseServerCertFile aaxs-test-server-licCert.der
    HdsDrmPackagerCredentialFile aaxs-test-server-pkgrCert.pfx
    HdsDrmPackagerCredentialPassword pwd=
    HdsDrmPolicyFile sample_policy.pol
    HdsDrmEnableKeyRotation true
    HdsDrmKeyRotationInterval 500
</Location>
```

Application level - Adobe Access

Parameter	Description	Default value
HDS/Recording/ContentProtection/FlashAccessV3/EnableKeyRotation	Whether to use Key Rotation with AAXS protection scheme	false
HDS/Recording/ContentProtection/FlashAccessV3/KeyRotationInterval	Key rotation interval to be used (in seconds), when enabling key rotation.	900 seconds
HDS/Recording/ContentProtection/FlashAccessV3/KeyRotationFilePath	The file containing the rotation keys to be used. This file will contain a sequence of rotated keys used to encrypt content. If no file is specified, randomly generated keys will be used. The keys must be 16 bytes in length and specified as hex values.	Randomly generated keys will be used (as described below)

The following Application.xml will enable key rotation at Application level :

```
<Application>
    <StreamMana ger>
        <Live>
            <AssumeAbsoluteTime>true</AssumeAbsoluteTime>
            <PublishTimeout>0</PublishTim eout>
            <Adj ustForZeroTimeStampMessages>2</AdjustForZeroTimeStampMessages>
            <AdjustForRecordingRollover>false</AdjustForRecordingRollover>
        </Live>
    </StreamManager >
    <HDS>
        <Recording>
            <ContentProtection enabled="true">
                <ProtectionScheme>FlashAccessV3</ProtectionScheme>
                <FlashAccessV3>
                    <ContentID >liveevent</ContentID>
                    <CommonKeyFile>common-key.bin</CommonKeyFile>
                    <LicenseServerURL>http://<aaxs-test-server>/</LicenseServerURL>
                    <TransportCertFile>
                        aaxs-test-server-trnsCert.der
                    </TransportCertFile>
                    <Licens eServerCertFile>
                        aaxs-test-server-licCert.der
                    </LicenseServerCertFile>
                    < PackagerCredentialFile>
                        aaxs-test-server-pkgrCert.pfx
                    </PackagerCredentialFile>
                    <PackagerCredentialPassword>pwd=</PackagerCredentialPassword>
                    <PolicyFil e>sample _policy.pol</PolicyFile>
                    <EnableKeyRotation>true</EnableKeyRotation>
                    <KeyRotationInterval>500</KeyRotationInterval>
                    <KeyRotationFilePath>sample_keys.txt</KeyRotationFilePath>
                </FlashAccessV3>
            </ContentProtection>
        </Recording>
    </HDS>
</Application>
```

Event level - Adobe Access

Parameter	Description	Default value
Recording/ContentProtection/FlashAccessV3/EnableKeyRotation	Whether to use Key Rotation with AAXS protection scheme	false
Recording/ContentProtection/FlashAccessV3/KeyRotationInterval	Key rotation interval to be used (in seconds), when enabling key rotation.	900 seconds
Recording/ContentProtection/FlashAccessV3/KeyRotationFilePath	The file containing the rotation keys to be used. This file will contain a sequence of rotated keys used to encrypt content. If no file is specified, randomly generated keys will be used. The keys must be 16 bytes in length and specified as hex values.	Randomly generated keys will be used (as described below)

The following Event.xml will enable key rotation at Event level :

```
<Event>
    <EventID>liveevent</EventID>
    <Recording>
        <FragmentDuration>4000</FragmentDuration>
        <SegmentDuration>400000</SegmentDuration>
        <DiskManagementDuration>3</DiskManagementDuration>
        <ContentProtection enabled="true">
            <ProtectionScheme>FlashAccessV3</ProtectionScheme>
            <FlashAccessV3>
                <ContentID>liveevent</ContentID>
                <CommonKeyFile>common-key.bin</CommonKeyFile>
                <LicenseServerURL>http://<aaxs-test-server>/</LicenseServerURL>
                <TransportCertFile>aaxs-test-server-trnsCert.der</TransportCertFile>
                <LicenseServerCertFile>
                    aaxs-test-server-licCert.der
                </LicenseServerCertFile>
                <PackagerCredentialFile>
                    aaxs-test-server-pkgrCert.pfx
                </PackagerCredentialFile>
                <PackagerCredentialPassword>pwd=</PackagerCredentialPassword>
                <PolicyFile>sample_policy.pol</PolicyFile>
                <EnableKeyRotation>true</EnableKeyRotation>
                <KeyRotationInterval>500</KeyRotationInterval>
                <KeyRotationFilePath>sample_keys.txt</KeyRotationFilePath>
            </FlashAccessV3>
        </ContentProtection>
    </Recording>
</Event>
```

***Note:** `HdsDrmKeyRotationFilePath` takes path relative to <AMS-Install>/applications/<application-name>/.*

Server level - PHDS

Parameter	Description	Default value
`PHDSEnableKeyRotation`	Whether to use Key Rotation with PHDS protection scheme	false
`PHDSKeyRotationInterval`	Key rotation interval to be used (in seconds), when enabling key rotation.	900 seconds

The following httpd.conf will enable key rotation at server level :

```
<Location /hds-live>
    HttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications"
    HttpStreamingContentPath "../applications" HdsFmsDirPath ".."
    HttpStreamingF4MMaxAge 2
    HttpStreamingBootstrapMaxAge 2
    HttpStreamingDrmmetaMaxAge 3600
    HttpStreamingFragMaxAge -1
    Options -Indexes FollowSymLinks
    HttpStreamingEncryptionScope server
    HttpStreamingProtectionScheme PHDS
    PHDSVideoEncryptionLevel 2
    PHDSPlaybackExpiration 24Hours
    PHDSOutputProtection None
    PHDSEnableKeyRotation true
    PHDSKeyRotationInterval 500
</Location>
```

Application level - PHDS

Parameter	Description	Default value
HDS/Recording/ContentProtection/PHDS/EnableKeyRotation	Whether to use Key Rotation with PHDS protection scheme	false
HDS/Recording/ContentProtection/PHDS/KeyRotationInterval	Key rotation interval to be used (in seconds), when enabling key rotation.	900 seconds

The following Application.xml will enable key rotation at Application level :

```
<Application>
    <StreamMana ger>
        <Live>
            <AssumeAbsoluteTime>true</AssumeAbsoluteTime>
            <PublishTimeout>0</PublishTim eout>
            <Adj ustForZeroTimeStampMessages>2</AdjustForZeroTimeStampMessages>
            <AdjustForRecordingRollover>false</AdjustForRecordingRollover>
        </Live>
    </StreamManager >
    <HDS>
        <Recording>
            <ContentProtection enabled="true">
            <ProtectionScheme>PHDS</ProtectionScheme>
            <PHDS>
                <VideoEncryptionLevel>2</VideoEncryptionLevel>
                <OutputProtection>None</OutputProtection>
                <PlaybackExpiration>24Hours</PlaybackExpiration>
                <EnableKeyRotation>true</EnableKeyRotation>
                <KeyRotationInterval>500</KeyRotationInterval>
            </PHDS>
            </ContentProtection>
        </Recording>
<   /HDS>
</Application>
```

Event level - PHDS

Parameter	Description	Default value
Recording/ContentProtection/PHDS/EnableKeyRotation	Whether to use Key Rotation with PHDS protection scheme	false
Recording/ContentProtection/PHDS/KeyRotationInterval	Key rotation interval to be used (in seconds), when enabling key rotation.	900 seconds

The following Event.xml will enable key rotation at Event level :<Event>

```
<Event>
    <EventID>liveevent</EventID>
    <Recording>
        <FragmentDuration>4000</FragmentDuration>
        <SegmentDuration>400000</SegmentDuration>
        <DiskManagementDuration>3</DiskManagementDuration>
        <ContentProtection enabled="true">
            <ProtectionScheme>PHDS</ProtectionScheme>
            <PHDS>
                <VideoEncryptionLevel>2</VideoEncryptionLevel>
                <OutputProtection>None</OutputProtection>
                <PlaybackExpiration>24Hours</PlaybackExpiration>
                <EnableKeyRotation>true</EnableKeyRotation>
                <KeyRotationInterval>500</KeyRotationInterval>
            </PHDS>
</ContentProtection>
</Recording>
</Event>
```

Disable JIT encryption for F4F content

When PHDS/Adobe Access protection is enabled, the server ingests a stream and packages it into F4F stream data. The unencrypted F4F data is taken as source and encrypted using the PHDS/Adobe Access configurations.In order to force the server to store the ingested stream as encrypted F4F data, and disable the just-in-time encryption of the F4F data, a special configuration is required.

The following table contains the configuration directive for enabling and disabling JIT encryption at server level:

httpd.conf tags:

Directive	Description	Default value
`HttpStreamingJITEncryption`	To disable just in time encryption, set the value to "false"	true

<AMS-Install>conf/_defaultRoot_/_defaultVHost_/Application.xml tags:

Directive	Description	Default value
`HDS/Recording/JITEncryption`	To disable just in time encryption, set the value to "false"	false

***Note:** The tags `HttpStreamingJITEncryption` and `JITEncryption` both must be set to false to disable JIT encryption.*

When JITEncryption is set to false:

- Specify server level encryption settings (PHDS/Adobe Access) at <*AMS-Install*>conf/_defaultRoot_/_defaultVHost_/Application.xml.
- The ingested stream is stored as encrypted F4F content. So, DRMmeta file is stored on the server inside the F4F content.

The following configurations in httpd.conf will disable JIT encryption server wide:

```
<Location /hds-live>
    HttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications"
    HttpStreamingContentPath "../applications" HdsFmsDirPath ".."
    HttpStreamingF4MMaxAge 2
    HttpStreamingBootstrapMaxAge 2
    HttpStreamingDrmmetaMaxAge 3600
    HttpStreamingFragMaxAge -1
    HttpStreamingJITEncryption false
    Options -Indexes FollowSymLinks
</Location>
```

The following configurations for <*AMSInstall*>conf/_defaultRoot_/_defaultVHost_/Application.xml enables PHDS protection:

```
<Application>
    <!-- This section provides th e means to control the  behavior  of  -->
    <!-- application-specific HTTP dynamic s treaming f unctionality. -->
    <HDS>
        <!-- This section cont rols the behavior of HTTP live recording -->
        <Recording>
            <!-- The enabled attribute can be set to  "true", "false" or "allow".     -->
            <!-- Content prote cted is enabled when the attribute is  set to "true",  -->
            <!-- and disabled when set to "false".                                   -->
            <!-- If enabled is set to "allow", only then Event.xml have right  to    -->
            <!-- override the ContentProtection tag completely. And none of the -->
            <!-- settings inside the ContentProtection here will be used.  And if   -->
            <!-- ContentProtection is also not specified in Event.xml, content      -->
            <!-- protection will be disabled by default.                            -->
        <JITEncryption>false</JITEncryption>
        <ContentProtection enabled="true">
            <ProtectionScheme>PHDS</ProtectionScheme>
            <PHDS>
                <CommonKeyFile>common-key.bin</CommonKeyFile>
                <VideoEncryptionLevel>2</VideoEncryptionLevel>
                <PlaybackExpiration>24Hours</PlaybackExpiration>
                <OutputProtection>None</OutputProtection>
            </PHDS>
        </ContentProtection>
    </Recording>
    </HDS>
</Application>
```

The following configurations at `<AMS-Install>`conf/_defaultRoot_/_defaultVHost_/Application.xml enables Adobe Access protection:

```
<Application>
    <!-- This section provides t he means to control the behavior  of - ->
    <!-- application-specific HTTP dynamic streaming fu nctionality. -->
    <HDS>
        <!-- This section controls the behavior of HTTP live recording -->
        <Recording>
            <!-- The enabled attribute can be set to "true", "false" or "allow".     -->
            <!-- Content protection is enabled when the attribute is set to "true ",  -->
            <!-- and disabled when set to "false".                                   -->
            <!-- If enabled is set to "allow", then Event.xml will -->
            <!-- override the ContentProtection tag completely.  And none of the    -->
            <!-- settings inside the ContentProtection will be used.  And if   -->
            <!-- ContentProtection is not specified in Event.xml, then content      -->
            <!-- protection will be disabled by default.                            -->
        <JITEncryption>false</JITEncryption>
            <ContentProtection enabled="true">
                <ProtectionScheme>FlashAccessV2</ProtectionScheme>
                < FlashAccessV2>
                    <ContentID>liveevent</ContentID>
                    <CommonKeyFile>common-key.bin</CommonKeyFile>
                    <LicenseServerURL>
                        http://<aaxs-test-server>/
                    </LicenseServerURL>
                    <TransportCertFile>
                        aaxs-test-server-trnsCert.der
                    </TransportCertFile>
                    <LicenseServerCertFile>
                        aaxs-test-server-licCert.der</LicenseServerCertFile>
                    <PackagerCredentialFile>
                        aaxs-test-server-pkgrCert.pfx
                    </PackagerCredentialFile>
                    <PackagerCredentialPassword>pwd=</PackagerCredentialPassword>
                    <PolicyFile>sample_policy.pol</PolicyFile>
                </ FlashAccessV2>
            </ContentProtection>
        </Recording>
    </HDS>
</Application>
```

Configure system for encrypted live stream in HLS and HDS

You do not need two different recording applications for HDS and HLS if JIT encryption is ON. The live content is stored unencrypted on the disk, and later encrypted dynamically using the HDS or HLS modules of Apache.By default JIT encryption is on unless the `HttpStreamingJITEncryption` and J`ITEncryption` tags are set to false.Publishing one set of streams to Adobe Media Server for delivery with live PHLS and PHDS requires special configuration when JIT Encryption is off.When PHDS is enabled when JIT encryption is off , the server ingests a stream and packages it into encrpypted F4F data. However, PHLS requires unencrypted data as its source. It's not possible to take the encrypted F4F data and encrypt it again for PHLS.To deliver protected content to Flash Player/AIR and iOS devices, configure your encoder to publish to two different applications, one for HDS and one for HLS.

1 Create two copies of the livepkgr application. Name them "livepkgr_hds" and "livepkgr_hls".

2 Configure the <*AMS-Install*>/conf/_defaultRoot_/_defaultVHost_/Application.xml as following:

```
<Application>
    <!-- This section provides the ways to control the behavi or of -->
    <!-- application-specific HTTP dynamic streaming functionality. -->
    < HDS>
        <!-- This section controls the behavior of HTTP live recording -->
        <Recording>
            <!-- The enabled attribute can be set to "true", "false" or "allow" . -->
            <!-- Content protected is enabled when the attribute is set to "true", -->
            <!-- and disabled when set to "false". -->
            <!-- If enabled is set to "allow", only then Event.xml  have right to -->
            <!-- override the ContentProtection tag completely.  And none of the -->
            <!-- settings inside the ContentProtection here will be used. And if -->
            <!-- ContentProtection is also not specified in Event.xml, content -->
            <!-- protection  will  be disabled  by  default. -->
        < JITEncryption>false</JITEncryption>
            <ContentProtection enabled="allow">
            </ContentProtection>
        </Recording>
    </HDS>
</Application>
```

3 Configure the <*AMS-Install*>/applications/livepkgr_hds/Application.xml as following:

```
<Application>
    <StreamManager>
        <Live>
            <AssumeAbsoluteTime>true</AssumeAbsoluteTime>
            <PublishTimeout>0</PublishTimeout>
            <AdjustForZeroTimeStampMessages>2</AdjustForZeroTimeStampMessages>
            <AdjustForRecordingRollover>false</AdjustForRecordingRollover>
         </Live>
    </StreamManager>
    <HDS>
        <Recording >
            <ContentProtection enabled="true" >
                <ProtectionScheme>PHDS</ProtectionScheme>
                    <PHDS>
                        <CommonKeyFile>common-key.bin</CommonKeyFile>
                        <VideoEncryptionLevel>2</VideoEncryptionLevel>
                        <PlaybackExpiration>24Hours</PlaybackExpiration>
                        <OutputProtection>None</OutputProtection>
                    </PHDS>
            </ContentProtection>
        </Recording>
    </HDS>
</Application>
```

4 Configure the httpd.conf files as follows:

For PHDS, use the following `Location` directive:

```
<Location /hds-live>
    HttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications/livepkgr_hds"
    HttpStreamingContentPath "../applications/livepkgr_hds"
    HttpStreamingURLSandboxLevel "App"
    HttpStreamingF4MMaxAge 2
    HttpStreamingBootstrapMaxAge 2
    HttpStreamingDrmmetaMaxAge 3600
    HttpStreamingFragMaxAge -1
    HttpStreamingJITEncryption false
    Options -Indexes FollowSymLinks
</Location>
```

For PHLS, use the following Location directive:

```
<Location /hls-live>
    HLSHttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications/livepkgr_hls"
    HttpStreamingContentPath "../applications/livepkgr_hls"
    HttpStreamingURLSandboxLevel "App"
    HLSMediaFileDuration 8000
    HLSSlidingWindowLength 6
    HLSFmsDirPath ".."
    HttpStreamingUnavailableResponseCode 503
    HLSEncryptionScope server
    HLSProtectionScheme PHLS
</Location>
```

5 Restart Apache.

6 Publish streams from Flash Media Live Encoder to the livepkgr_hds and livepkgr_hls applications. Use the stream name **livestream%i?adbe-live-event=liveevent**.

7 The request URL for PHDS is http://<serveruri>/hds-live/_definst_/<liveevent>.f4m and the request URL for PHLS is http://<serveruri>/hls-live/_definst_/<liveevent>.m3u8. Because the directive `HttpStreamingURLSandboxLevel` is set to "App", the request URL doesn't use the application name.

Note: *In this case, copy the common-key.bin from <AMS Install>/creds directory to <AMS Install>/applications/livepkgr_hds/.*

Similarly, by following the above mentioned steps, Adobe Access configurations can also be used with HDS and HLS.

VOD use case

Configure PHDS for on-demand streaming at the following levels:

Server—*rootinstall*/Apache2.2/conf/httpd.conf

Stream—create a jit.conf file and copy it to the same directory as the content.

Getting started

To quickly get started with PHDS, you need to understand the following directives:

Directive	Default value	Description
EncryptionScope	None	Possible values are content and server. When the value is content, PHDS configuration settings in the jit.conf file override settings in the httpd.conf. file. When the value is server, the server uses configuration settings in the httpd.conf file.
ProtectionScheme	None	A string determining the type of protection. For PHDS, use PHDS.

The simplest way to configure on-demand PHDS is to uncomment two lines in the Apache httpd.conf file:

```
<IfModule jithttp_module>
<Location /hds-vod>
    HttpStreamingJITPEnabled true
    HttpStreamingContentPath "../webroot/vod"
    JitFmsDirPath ".."
    Options -Indexes FollowSymLinks

# Uncomment the following directives to enable encryption
# for this location.
   EncryptionScope server
   ProtectionScheme phds
</Location>
</IfModule>
```

***Note:** This configuration will enable PHDS at the server level.*

The sample1_1500kbps.f4v media file comes with the default installation of AMS under <*root-install*>/webroot. Play back the media file sample1_1500kbps.f4v using the following URI:http://<*server-ip*>/hds-vod/sample1_1500kbps.f4v.f4m

Detailed configuration

The following sections provides details configurations for both PHDS and Adobe Access.

Server level

The following sections explain how content protection can be applied across the server:

Common configurations

Directive	Default value	Description
`EncryptionScope`	content	Possible values are `content` and `server`. When the value is `content`, PHDS configuration settings in the jit.conf file override settings in the httpd.conf. file. When the value is `server`, the server uses configuration settings in the httpd.conf file. Serverwide configuration that sets encryption policy.server - ALL content is protected according to the apache configuration (jit.conf is ignored).content - Content is protected/unprotected according the to jit.conf file.off - ALL content are unprotected (jit.conf is ignored) .
`ProtectionScheme`	PHDS	A string determining the type of protection. Possible values are PHDS and FlashAccessV2.

PHDS configurations

Configure the following directives for the jithttp_module in the Apache httpd.conf file:

Directive	Default value	Description
`PHDSCommonKeyFile`	creds/common-key.bin This file is generated during installation.	A common key used to protect content at this location.
`PHDSPlaybackExpiration`	24Hours	The duration within which content playback is available. Possible values are `24Hours` and `Unlimited`
`PHDSOutputProtection`	None	The required hardware Output Protection of media on the client. Possible values are None, BestEffort, and Required.
`PHDSVideoEncryptionLevel`	2	The level of encryption for the content (0-low,1-medium, 2-high). Lower settings provide partial encryption. A subset of the samples (like video keyframes) are encrypted. Partial encryption can improve playback performance on the client, because there are fewer frames to decrypt.

Adobe Access configurations

Directive	Default Value	Description
`JitDrmCommonKeyFile`	None	A common key used to protect content at this location. JitDrmCommonKeyFile path is relative to rootinstall/Apache2.2.

Directive	Default Value	Description
JitDrmLicenseServerURL	None	The URL of the license server used for protecting content.
JitDrmTransportCertFile	None	The transport certificate used for protecting content.
JitDrmPackagerCredentialFile	None	The Packager credential used for protecting content.
JitDrmPackagerCredentialPassword	None	The Packager credential password for the configured packager credential file.
JitDrmPolicyFile	None	Policy for protecting content.

The following example adds a new `Location` directive. Request that include `/phds` serve protected content. This configuration doesn't define `PHDSPlaybackExpiration`, `PHDSVideoEncryptionLevel`, or `PHDSCommonKeyFile`, but relies on their default values:

```
LoadModule jithttp_module modules/mod_jithttp.so
<IfModule jithttp_module>

<Location /phds>
    HttpStreamingJITPEnabled true
    HttpStreamingContentPath "../webroot/vod"
    JitFmsDirPath ".."
    Options -Indexes FollowSymLinks
    EncryptionScope server
    ProtectionScheme phds
</Location>
```

When a media player request content from the /webroot/vod folder, it is protected. For example, request the following URL from the sample video player:

http://localhost:8134/phds/sample1_1500kbps.f4v.f4m

To verify that the content is protected, enter the same URL into the address bar of a web browser. The XML response contains a `<drmAdditionalHeader>` element like the following:

```
 <?xml version="1.0" encoding="UTF-8" ?>
- <manifest xmlns="http://ns.adobe.com/f4m/1.0">
  <id>sample1_1500kbps.f4v</id>
  <streamType>recorded</streamType>
  <duration>114.61450000000001</duration>
  <bootstrapInfo profile="named" id="bootstrap3628">AAABq2Fic3QAAAAAAAA</bootstrapInfo>
  <drmAdditionalHeader drmContentId="sample1_1500kbps.f4v"
id="drmMetadata9839">AgARfEFkZGl0aW9uYWxIZWFkZXIDAAp</drmAdditionalHeader>
- <media streamId="sample1_1500kbps.f4v" url="sample1_1500kbps.f4v"
bootstrapInfoId="bootstrap3628" drmAdditionalHeaderId="drmMetadata9839">
  <metadata>AgAKb25NZXRhRGF0</metadata>
  </media>
  </manifest>
```

Note: *The `<bootstrapInfo>`, `<drmAdditionalHeader>`, and `<metadata>` information has been abridged for readability.*

The following example adds a new `Location` directive. Request that include `/hds-vod-fax` serve protected content through Adobe Access:

```
<Location /hds-vod-fax>
    HttpStreamingJITPEnabled true
    HttpStreamingContentPath "../webroot/vod"
    HttpStreamingJITConfAllowed true
    JitFmsDirPath ".."
    Options -Indexes FollowSymLinks
    EncryptionScope server
    ProtectionScheme FlashAccessV2
    JitDrmCommonKeyFile common-key.bin
    // Common key to be used to protect content at this location. No default
    JitDrmLicenseServerURL http:

    // License server URL used when protecting content at this location. No default
    JitDrmTransportCertFile aaxs-test-server-trnsCert.der
    // Transport certification used when protecting content at this location. No default
    JitDrmLicenseServerCertFile aaxs-test-server-licCert.der
    // License server certificate used when protecting content at this location.
    // No default.
    JitDrmPackagerCredentialFile aaxs-test-server-pkgrCert.pfx
    // Packager credential used when protecting content at this location. No default
    JitDrmPackagerCredentialPassword pwd=
    // Packager credential password for the configured packager credential file.
    // No default
    JitDrmPolicyFile sample_policy.pol
    //Policy to be used when protecting content at this location . No default
</Location>
```

Note: *JitDrmCommonKeyFile takes path relative to <AMS-Install>/Apache2.2.*

Stream level

To configure encryption parameters for individual sets of media, follow the configurations mentioned below.

Common configurations

Element	Default value	Description
`//manifest/hds:content-protection enabled`	false	To enable content protection with Adobe Access or PHDS, set the enabled attribute to `"true"`.
`//manifest/hds:content-protection/hds:protection-scheme`	PHDS	The type of protection. The possible values are `PHDS` and `FlashAccessV2` only. For PHDS, use `PHDS`.

PHDS configurations

Element	**Default value**	**Description**
`//manifest/hds:content-protection/hds:phds/hds:common-key-file`	creds/common-key.bin	Path to a common key file generated when the server installs. The file contains a16-byte/128-bit random key. This path can be absolute or relative to the jit.conf file.
`//manifest/hds:content-protection/hds:phds/hds:video-encryption-level`	2	The level of encryption for the content (0-low,1-medium,2-high). Lower settings provide partial encryption. A subset of the samples (like video keyframes) are encrypted. Partial encryption can improve playback performance on the client because there are fewer frames to decrypt.
`//manifest/hds:content-protection/hds:phds/hds:playback-expiration`	24Hours	The protection policy. The policy determines the duration within which content playback is available. Possible values are `24Hours` and `Unlimited`.
`//manifest/hds:content-protection/hds:phds/hds:output-protection`	None	The required hardware Output Protection of media on the client. Possible values are None, BestEffort, and Required.

Adobe Access configurations

Element	**Default value**	**Description**
`//manifest/hds:content-protection/hds:flash-access/hds:common-key-file`	None	The path to common key file. File contains 16-byte/128-bit random key.The path must be absolute or relative to the jit.conf file.
`//manifest/hds:content-protection/hds:flash-access/hds:content-id`	None	The Content ID to be used for content protection. If not specified, the salt is the filename. If specified, the salt is shared with all content in the directory.
`//manifest/hds:content-protection/hds:flash-access/hds:license-server-url`	None	The License Server URL.
`//manifest/hds:content-protection/hds:flash-access/hds:transport-cert-file`	None	The path to transport cert file. The file is in DER format.The path should be absolute or relative to the jit.conf file.
`//manifest/hds:content-protection/hds:flash-access/hds:license-server-cert-file`	None	The path to license cert file. File is in DER format.The path should be absolute or relative to the jit.conf file.

Element	Default value	Description
`//manifest/hds:content-protection/hds:flash-access/hds:packager-credential-file`	None	The path to packager credential cert file. File is in PFX format.The path should be absolute or relative to the jit.conf file.
`//manifest/hds:content-protection/hds:flash-access/hds:packager-credential-password`	None	The packager credential password.
`//manifest/hds:content-protection/hds:flash-access/hds:policy-file`	None	The path to a policy file. File is in Adobe Access policy format.The path should be absolute or relative to the jit.conf file.

The following httpd.conf file sets `EncryptionScope` to `content`. This setting tells the server that configuration settings in the jit.conf file override settings in the httpd.conf file. Use this setting to configure PHDS/AdobeAccess for individual sets of media.

```
LoadModule jithttp_module modules/mod_jithttp.so
<IfModule jithttp_module>
<Location /hds-vod>
    HttpStreamingJITPEnabled true
    HttpStreamingContentPath "../webroot/vod"
    JitFmsDirPath ".."
    Options -Indexes FollowSymLinks
    EncryptionScope content
</Location>
```

The following is the accompanying jit.conf file, which is in the same directory as the on-demand media files (/webroot/vod), which will enable PHDS:

```
<?xml version="1.0" encoding="utf-8"?>
<manifest xmlns="http://ns.adobe.com/f4m/1.0" xmlns:hds="http://ns.adobe.com/hds-package/1.0>

    <frame-rate>29.97</frame-rate>
    <frames-per-keyframe-interval>60</frames-per-keyframe-interval>
    <hds:content-protection enabled="true">
        <hds:protection-scheme>phds</hds:protection-scheme>
            <hds:phds>
                <hds:common-key-file>
                    C:\Program Files\Adobe\Adobe Media Server 5\creds\common-key.bin
                </hds:common-key-file>
                <hds:video-encryption-level>0</hds:video-encryption-level>
                <hds:playback-expiration>unlimited</hds:playback-expiration>
            </hds:phds>
    </hds:content-protection>

</manifest>
```

The following is the accompanying jit.conf file, which is in the same directory as the on-demand media files (/webroot/vod), which will enable Adobe Access:

```
<?xml version="1.0" encoding="utf-8"?>
<manifest xmlns="http://ns.adobe.com/f4m/1.0" xmlns:hds="http://ns.adobe.com/hds-
package/1.0">
        <hds:FlashAccessV2>
            <hds:content-id>jit_fax2</hds:content-id>
            <hds:common-key-file>common-key.bin</hds:common-key-file>
            <hds:license-server-url>http://<aaxs-test-server></hds:license-server-url>
            <hds:transport-cert-file>aaxs-test-server-trnsCert.der</hds:transport-cert-file>
            <hds:license-server-cert-file>aaxs-test-server-licCert.der</hds:license-server-
cert-file>
            <hds:packager-credential-file>aaxs-test-server-pkgrCert.pfx</hds:packager-
credential-file>
            <hds:packager-credential-password>pwd=</hds:packager-credential-password>
            <hds:policy-file>sample_policy.pol</hds:policy-file>
        </hds:FlashAccessV2>
    </hds:content-protection>
</manifest>
```

Note: *-key-file takes path relative to <AMS-Install>/webroot/vod.*

Key rotation

Adobe Media Server 5 supports Key Rotation for protected HTTP Dynamic Streaming when used with Adobe Access and PHDS. You can encrypt content packaged with AMS 5 using a set of keys. You can periodically change the encryption key and specify how often the content encryption key is to be changed.

Adobe Access Settings

Parameter	Description	Default value
`JitDrmEnableKeyRotation`	Whether to use Key Rotation with FAXS protection scheme. In this case, randomly generated keys are used.	false
`JitDrmKeyRotationInterval`	Key rotation interval to be used (in seconds), when enabling key rotation.	900 seconds

The following httpd.conf will enable key rotation at server level :

```
<Location /hds-vod>
    HttpStreamingJITPEnabled true
    HttpStreamingContentPath "../webroot/vod"
    HttpStreamingJITConfAllowed true
    JitFmsDirPath ".."
    Options -Indexes FollowSymLinks
    EncryptionScope server
    ProtectionScheme  FlashAccessV3
    JitDrmCommonKeyFile ../creds/common-key.bin
    JitDrmLicenseServerURL http://ip-address:8090
    JitDrmTransportCertFile dme/transport-cert-file.der
    JitDrmLicenseServerCertFile dme/transport-cert-file.der
    JitDrmPackagerCredentialFile dme/transport-cert-file.pfx
    JitDrmPackagerCredentialPassword kY2IUPnQuG0=
    JitDrmPolicyFile dme/local_chain.pol
    JitDrmEnableKeyRotation true
    JitDrmKeyRotationInterval 16
</Location>
```

Jit.conf

```
<?xml version="1.0" encoding="utf-8"?>
<manifest xmlns="http://ns.adobe.com/f4m/1.0" xmlns:hds="http://ns.adobe.com/hds-
package/1.0">
        <hds:content-protection enabled="true">
        <hds:protection-scheme> FlashAccessV3 </hds:protection-scheme>
        <hds:Flashaccessv3>
            <hds:content-id>jit_fax3</hds:content-id>
            <hds:common-key-file>../../creds/common-key.bin</hds:common-key-file>
            <hds:license-server-url>http://10.192.37.195:8090/</hds:license-server-url>
            <hds:transport-cert-file>../dme/transport-cert-file.der</hds:transport-cert-
file>
            <hds:license-server-cert-file>../dme/transport-cert-file.der</hds:license-
server-cert-file>
            <hds:packager-credential-file>../dme/transport-cert-file.pfx</hds:packager-
credential-file>
            <hds:packager-credential-password>kY2IUPnQuG0=</hds:packager-credential-
password>
            <hds:policy-file>../dme/local_chain.pol</hds:policy-file>
            <hds:enable-key-rotation>true</hds:enable-key-rotation>
            <hds:key-rotation-interval>900</hds:key-rotation-interval>
        </hds:Flashaccessv3>
        </hds:content-protection>
</manifest>
```

PHDS - Settings

This section explains key rotation settings for PHDS.

Parameter	Description	Default value
PHDSEnableKeyRotation	Whether to use Key Rotation with PHDS protection scheme. In this case, randomly generated keys are used.	false

Parameter	Description	Default value
`PHDSKeyRotationInterval`	Key rotation interval to be used (in seconds), when enabling key rotation.	900 seconds

The following httpd.conf will enable key rotation at server level :

```
<Location /hds-vod>
    HttpStreamingJITPEnabled true
    HttpStreamingContentPath "../webroot/vod"
    HttpStreamingJITConfAllowed true
    JitFmsDirPath ".."
    Options -Indexes FollowSymLinks
    EncryptionScope server
    ProtectionScheme  PHDS
    PHDSCommonKeyFile ../creds/common-key.bin
    PHDSEnableKeyRotation true
    PHDSKeyRotationInterval 16
</Location>
```

Jit.conf

```
<?xml version="1.0" encoding="utf-8"?>
<manifest xmlns="http://ns.adobe.com/f4m/1.0" xmlns:hds="http://ns.adobe.com/hds-
package/1.0">
    <hds:content-protection enabled="true">
        <hds:protection-scheme> PHDS </hds:protection-scheme>
            <hds:PHDS>
                <hds:content-id>jit_phds</hds:content-id>
                <hds:common-key-file>../../creds/common-key.bin</hds:common-key-file>
                <hds:enable-key-rotation>true</hds:enable-key-rotation>
                <hds:key-rotation-interval>900</hds:key-rotation-interval>
            </hds:PHDS>
    </hds:content-protection>
</manifest>
```

License chaining

Adobe Media Server will support embedding leaf licenses in the DRM metadata from the policy generated using a chained license. Adobe Media Server will need the license server credential and the credential password configured so that the root license from the policy can be used to encrypt the CEK contained in the embedded leaf license.

If the configuration for embedding the leaf license is turned off, Adobe Media Server will still support such a policy except that the leaf license will not be embedded in the DRM metadata.

Note: *The support will be limited to a single license server credential and credential-password pair.*

The following table provides the required configuration:

Parameter	Description	Default value
JitDrmEmbedLeafLicense	Enables embedding of leaf licenses in DRM metadata. Possible values are "true" or "false". ***Note:*** *The policy file must be created using a chained license.*	false
JitDrmLicenseServerCredentialFile	Required if HdsDrmEmbedLeafLicense is set to true. The license server credential used when protecting content at this location.	NA
JitDrmLicenseServerCredentialPassword	Required if HdsDrmEmbedLeafLicense is set to true. The license server credential password for the configured license server credential file.	NA

The following httpd.conf will enable key rotation at server level :

```
<Location /hds-vod>
    HttpStreamingJITPEnabled true
    HttpStreamingContentPath "../webroot/vod"
    HttpStreamingJITConfAllowed true
    JitFmsDirPath ".."
    Options -Indexes FollowSymLinks
    EncryptionScope server
    ProtectionScheme  FlashAccessV3
    JitDrmCommonKeyFile ../creds/common-key.bin
    JitDrmLicenseServerURL http://ip-address:8090
    JitDrmTransportCertFile dme/transport-cert-file.der
    JitDrmLicenseServerCertFile dme/transport-cert-file.der
    JitDrmPackagerCredentialFile dme/transport-cert-file.pfx
    JitDrmPackagerCredentialPassword kY2IUPnQuG0=
    JitDrmPolicyFile dme/local_chain.pol
    JitDrmEmbedLeafLicense true
    JitDrmLicenseServerCredentialFile dme/transport-cert-file.pfx
    JitDrmLicenseServerCredentialPassword kY2IUPnQuG0=
</Location>
```

Jit.conf

```
<?xml version="1.0" encoding="utf-8"?>
<manifest xmlns="http://ns.adobe.com/f4m/1.0" xmlns:hds="http://ns.adobe.com/hds-
package/1.0">
        <hds:content-protection enabled="true">
        <hds:protection-scheme> FlashAccessV3 </hds:protection-scheme>
        <hds:Flashaccessv3>
            <hds:content-id>jit_fax3</hds:content-id>
            <hds:common-key-file>../../creds/common-key.bin</hds:common-key-file>
            <hds:license-server-url>http://localhost:8090/</hds:license-server-url>
            <hds:transport-cert-file>../dme/transport-cert-file.der</hds:transport-cert-
file>
            <hds:license-server-cert-file>../dme/transport-cert-file.der</hds:license-
server-cert-file>
            <hds:packager-credential-file>../dme/transport-cert-file.pfx</hds:packager-
credential-file>
            <hds:packager-credential-password>kY2IUPnQuG0=</hds:packager-credential-
password>
            <hds:policy-file>../dme/local_chain.pol</hds:policy-file>
            <hds:embed-leaf-license>true</hds:embed-leaf-license>
            <hds:license-server-credential-file>../dme/transport-cert-file.pfx</hds:license-
server-credential-file>
            <hds:license-server-credential-password>kY2IUPnQuG0=</hds:license-server-
credential-password>
            </hds:Flashaccessv3>
        </hds:content-protection>
</manifest>
```

SWF verification for Protected HTTP Dynamic Streaming

SWF verification prevents unauthorized SWF files from accessing content. To use SWF verification, you must enable Protected HTTP Dynamic Streaming (PHDS).

Create a list of authorized SWF files, called a *whitelist*. These files are specified in the embedded license and sent to the client inside the DRM metadata. On the client, SWF verification is enforced by Adobe Access inside of Flash Player and AIR.

To create the whitelist, use Whitelist tool (*rootinstall*/tools/Whitelist).

Workflow

1. Enable PHDS.
2. Use the whitelist tool to generate a whitelist of authorized SWF files. The whitelist file can have any name. It must have the .whitelist or .airwhitelist extension.
3. Copy the whitelist to the server.
4. Enable SWF verification and indicate the location of the whitelist in the following locations:
    - (Live)—Application.xml or Event.xml
    - (On-demand)—httpd.conf or jit.conf
5. Publish a stream to the livepkgr application on Adobe Media Server.
6. Request a stream from an OSMF media player. The syntax of the request URL does not change for SWF verification.

    The server embeds the SWF hashes from the whitelist into the .drmmeta file. Flash Player attempts to verify the SWF hash during DRM authentication.

7 (Live) The server looks for the whitelist in the following order:

  a The application folder. (The default application for live HTTP streaming is *rootinstall*/applications/livepkgr).

  b A path in the `/SWFVerification/WhitelistFolder` element of Application.xml

  c A path in the `/SWFVerification/WhitelistFolder` element of Event.xml

8 (On-demand) The server looks for the whitelist in the httpd.conf/jit.conf file in the same folder as the on-demand content.

If the hashes don't match, Flash Player throws an runtime error (3310) and the OSMF media player stops requesting fragments.

SWF verification configurations for live PHDS

To enable SWF verification for live PHDS, enable PHDS at the server level (httpd.conf), the application level (Application.xml) or the event level (Event.xml).

Configure SWF verification for live HDS at the server level (httpd.conf)

Add the following elements to the `hds-live` directive to enable SWF verification:

Element	Description	Default
`PHDSSWFVerification`	The container for SWF verification configuration. To enable SWF verification, set the enabled attribute to "true".	"false"
`PHDSSWFWhiteListFolder`	Specify the location of SWF whitelist	The application folder of the live event.

Configure SWF verification for live HDS at the application level (Application.xml) or at the event level (Event.xml).

In Application.xml, `SWFVerification` is located at `//Application/HDS/Recording/ContentProtection/PHDS/SWFVerification`.In Event.xml, `SWFVerification` is located at `//Event/Recording/ContentProtection/PHDS/SWFVerification`.

Element	Description	Default
`/SWFVerification`	The container for SWF verification configuration. To enable SWF verification, set the `enabled` attribute to `"true"`.	"false"
`/SWFVerification/WhiteListFolder`	A path to the folder containing the whitelist. The folder can contain more than one whitelist file. The path can be absolute or relative. A relative path in the Application.xml file is relative to the application folder. A relative path in the Event.xml file is relative to the event folder. Backwards relative paths are not supported for security reasons. This configuration is optional. If no value is given, the server looks in the application folder of the live event.	The application folder of the live event.

Configure the following settings in the Apache httpd.conf file to configure cache control for the bootstrap, fragment, manifest and drmmeta responses:

- `HttpStreamingBootstrapMaxAge`
- `HttpStreamingFragMaxAge`
- `HttpStreamingF4MMaxAge`
- `HttpStreamingDrmmetaMaxAge`

For detailed information about each configuration, see "Configure live and on-demand HTTP Streaming at the server level (httpd.conf)" on page 76.

SWF verification configurations for on-demand PHDS

SWF verification is configured under PHDS. To enable SWF verification, enable PHDS. You can enable on-demand PHDS at the server level (httpd.conf) or at the stream level (jit.conf).

Configure SWF verification for on-demand PHDS at the server level (httpd.conf) or at the stream level (jit.conf).

Use the following elements to enable and configure SWF verification in the httpd.conf file:

Element	**Description**	**Default**
PHDSSWFVerification	The container for SWF verification configuration. To enable SWF verification, set the enabled attribute to `"true"`.	`"false"`
PHDSSWFWhiteListFolder	Optional setting to specify where the SWF whitelist can be found. The folder can contain more than one whitelist files. This can be overridden by jit.conf if the Apache configuration is overridable. This configuration is optional. If no value is given, the server looks in the folder containing the jit.conf file.	The folder containing the media.

Use the following elements to enable and configure SWF verification in the jit.conf file. Copy the jit.conf file to the same directory as the on-demand media.

Element	**Description**	**Default**
`//manifest/hds:content-protection/hds:phds/hds:swf-verification`	The container for SWF verification configuration. To enable SWF verification, set the `enabled` attribute to `"true"`.	`"false"`
`//manifest/hds:content-protection/hds:phds/hds:swf-verification/hds:white-list-folder`	A path to the folder containing the whitelist. The folder can contain more than one whitelist file. The path can be absolute or relative. A relative path is relative to the folder containing the jit.conf file. Backwards relative paths are not supported for security reasons. This configuration is optional. If no value is given, the server looks in the folder containing the jit.conf file.	The folder containing the media.

Whitelist tool

Use the whitelist tool to generate a list of verified SWF and AIR files. The server uses the whitelist to perform SWF verification for Flash Player and AIR applications.

The whitelist tool takes SWF files, AIR certificate files, and AIR signature files and creates a SHA256 hash for each file. The tool writes the hashes as Base64 encoded text to one or more text files and outputs the text files. The text files use the filename extensions .whitelist and .airwhitelist.

The whitelist tool is located in the following directory:

***rootinstall*/tools/Whitelist**

Use the following command line syntax to run the whitelist tool:

```
whitelist --in <file|dir> [--outDir <output dir>] [--out <output file>] [--version]
```

The following table lists the command line options and arguments for the whitelist tool:

Option	Optional	Description
`--in <file|dir>`	No	A SWF file, an AIR signature file, or an AIR certificate file. A directory containing SWF files. The `dir` parameter does not support AIR files. To specify multiple files or directories, use multiple `--in` options. For SWF files, the tool outputs a file with the extension .whitelist. For AIR signature and certificate files, the tool outputs a file with the extension .airwhitelist.
`--log <file|dir>`	Yes	An existing directory path where default whitelist.properties file is present or the full path name to the properties file. Customize logging in the .properties file. The whitelist tool supports log4j Apache logging. By default, logging messages are routed to the console. To reroute them, use the `--log` option.
`--out <output file>`	Yes	The name for the .whitelist file and the .airwhitelist file. If `--out` is not specified, creates .whitelist and .airwhitelist files for each .swf file and .xml file. If `--out` is specified, `--outDir` is ignored and the file is saved to the directory the tool is being run from.
`--outDir <outputdir>`	Yes	Creates an output directory and saves the .whitelist file to the directory. If `--outDir` is not specified, the .whitelist files and .airwhitelist files are created in the directory the tool is being run from. If `--outDir` is a relative path, it is relative to the directory the tool is being run from.
`--version`	Yes	Prints the SWF verification version number in the .whitelist file.

The following table lists examples of running the whitelist tool:

Example	Result
`whitelist --in foo.swf --in bar.swf`	Creates a foo.swf.whitelist and a bar.swf.whitelist in the current directory.
`whitelist --in signature.xml --in bar.swf`	Creates signature.xml.airwhitelist and bar.swf.whitelist in the current directory.
`whitelist --in foo.swf --in mydir` In this example, mydir is a directory containing bar.swf.	Creates a foo.swf.whitelist and a bar.swf.whitelist in the current directory.
`whitelist --in signature.xml --in mydir` In this example, mydir is a directory containing bar.swf.	Creates a signature.xml.airwhitelist and a bar.swf.whitelist in the current directory.
`whitelist --in foo.swf --in bar.swf --outDir outputdir`	Creates an outputdir/foo.swf.whitelist file and an outputdir/bar.swf.whitelist file.
`whitelist --in signature.xml --in bar.swf --outDir outputdir`	Creates an outputdir/signature.xml.airwhitelist file and an outputdir/bar.swf.whitelist file.
`whitelist --in foo.swf --in mydir --out outputfile` In this this example, mydir is a directory containing bar.swf.	Creates an outputfile.whitelist file in the current directory containing hashes for foo.swf and mydir/bar.swf.
`whitelist --in signature.xml --in mydir --out outputfile` In this this example, mydir is a directory containing bar.swf.	Creates a an outputfile.airwhitelist file containing hashes for signature.xml. Creates an outputfile.whitelist file contaning hashes for bar.swf. Both files are created in the current directory.
`whitelist --in foo.swf --in mydir -out outputfile -outDir outputdir` This example, mydir is a directory containing bar.swf.	Creates an outputfile.whitelist in the current directory containing a hash for foo.swf and mydir/bar.swf. Warning: When the `--out` option is specified, the tool ignores the `--outDir` option.
`whitelist --in signature.xml --in mydir --out outputfile --outDir outputdir` In this example, mydir is a directory containing bar.swf.	Creates an outputfile.airwhitelist file that contains the hashes for signature.xml. Creates an outputfile.whitelist file that contains hashes for mydir/bar.swf. Both files are created in the current directory. Warning: When the `--out` option is specified, the tool ignores the `--outDir` option.
`whitelist --version`	Displays "version 1.0".

If an input files has the same name as a previously input file, both files are added to the whitelist.

```
whitelist --in c:\myfolder\signature.xml --in c:\yourfolder\signature.xml --outDir
c:\out\signature.xml
```

The following is the output:

```
# c:\myfolder\signature.xml
XXXXXXXXXXXXXXXXXXXXXX
# c:\yourfolder\signature.xml
XXXXXXXXXXXXXXXXXXXXXXXXXXX
```

The following is the whitelist format for an individual hash:

```
# foo.swf
PGfcEwgUKWScivIRucIwG5jT
```

The following is the whitelist format for an AIR file:

```
# C:\air\signatures.xml
A167FBF93528C87BBCDAC2B8CD0829479DDA6912.2
```

The following is the whitelist format for multiple hashes when using the `--out` option:

```
# foo.swf
PGfcEwgUKWScivIRucIwG5jT

# bar.swf
TcsQWLLi7h7WNjHqcLzzl0J15Srvdzkz2inCTKQLOHw=

# mydir/bar.swf
TcsQWLLi7h7WNjHqcLzzl0J15Srvdzkz2inCTKQLOHw=
```

Configuring content protection for HLS

Use Adobe Media Server 5 to serve protected content over HTTP to devices that support Apple HTTP Live Streaming.

The Adobe Media Server installer generates the required certificates and keys to the *rootinstall*/creds directory. To generate new keys, use the scramble tool. See Scramble tool.

Overview

The content can be protected using three modes:

- Vanilla
- PHLS
- Adobe Access 4.0

To enable a specific encryption scheme, use the HLSProtectionScheme directive.

Vanilla

Vanilla mode is used for plain AES encryption.

PHLS

PHLS mode is a non-DRM solution. You do not need to set up a license or key server. The key is always served in a local mode.

Adobe Access

Adobe Access mode offers a complete DRM solution. It supports all the Adobe Access 3.0 features, along with remote key serving for HLS. Local key serving mode also works with Adobe Access 2.0 or higher license servers. The remote key serving mode works only with an Adobe Access 4.0 compliant server.

Adobe Access SDK is a Digital Rights Management (DRM) platform that makes it possible to protect and securely deliver video and audio content for playback on consumer devices such as personal computers. Adobe Access is a flexible platform that enables content owners to protect their content and maintain control over distribution. Content owners can protect and manage their rights by creating licenses for each digital media file, ensuring that a wide variety of the highest-quality content is made available to consumers.

Adobe Access supports a wide range of business models, including video on demand, rental, and electronic sell-through. You can distribute content protected with Adobe Access by streaming through Adobe Media Server software, offering progressive download via HTTP using Adobe's HTTP Dynamic Streaming technology, or permitting downloads to a content library for local playback at the consumer's convenience.

To enable DRM support for HTTP Live Streaming, use Adobe Access iOS library. The policy files generated for Adobe Access 2.0 and Adobe Access 3.0 will also work in local key serving mode.

AMS supports different content encryption keys for content at the different levels (server, application, stream, and event). The keys are generated according to the location of the content and the location of the Common Key.

Key rotation

You can periodically change the encryption key and specify how often the content encryption key is to be changed.

Out-of-Band DRM metadata

The HLS module of AMS supports serving of BER encoded DRM metadata out-of-band. The requested URL format is same as for a playlist except that the URL format has .drmmeta instead of .m3u8 at the end of the URL. The metadata embedded in the m3u8 file is base64-encoded but the metadata served out-of-band in the .drmmeta file is binary data.

Player binding

Adobe Media Server supports whitelist-based player binding when the protection scheme is PHLS. This is similar to the HTTP SWF Verification.

License chaining

Adobe Media Server will support embedding leaf licenses in the DRM metadata from the policy generated using a chained license. Adobe Media Server will need the license server credential and the credential password configured so that the root license from the policy can be used to encrypt the CEK contained in the embedded leaf license.

Live use case

Getting started

To configure PHLS with basic settings, perform the following steps:

1 Navigate to the <*root-install*>/Apache 2.2/conf/ directory. Edit the httpd.conf file and add the following tags under `<Location hls-live>`:

```
<Location /hls-live>
    HLSHttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications"
    HttpStreamingContentPath "../applications"
    HLSMediaFileDuration 8000
    HLSSlidingWindowLength 6
    HLSFmsDirPath ".."
    HttpStreamingUnavailableResponseCode 503
    HLSEncryptionScope server
    HLSProtectionScheme PHLS
</Location>
```

***Note:** This configuration will enable PHLS at the server level.*

2 Publish a live stream called “livestream?adbe-live-event=liveevent” to livepkgr.

3 Playback the stream using the URI http://<*server-ip*>:8134/hls-live/livepkgr/_definst_/liveevent/livetsream.m3u8

Detailed configuration

The following sections provides detailed configurations.

Server level

You can configure HLS at the server level to apply content protection across all deployed applications.

Vanilla

The following table contains the directives for the hlshttp_module in the Apache httpd.conf file:

Directive	Default	Description
`HLSEncryptionScope`	Off	Defines the encryption scope. The following are possible values: `server` —Apache encryption settings are applied to all content. The server ignores content-specific encryption configurations in Event.xml and Application.xml (live) and jit.conf (on-demand). `content` —Apache encryption settings are ignored. The server uses encryption settings from Event.xml or Application.xml (live) or from jit.conf (on-demand). `Off` —Encryption is off for the whole server.
`HLSEncryptCipherKeyFile`	None	The path of the default cipher key used to encrypt the content.
`HLSEncryptKeyURI`	None	The URI that the client uses to fetch the encryption key.

Publishing and playback

1 Open the *rootinstall*/Apache2.2/conf/httpd.conf file and locate the hlshttp_module:

```
<IfModule hlshttp_module>
    ...
<Location /hls-live>
    ...
```

2 Uncomment the following:

```
# Uncomment the following directives to enable encryption
# for this location:
    HLSEncryptionScope server
    HLSEncryptionCipherKeyFile "../creds/liveeventkey.bin"
    HLSEncryptKeyURI "https://<ServerName>/hls-key/liveeventkey.bin"
```

Substitute the fully qualified domain name of your Adobe Media Server for the `<ServerName>` parameter.

3 Follow the steps in "Serve encryption keys to the client" on page 176 to configure the server to serve keys with or without SSL. These steps configure the `/hls-key` path in the `HLSEncryptKeyURI` directive.

4 Open Flash Media Live Encoder and publish a stream with the following settings:

- Format—H.264
- Keyframe Frequency—4 seconds
- AMS URL—rtmp://<server-name>/livepkgr
- Stream—encryption?adbe-live-event=encryption

5 Request the following URL from an iOS device:

http://<servername>/hls-live/livepkgr/_definst_/encryption/encryption.m3u8

6 To verify that the stream is encrypted, run the Apple Media Stream Validator Tool on the stream. See Technical Note TN2224.

PHLS and Adobe Access

The following table contains the directives for the hlshttp_module in the Apache httpd.conf file:

Parameter	Required With	Default	Description
HLSProtectionScheme	Adobe Access 4.0, PHLS	Vanilla	Determines the protection scheme used for content. Protection scheme can be Vanilla, AdobeAccessV4 or PHLS. HLSProtectionScheme is effective if encryption is enabled.Use HLSEncryptionScope parameter to determine the status of encryption.
HLSEmbedMetadata	Adobe Access 4.0, PHLS	true for VOD and false for live	(Optional) Enables embedding of metadata in the playlist. The possible values are "true" or "false". Note that false will only work when `HLSMetaPackagingEnabled` is set to true.
HLSMetaPackagingEnabled	Adobe Access 4.0, PHLS	true	(Optional) Enables just in time packaging of metadata for this location. The possible values are "true" or "false". This configuration is not valid for the Vanilla protection scheme.
HLSMetaMaxAge	Adobe Access 4.0, PHLS	60*60 secs (1 hour)	(Optional) Specifies the max-age to set in the Cache-Control header for M3U8 responses. Specified in secs. -1 means no Cache-Control header is set. If not specified, the default value will be assumed. This configuration is used only when the `HLSMetaPackagingEnabled` value is set to true.
PHLSCommonKeyFile	(Optional) PHLS	For PHLS ../creds/commonKey.bin	Contains the name of the Common key file in ../creds/common-key.bin.
HLSDrmContentID/ PHLSContentID	(Optional) PHLS	eventId	Content ID for mapping the license.
HLSDrmLicenseServerURL	Adobe Access 4.0	None	URL of License server used for protecting content
HLSDrmTransportCertFile	Adobe Access 4.0	None	Transport certificate file used for protecting content

Parameter	Required With	Default	Description
HLSDrmLicenseServerCertFile	Adobe Access 4.0	None	File containing license server certificate used for protecting content
HLSDrmPackagerCredentialFile	Adobe Access 4.0	None	File containing Packager credential used for protecting content
HLSDrmPackagerCredentialPassword	Adobe Access 4.0	None	Packager credential password for the configured packager credential file
HLSDrmPolicyFile	Adobe Access 4.0	None	Path and Name of the Policy File to be used for protecting content
HLSDrmKeyServerURL	Adobe Access 4.0	None	Key server URL for embedding in the served playlist.
PHLSOutputProtection	(Optional) PHLS	None	The required hardware Output Protection of media on the client. Possible values are None, BestEffort, and Required.
PHLSPlaybackExpiration	(Optional) PHLS	24Hours	The duration of the time for which the content is available for playback. Possible values are 24Hours and Unlimited.

PHLS

Edit the httpd.conf file and add the following tags under `<Location hls-live>`:

```
<Location /hls-live>
    HLSHttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications"
    HttpStreamingContentPath "../applications"
    HLSMediaFileDuration 10000
    HLSSlidingWindowLength 6
    HLSFmsDirPath ".."
    HttpStreamingUnavailableResponseCode 503
    HLSEncryptionScope  server
    HLSProtectionScheme  PHLS
    PHLSContentID httpd_conf
    PHLSCommonKeyFile  "../creds/liveeventkey.bin"
    PHLSOutputProtection  None
    PHLSPlaybackExpiration  Unlimited
</Location>
```

For details on the configuration elements, see the table mentioned above.

Adobe Access

Edit the httpd.conf file and add the following tags under `<Location hls-live>`:

```
<Location /hls-live>
    HLSHttpStreamingEnabled true
    HttpStreamingLiveEventPath "../applications"
    HttpStreamingContentPath "../applications"
    HLSMediaFileDuration 10000
    HLSSlidingWindowLength 6
    HLSFmsDirPath ".."
    HttpStreamingUnavailableResponseCode 503
    HLSEncryptionScope  server
    HLSProtectionScheme  AdobeAccessV4
    HLSDrmContentID httpd_conf
    HLSDrmCommonKeyFile  "<path to common key file>"
    HLSDrmLicenseServerURL  "<url of license server"
    HLSDrmTransportCertFile  "path to transport certificate file"
    HLSDrmLicenseServerCertFile  "<path to license server certificate file>"
    HLSDrmPackagerCredentialFile  "<path to packager credential file>"
    HLSDrmPackagerCredentialPassword  ??????
    HLSDrmPolicyFile  "<path to policy file"
    HLSDrmKeyServerURL  "<key server url>"
</Location>
```

For details on the configuration elements, see the table mentioned abo

***Note:** For local key delivery it is recommended that HLSDrmKeyServerURL be set to the dummy URL http://faxs.adobe.com.*

For information on publishing and playback, see “Vanilla” on page 159.

Application and Event level

You can also configure HLS at an application or event level.

Both the Application.xml file and the Event.xml file have an HLS container that holds the live vanilla encryption configuration settings. In Application.xml, the container is located under //Application/HDS/HLS. In Event.xml, the container is located under //Event/HLS.

Application level

Vanilla

Element	Default	Description
/HLS	None	Container for content protection settings.

Element	**Default**	**Description**
`/HLS/Encryption`	None	Set the `enabled` attribute to `"allow"` to allow vanilla encryption configurations in the Event.xml file to override settings in the Application.xml file. Set the `enabled` attribute to `"true"` to configure vanilla encryption in the Application.xml file. These configurations apply to all live events in the application. The default value of the `enabled` attribute in the Application.xml file is `"allow"`. The default value in the Event.xml file is `"false"`.
`/HLS/Encryption/KeyFile`	None	The path of the default cipher key used to encrypt the content.
`/HLS/Encryption/KeyURI`	None	The URI that the client uses to fetch the encryption key.

Configure live vanilla encryption at the application level

1 Open the *rootinstall*/Apache2.2/conf/httpd.conf file and locate the hlshttp_module:

```
<IfModule hlshttp_module>
    ...
<Location /hls-live>
    ...
```

2 Uncomment the `HLSEncryptionScope` directive and set it to `content`:

```
# Uncomment the following directives to enable encryption
# for this location:
    HLSEncryptionScope content
#   HLSEncryptionCipherKeyFile "../creds/liveeventkey.bin"
#   HLSEncryptKeyURI "https://<ServerName>/hls-key/liveeventkey.bin"
```

Substitute the IP address or DNS of your Adobe Media Server for the `<ServerName>` parameter.

3 Edit the Application.xml file in the *rootinstall*/applications/livepkgr folder to include the following:

```
<Application>
    <HDS>
        <HLS>
            <Encryption enabled="true">
                <KeyFile>C:\Program Files\Adobe\Adobe Media Server 
5\creds\liveeventkey.bin</KeyFile>
                <KeyURI>http://<server-ip>/hls-key/liveeventkey.bin</KeyURI>
            </Encryption>
        </HLS>
    </HDS>
</Application>
```

4 Follow the steps in "Serve encryption keys to the client" on page 176to configure the server to serve keys with or without SSL. These steps configure the `/hls-key` path in the `KeyURI` directive.

5 Open Flash Media Live Encoder and publish a stream with the following settings:

- Format—H.264
- Keyframe Frequency—4 seconds
- AMS URL—rtmp://<server-name>/livepkgr

- Stream—encryption?adbe-live-event=encryption

6 Request the following URL from an application developed using the SDK on an iOS device:

http://<servername>/hls-live/livepkgr/_definst_/encryption/encryption.m3u8

7 To verify that the stream is encrypted, run the Apple Media Stream Validator Tool on the stream. See Technical Note TN2224.

Note: *To create a live event, create a copy of the livepkgr directory located at rootinstall/applications/livepkgr/events/_definst_/liveevent. The name of the copied directory must be the same as the name of the event.*

PHLS

Edit the Application.xml file in the *rootinstall*/applications/livepkgr folder to include the following:

```
<Application>
    <HDS>
        <HLS>
            <Encryption  enabled="true"  protection-scheme="PHLS" >
                <PHLS>
                    <ContentID>app_event_xml</ContentID>
                    <CommonKeyPath>common.bin</CommonKeyPath>
                    <KeyServerURL>faxs://example.com</KeyServerURL>
                    <OutputProtection>None</OutputProtection>
                    <PlaybackExpiration>Unlimited</PlaybackExpiration>
                </PHLS>
            </Encryption >
        </HLS>
    </HDS>
</Application>
```

For more information about the elements, see the table in the server level configuration.

For information on publishing and playback, see "Vanilla" on page 162.

Adobe Access

Edit the Application.xml file in the *rootinstall*/applications/livepkgr folder to include the following:

```
<Application>
    <HDS>
        <HLS>
            <Encryption  enabled="true"  protection-scheme="AdobeAccessV4" >
                <AdobeAccessV4>
                    <ContentID>app_event_xml</ContentID>
                    <CommonKeyPath>common.bin</CommonKeyPath>
                    <LicenseServerURL>license server url </LicenseServerURL>
                    <TransportCertPath>transport.der</TransportCertPath>
                    <LicenseServerCertPath>server.der</LicenseServerCertPath>
                    <PackagerCredentialPath>
                        production_packager.pfx
                    </PackagerCredentialPath>
                    <PackagerCredentialPwd>??????</PackagerCredentialPwd>
                    <PolicyPath>policy.pol</PolicyPath>
                    <KeyServerURL>http://faxs.adobe.com</KeyServerURL>
                </AdobeAccessV4>
            </Encryption >
        </HLS>
    </HDS>
</Application>
```

For more information about the elements, see the table in the server level configuration.

For information on publishing and playback, see “Vanilla” on page 162.

Event level

Vanilla

1 Open the rootinstall/Apache2.2/conf/httpd.conf file and locate the hlshttp_module:

```
<IfModule hlshttp_module>
    ...
<Location /hls-live>
    ...
```

2 Uncomment the `HLSEncryptionScope` directive and set it to `content`:

```
# Uncomment the following directives to enable encryption
# for this location:
    HLSEncryptionScope content
#   HLSEncryptionCipherKeyFile "../creds/liveeventkey.bin"
#   HLSEncryptKeyURI "https://<ServerName>/hls-key/liveeventkey.bin"
```

Substitute the IP address or DNS of your Adobe Media Server for the `<ServerName>` parameter.

3 Edit the Event.xml file in the *rootinstall*/applications/livepkgr/_definst_/encryption folder to include the following:

```
<Event>
        <HLS>
            <Encryption enabled="true">
                <KeyFile>
                    C:\Program Files\Adobe\Adobe Media Server5
                    \creds\liveeventkey.bin
                </KeyFile>
                <KeyURI>http://<server-ip>/hls-key/liveeventkey.bin</KeyURI>
            </Encryption>
        </HLS>
</Event>
```

4 Follow the steps in "Serve encryption keys to the client" on page 176 to configure the server to serve keys with or without SSL. These steps configure the `/hls-key` path in the `KeyURI` directive.

5 Open Flash Media Live Encoder and publish a stream with the following settings:

- Format—H.264
- Keyframe Frequency—4 seconds
- AMS URL—rtmp://<server-name>/livepkgr
- Stream—encryption?adbe-live-event=encryption

6 Request the following URL from an iOS device:

http://<ServerName>/hls-live/livepkgr/_definst_/encryption/encryption.m3u8

7 To verify that the stream is encrypted, run the Apple Media Stream Validator Tool on the stream. See Technical Note TN2224.

PHLS

Edit the Event.xml file in the *rootinstall*/applications/livepkgr/_definst_/encryption folder to include the following:

```
<Event>
    <HLS>
        <Encryption  enabled="true"  protection-scheme="PHLS" >
            <PHLS>
                <ContentID>app_event_xml</ContentID>
                <CommonKeyPath>common.bin</CommonKeyPath>
                <KeyServerURL>faxs://example.com</KeyServerURL>
                <OutputProtection>None</OutputProtection>
                        <PlaybackExpiration>Unlimited</PlaybackExpiration>
                </PHLS>
       </Encryption >
       </HLS>
</Event>
```

For more information about the elements, see the table in the server level configuration.

For more information on publishing and playback, see "Vanilla" on page 165.

Adobe Access

Edit the Event.xml file in the *rootinstall*/applications/livepkgr/_definst_/encryption folder to include the following:<Event>

```
<Event>
    <HLS>
        <Encryption  enabled="true"  protection-scheme="AdobeAccessV4" >
            <AdobeAccessV4>
                <ContentID>app_event_xml</ContentID>
                <CommonKeyPath>common.bin</CommonKeyPath>
                <LicenseServerURL>license server url </LicenseServerURL>
                <TransportCertPath>transport.der</TransportCertPath>
                <LicenseServerCertPath>server.der</LicenseServerCertPath>
                <PackagerCredentialPath>
                    production_packager.pfx
                </PackagerCredentialPath>
                <PackagerCredentialPwd>??????</PackagerCredentialPwd>
                <PolicyPath>policy.pol</PolicyPath>
                <KeyServerURL>http://faxs.adobe.com</KeyServerURL>
            </AdobeAccessV4>
        </Encryption >
    </HLS>
</Event>
```

For more information about the elements, see the table in the server level configuration.

For more information on publishing and playback, see "Vanilla" on page 165.

Live events

To generate unique content encryption keys (CEKs) for Adobe Access, the URL path (relative to the configured content path) up to the stream, but not including the stream name, is used as the Content ID. For example, Content ID for path http://example.com/hls-live/livepkgr/definst/liveevent/livestream.m3u8 would be livepkgr/definst/liveevent.

VOD use case

Configure PHLS for on-demand streaing at the following levels:

Server—*rootinstall*/Apache2.2/conf/httpd.conf

Stream—create a jit.conf file and copy it to the same directory as the content.

Getting started

To configure PHLS with basic settings, perform the following steps:

❖ Navigate to <*root-install*>/Apache 2.2/conf/. Edit the file httpd.conf and add the tags `HLSEncryptionScope` and `HLSProtectionScheme` under the `<Location /hls-vod>` directive:

```
<Location /hls-vod>
    HLSHttpSt  reamingEna bled true
    HLSMediaFileDuration 8000
    HttpStreamingContentPath "../webroot/vod"
    HLSFmsDirPath ".."
    HLSJITConfAllowed true
    HLSEncryptionScope server
    HLSProtectionScheme PHLS
    Options -Indexes FollowSymLinks
</Location>
```

Note: *This configuration will enable PHLS at the server level with default configurations.*

The sample1_1500kbps.f4v media file comes with the default installation of AMS under <*root-install*>/webroot. You can play the media file using the following URI:http://<*server-ip*>/hls-vod/sample1_1500kbps.f4v.m3u8

Detailed configuration

The following sections provides the detailed configurations.

Server level

You can configure HLS at the server level to apply content protection at server level to all streams requested through the location directives.

Vanilla

Configure the following directives for the hlshttp_module in the Apache httpd.conf file:

Directive	Default	Description
`HLSEncryptionScope`	Off	Defines the encryption scope. The following are possible values: `server` —Apache encryption settings are applied to all content. The server ignores content-specific encryption configurations in Event.xml and Application.xml (live) and jit.conf (on-demand). `content` —Apache encryption settings are ignored. The server uses encryption settings from Event.xml or Application.xml (live) or from jit.conf (on-demand). `Off` —Encryption is off for the whole server.
`HLSEncryptCipherKeyFile`	None	The path of the default cipher key used to encrypt the content.
`HLSEncryptKeyURI`	None	The URI that the client uses to fetch the encryption key. See "Serve encryption keys to the client" on page 176.

To configure vanilla content protection at the server level, set `HLSEncryptionScope` to `server` in the httpd.conf file. This configuration tells the server to use the settings in the httpd.conf file for all requests to this `Location` directive.

1 Open the rootinstall/Apache2.2/conf/httpd.conf file and locate the hlshttp_module:

```
<IfModule hlshttp_module>
    ...
<Location /hls-vod>
    ...
```

2 Uncomment the following:

```
# Uncomment the following directives to enable encryption
# for this location:
    HLSEncryptionScope server
    HLSEncryptionCipherKeyFile "../creds/vodkey.bin"
    HLSEncryptKeyURI "https://<ServerName>/hls-key/vodkey.bin"
```

Substitute the IP address or DNS of your Adobe Media Server for the `<ServerName>` parameter.

3 Follow the steps in "Serve encryption keys to the client" on page 176to configure the server to serve keys with or without SSL. These steps configure the `/hls-key` path in the `HLSEncryptKeyURI` directive.

4 Request the following URL from an iOS device:

http://<ServerName>/hls-vod/sample2_1000kbps.f4v.m3u8

5 To verify that the stream is encrypted, run the Apple Media Stream Validator Tool on the stream. See Technical Note TN2224.

The following table contains the directives for the hlshttp_module in the Apache httpd.conf file:

Parameter	**Required With**	**Default**	**Description**
HLSProtectionScheme	Adobe Access 4.0, PHLS	Vanilla	Determines the protection scheme used for content. Protection scheme can be Vanilla, AdobeAccessV4 or PHLS. HLSProtectionScheme is effective if encryption is enabled.Use HLSEncryptionScope parameter to determine the status of encryption.
HLSDrmCommonKeyFile	Adobe Access 4.0, (Optional) PHLS	For PHLS ../creds/commonKey.bin	Contains the name of the Common key file in ../creds/common-key.bin.
HLSDrmContentID / PHLSContentID	(Optional) Adobe Access 4.0, (Optional) PHLS	eventId	Content ID for mapping the license.
HLSDrmLicenseServerURL	Adobe Access 4.0	None	URL of License server used for protecting content
HLSDrmTransportCertFile	Adobe Access 4.0	None	Transport certificate file used for protecting content
HLSDrmLicenseServerCertFile	Adobe Access 4.0	None	File containing license server certificate used for protecting content
HLSDrmPackagerCredentialFile	Adobe Access 4.0	None	File containing Packager credential used for protecting content
HLSDrmPackagerCredentialPassword	Adobe Access 4.0	None	Packager credential password for the configured packager credential file
HLSDrmPolicyFile	Adobe Access 4.0	None	Path and Name of the Policy File to be used for protecting content
HLSDrmKeyServerURL	Adobe Access 4.0	None	Key server URL for embedding in the served playlist.
PHLSOutputProtection	(Optional) PHLS	None	The required hardware Output Protection of media on the client. Possible values are None, BestEffort, and Required.
PHLSPlaybackExpiration	(Optional) PHLS	24 Hours	The duration of the time for which the content is available for playback. Possible values are 24Hours and Unlimited.

PHLS

Edit the file httpd.conf and update the `<Location /hls-vod>` directive as follows:

```
<Location /hls-vod>
    HLSHttpStreamingEnabled true
    HLSMediaFileDuration 8000
    HttpStreamingContentPath "../webroot/vod"
    HLSFmsDirPath ".."
    HLSJITConfAllowed true
    HLSEncryptionScope  server
    HLSProtectionScheme  PHLS
    PHLSContentID httpd_conf
    PHLSCommonKeyFile  "../creds/liveeventkey.bin"
    PHLSOutputProtection  None
    PHLSPlaybackExpiration  Unlimited
</Location>
```

Request the following URL from an iOS device:

http://<ServerName>/hls-vod/sample2_1000kbps.f4v.m3u8

For more information on the elements, see "Vanilla" on page 168.

Adobe Access

Edit the file httpd.conf and update the `<Location /hls-vod>` directive as follows:

```
<Location /hls-vod>
    HLSHttpStreamingEnabled true
    HLSMediaFileDuration 8000
    HttpStreamingContentPath "../webroot/vod"
    HLSFmsDirPath ".."
    HLSJITConfAllowed true
    HLSEncryptionScope server
    HLSProtectionScheme AdobeAccessV4
    HLSDrmContentID httpd_conf
    HLSDrmCommonKeyFile  "<path to common key file>"
    HLSDrmLicenseServerURL  "<url of license server>"
    HLSDrmTransportCertFile  "path to transport certificate file"
    HLSDrmLicenseServerCertFile  "<path to license server certificate file>"
    HLSDrmPackagerCredentialFile  "<path to packager credential file>"
    HLSDrmPackagerCredentialPassword  ??????
    HLSDrmPolicyFile  "<path to policy file"
    HLSDrmKeyServerURL  "<key server url>"
</Location>
```

Request the following URL from an iOS device:

http://<ServerName>/hls-vod/sample2_1000kbps.f4v.m3u8

Note: *For local key delivery, it is recommended that* `HLSDrmKeyServerURL` *be set to the dummy URL http://faxs.adobe.com.*

For more information on the elements, see "Vanilla" on page 168.

Stream level

To configure individual sets of media, in the httpd.conf file, set `HLSEncryptionScope` to `content`. This setting tells the server that configuration settings in the jit.conf file override settings in the httpd.conf file.

Configure the following elements in a jit.conf file in the same directory as the on-demand media:

Element	Default value	Description
//manifest/hds:encryption	None	The parent element for configuration. This element has with an enabled attribute. To enable content for protection with PHLS, set the enabled attribute to "true". The value is "false" by default.
//manifest/hds:encryption/hds:keyfile	None	The path of the default cipher key used to encrypt the content.
//manifest/hds:encryption/hds:keyuri	None	The URI that the client uses to fetch the encryption key. See "Serve encryption keys to the client" on page 176.

Vanilla

1 To configure live PHLS at the stream level, open the rootinstall/Apache2.2/conf/httpd.conf file and locate the hlshttp_module:

```
<IfModule hlshttp_module>
    ...
<Location /hls-vod>
    ...
```

2 Uncomment HLSEncryptionScope and set it to content:

```
# Uncomment the following directives to enable encryption
# for this location:
    HLSEncryptionScope content
#   HLSEncryptCipherKeyFile
#   HLSEncryptKeyURI
```

3 Create a jit.conf configuration file and copy it to the same directory as the on-demand media files.

```
<hds:hls>
    <hds:encryption enabled="true">
    <hds:keyfile>../creds/content.key</hds:keyfile>
    <hds:keyuri>https://<server-name>/hls-key/content.key</hds:keyuri>
    </hds:encryption>
</hds:hls>
```

4 Follow the steps in "Serve encryption keys to the client" on page 176to configure the server to serve keys with or without SSL. These steps configure the /hls-key path in the /hds:keyuri element.

5 Copy the vodkey.bin file from *rootinstall*/creds to *rootinstall*/webroot/keys.

6 Request the following URL from an iOS device:

http://<servername>/hls-vod/sample2_1000kbps.f4v.m3u8

7 To verify that the stream is encrypted, run the Apple Media Stream Validator Tool on the stream. See Technical Note TN2224.

PHLS

See the following sample configuration:

```
<?xml version="1.0" encoding="utf-8"?>
    <manifest xmlns="http://ns.adobe.com/f4m/1.0"
        xmlns:hds="http://ns.adobe.com/hds-package/1.0">
            <hds:hls>
                <hds:encryption enabled="true"  protection-scheme="PHLS" >
                    <hds:PHLS>
                        <hds:content-id>jit_conf</hds:content-id>
                        <hds:common-key-file>
                            root_install/creds/vodkey.bin
                        </hds:common-key-file>
                        <hds:output-protection>None</hds:output-protection>
                        <hds:playback-expiration>Unlimited</hds:playback-expiration>
                    </hds:PHLS>
                </hds:encryption>
            </hds:hls>
    </manifest>
```

Request the following URL from an iOS device:

http://<ServerName>/hl-vod/sample2_1000kbps.f4v.m3u8

For configuring the server with PHLS, see the steps mentioned in the Vanilla section. For details on the configuration elements, see the table above.

Adobe Access

See the following sample configuration:

```
<?xml version="1.0" encoding="utf-8"?>
    <manifest xmlns="http://ns.adobe.com/f4m/1.0"
        xmlns:hds="http://ns.adobe.com/hds-package/1.0">
    <hds:hls>
        <hds:encryption enabled="true"  protection-scheme="AdobeAccessV4">
            <hds:AdobeAccessV4>
                <hds:content-id>jit_conf</hds:content-id>
                <hds:common-key-file>
                    root_install/creds/vodkey.bin
                </hds:common-key-file>
                <hds:license-server-url>
                    http://mylicenseserver.myhost.com
                </hds:license-server-url>
                <hds:transport-cert-file>
                    production_transport.der
                </hds:transport-cert-file>
                <hds:license-server-cert-file>
                    production_license_server.der
                </hds:license-server-cert-file>
                <hds:packager-credential-file>
                    production_packager.pfx
                </hds:packager-credential-file>
                <hds:packager-credential-password>
                    ??????
                </hds:packager-credential-password>
                <hds:policy-file>policy.pol</hds:policy-file>
                <hds:key-server-url>http://faxs.adobe.com</hds:key-server-url>
            </hds:AdobeAccessV4>
        </hds:encryption>
    </hds:hls>
</manifest>
```

For local key delivery, it is recommended that HLSDrmKeyServerURL be set to the dummy URL http://faxs.adobe.com.

Request the following URL from an iOS device:

http://<ServerName>/hl-vod/sample2_1000kbps.f4v.m3u8

For configuring the server with PHLS, see the steps mentioned in the Vanilla section. For details on the configuration elements, see the table above.

VOD streams

To generate unique content encryption keys (CEKs) for Adobe Access, the URL path (relative to the configured content path) including the stream name is used as Content ID. For example, Content ID for path http://example.com/hls-vod/mymedia/sample.f4v.m3u8 would be mymedia/sample.f4v.

To change the default Content ID, specify the new Content ID in the configuration files event.xml, application.xml or jit.conf.

Multiple renditions of the same content require the same CEK for each rendition. To enable the same CEK across multiple renditions of the same content, configure the content-id in:

- application.xml
- event.xml (for Live Events) or jit.conf (for VOD Events).

You can protect the renditions using the Adobe Access configurations.

License chaining

If the configuration for embedding the leaf license is turned off, Adobe Media Server will still support such a policy except that the leaf license will not be embedded in the DRM metadata.

***Note:** The support will be limited to a single license server credential and credential-password pair.*

The following table provides the configuration details:

Parameter	Description	Required with	Default value
HLSDrmEmbedLeafLicense (Server level) HLS/Encryption/AdobeAccessV4/EmbedLeafLicense(Application and Eventlevel) hds:hls/hds:encryption/hds:AdobeAccessV4/hds:embed-leaf-license (VOD Use case - Stream level)	(Optional) Enables embedding of leaf licenses for policies generated using chained licenses. Possible values are "true" or "false".	AdobeAccessV4	false
HLSDrmLicenseServerCredentialFile (Server level) HLS/Encryption/AdobeAccessV4/LicenseServerCredentialFile (Application and Eventlevel) hds:hls/hds:encryption/hds:AdobeAccessV4/hds:license-server-credential-file (VOD Use case -Stream level)	Required if HLSDrmEmbedLeafLicense is set to true. The license server credential used when protecting content at this location.	AdobeAccessV4	NA
HLSDrmLicenseServerCredentialPassword (Server level) HLS/Encryption/AdobeAccessV4/LicenseServerCredentialPassword (Application and Eventlevel) hds:hls/hds:encryption/hds:AdobeAccessV4/hds:li cense-server-credential-password (VOD Use case -Stream level)	Required if HLSDrmEmbedLeafLicense is set to true. The license server credential password for the configured license server credential file.	AdobeAccessV4	NA

Key rotation

To enable the feature, you must add the following configuration directives in the httpd.conf file:

Directive	Required with	Default Value	Description
HLSDrmEnableKeyRotation / PHLSEnableKeyRotation (Server level) EnableKeyRotation (Application and Eventlevel) hds:hls/hds:encryption/hds:FlashAccessV4/hds:enable-key-rotation (VOD Use case -Stream leve)	Optional with FlashAccessV4 and PHLS	true	Enabled by default. To enable key rotation set the attribute to "false".

Directive	Require d with	Default Value	Description
HLSDrmKeyRotationInterval / PHLSKeyRotationInterval (Server level) KeyRotationInterval (Application and Eventlevel) hds:hls/hds:encryption/hds:FlashAccessV4/hds:key-rotation-interval (VOD Use case -Stream leve)	Optional with FlashAccessV4 and PHLS	15	The key is changed after the specified number of seconds.

For HDS streams, the key rotation does not have any impact on the performance of the client or on scaling impact of the license server because rotating the key is handled in-band.

In HLS key rotation results in a key request from the key server when using remote key delivery. For local, the rotated key is in the updated M3U8 file.

Out-of-Band DRM metadata

To enable this feature, you must add the following configuration directives in the httpd.conf file:

Directive	Require d	Default Value	Description
HLSEmbedMetadata	No	For VOD, true For Live, false	Enables embedding of metadata in the playlist. The false value is applicable only when HLSMetaPackagingEnabled is set to true.
HLSMetaMaxAge	No	3600 seconds	The maximum age in the Cache-Control header for m3u8 responses. A value of -1 specifies that no Cache-Control header is set. If no value is specified, default value,3600 seconds, is assumed.
HLSMetaPackagingEnabled	No	true	(Optional) Enables just in time packaging of metadata for this location. The possible values are "true" or "false".

Player binding

A whitelist file (with extension .airwhitelist) is a text file that contains multiple entries where each entry corresponds to an application identifying four fields (publisher-id, app-id, min-ver, max-ver). The publisher-id is mandatory and rest of the fields are optional. The file can be generated by passing the certificate(s) used to sign the application(s) to the whitelist tool . Currently the whitelist tool only supports extracting publisher-id but the rest of the fields (if required) can be updated manually. Player binding can be enabled by configuration and a folder needs to be specified from where Adobe Media Server can locate the whitelist files (multiple whitelist files and multiple entries in a whitelist files are supported). Adobe Media Server will add the list of identifiers picked up from the whitelist files to the license it embeds in the metadata.

To enable the feature, you must add the following configuration directives in the httpd.conf file:

Directive	Required with	Default Value	Description
PHLSPlayerBindingEnabled (Server level) HLS/Encryption/PHLS/PlayerBindingEnabled (Application and Eventlevel) hds:hls/hds:encryption/hds:phls/hds:player-binding (VOD Use case - Stream leve)	PHLS	false	Enables player binding using white-list. Possible values are "true" or "false".
PHLSWhitelistFolder (Server level) HLS/Encryption/PHLS/WhitelistFolder (Application and Eventlevel) hds:hls/hds:encryption/hds:phls/hds:whitelist-folder (VOD Use case - Stream leve)	PHLS	NA	(Required if `HLSDrmPlayerBindingEnabled` is true) The directory location containing the white-list files. This will work only when `HLSDrmPlayerBindingEnabled` is set to true.

Serve encryption keys to the client

The following PHLS configurations specify the path the client uses to fetch the encryption key:

- `HLSEncryptKeyURI`
- `//manifest/hds:hls/hds:encryption/hds:keyuri`
- `//Application/HDS/HLS/Encryption/KeyURI`
- `//Event/HLS/Encryption/KeyURI`

For both on-demand and live vanilla encryption, serve encryption keys to the client through the Apache HLS module. The module unscrambles the key before serving the request.

Note: *Note that the key files used for configuring encryption always needs to scrambled.*

You can enable client authentication over SSL to ensure that key files are served securely. A reference configuration file and the Apple CA bundle are installed to the following locations:

***rootinstall*/Apache2.2/conf/httd-hls-secure.conf**

***rootinstall*/creds/certs/ca**

The httpd-hls-secure.conf file demonstrates how to configure a virtual host at the default SSL port with client authentication enabled for the location /hls-key with cipher key hosting enabled. However, this is only a reference configuration. To guarantee authentication for a production system, customize the configuration for your deployment.

Note: *The SSL certificate presented by the iOS client must be current. If the client presents an expired certificate, client authentication fails and an error message displays to the user (on the client). iOS clients with older iOS installations may encounter this problem.*

Serve key files with SSL client authentication

1 Uncomment the following lines in the Apache httpd.conf file:

```
"#LoadModule ssl_module"
"#Include conf/httpd-hls-secure.conf"
```

2 Customize the SSL properties in the rootinstall/Apache2.2/conf/httpd-hls-secure.conf file based on the deployment. This customization includes getting an SSL certificate from a recognized CA.

Important: *The SSL certificate generated for the server must have a CN that is a FQDN (Fully Qualified Domain Name), even in a test environment. If not, the iOS client may not present its client certificate and client authentication fails. If client authentication fails, the key file is not served and the iOS client crashes. This is a known Apple bug.*

3 Restart Apache.

Serve key files without SSL

1 Add the following to the Apache httpd.conf file under the line `<IfModule hlshttp_module>`:

```
<Location /hls-key>
    HLSEncryptHostCipherKey true
    HLSFmsDirPath ".."
    HLSEncryptKeyRepository "../creds"
</Location>
```

The `Location` path can be any value. Point the `HLSEncryptKeyRepository` directive to the location of the keys. The keys are in the *rootinstall*/creds folder by default.

2 Restart Apache.

Use the following parameters in the Apache httpd.conf file to configure key hosting:

Parameter	Description	Default value
`HLSEncryptHostCipherKey`	Enable (`true`) or disable (`false`) cipher key hosting from this location.	false
`HLSEncryptKeyRepository`	The path of the folder that contains the key file.	None

Dynamic Content Encryption Key

AMS supports different content encryption keys for content at the different levels (server, application, stream, and event). The keys are generated according to the location of the content and the location of the Common Key.

Delivering Content Encryption Keys

The Content Encryption Key delivery mode is specified in the policy file. For the Adobe Access 4.0 protection scheme, set the policy using HLSDrmPolicyFile parameter. To select the policy file for the PHDS protection scheme, HLSDrmOutputProtection and HLSDrmPlaybackExpiration are used.

The key server URL is based on the key delivery mode specified in the policy file. For remote key serving, use the KeyServerURL parameter to specify the URL of key server. The URL format for remote key serving is https://<customers-keyserver-uri>. For example, https://faxs.adobe.com. For local key serving, the value of KeyServerURL should always be faxs://faxs.adobe.com.

Note: *PHLS supports only local key delivery and AMS cannot deliver CEKs as long as DRM is enabled.*

Adaptive bitrate streaming

In order to support adaptive bitrate, HTTP Live Streaming requires a variant playlist file that referes to individual playlist files having different renditions of the same content. The Adobe Access for iOS SDK requires that each stream referred to in a variant playlist must be encrypted using the same policy and the same content encryption key. Hence each encrypted stream will have the same DRM metadata referred in `#EXT-X-FAXS-CM` tag (embedded or served out of band).

The Adobe Access Server protected variant playlist also needs to include the `#EXT-X-FAXS-CM` tag. The value of `#EXT-X-FAXS-CM` tag in variant playlist is the relative URI referring to the DRM metadata of one of the individual streams.At the client, the `#EXT-X-FAXS-CM` tag in variant playlist will be used to create the DRM session. The same DRM session will be used for all encrypted M3U8 files inside the variant playlist.

Here's an example of Adobe Access protected variant playlist:

```
#EXTM3U
#EXT-X-FAXS-CM:URI="hls-vod-
faxsv4/sample_mbr_mp4_main_3_1/8_mp4_AAC_212Kbps_720_480_main_3_1.mp4.drmmeta"
#EXT-X-STREAM-INF:PROGRAM-ID=41,BANDWIDTH=212000, CODECS="avc1.77.31, mp4a.40.5" hls-vod-
faxsv4/sample_mbr_mp4_main_3_1/8_mp4_AAC_212Kbps_720_480_main_3_1.mp4.m3u8
#EXT-X-STREAM-INF:PROGRAM-ID=41,BANDWIDTH=307000, CODECS="avc1.77.31, mp4a.40.5" hls-vod-
faxsv4/sample_mbr_mp4_main_3_1/8_mp4_AAC_307Kbps_720_480_main_3_1.mp4.m3u8
#EXT-X-STREAM-INF:PROGRAM-ID=41,BANDWIDTH=512000, CODECS="avc1.77.31, mp4a.40.5"
http://my.server.com/hls-vod-
faxsv4/sample_mbr_mp4_main_3_1/8_mp4_AAC_512Kbps_720_480_main_3_1.mp4.m3u8
```

Note: *This variant playlist needs to be served in Adobe Access M3U8 format. For instance, you need to append ?faxs=1 to the URL like http://my.server.com/variantPlaylist.m3u8?faxs=1.*

Chapter 3: Getting started developing applications

Application architecture

The client application is written in ActionScript™ and compiles to a SWF file. The server application is written in Server-Side ActionScript (which is like ActionScript 1.0, but runs on the server, rather than on the client). A media application usually has recorded or live audio and video that it streams from server to client, client to server, or server to server.

A typical Adobe Media Server application has these parts:

Client The client displays a user interface, such as controls to start, stop, or pause a video. The client can run in Flash Player, Adobe AIR, or Flash Lite 3. You can develop clients with Adobe Flash Professional, Adobe Flash Builder, or the Flex SDK. You can also use prebuilt clients. Adobe recommends using a player that is based on OSMF (Open Source Media Framework), such as Strobe Media Playback.

Client-side ActionScript The client contains ActionScript code that handles user interaction and connects to the server. Adobe Media Server 3 and later support ActionScript 3.0, ActionScript 2.0, and ActionScript 1.0.

Video or audio files Many media applications stream recorded audio or video from the server to clients or between clients. Adobe Media Server supports playback of a variety of stream formats, including Flash Video (FLV), MPEG-3 (MP3), MPEG-4 (MP4 and F4V), and RAW.

Camera or microphone You can use Flash Media Live Encoder to stream live video or audio to the server. You can also create your own client that captures live audio and video. In both cases, you need a camera and a microphone to capture the video and audio.

Server-Side ActionScript Most applications use Server-Side ActionScript code written in a file with the suffix *.asc*, called an *ActionScript Communication File*. The file is named either main.asc, or *myApplication*.asc. You can use server-side code to control access to applications, define what happens when users connect and disconnect, create playlists, connect to external data sources, and so on.

Adobe Media Server installs with four server-side applications (also called *streaming services*): live, vod, livepkgr, and multicast. Each service provides a different type of streaming.

Set up a development environment

You can use any edition of the server, including the free developer edition, to develop and test applications. To write client-side code, use Flash Professional, Flash Builder, or the Flex SDK. To write server-side code, you can use any text editor or IDE, including Flash Professional and Flash Builder.

You can write client-side code and server-side code on the same computer running Adobe Media Server, or you can write it on a remote computer. To test the application, copy the server-side code to Adobe Media Server.

If you have an account with a Adobe Media Server solution partner, the partner tells you how to configure your development environment to use their resources.

Set up a development environment:

1 Run the Adobe Media Server installer to install the server.

2 Verify that the server is installed successfully.

3 Do one of the following:

- Install Flash Professional from www.adobe.com/go/flash.
- Install Flash Builder from www.adobe.com/go/flashbuilder.
- Install the Flex SDK from opensource.adobe.com.

4 To capture and encode live video, do the following:

a Connect a camera and a microphone to the computer.

b Download and install Adobe Media Live Encoder from www.adobe.com/go/fmle.

***Note:** Adobe Media Live Encoder captures audio and video, encodes it, and sends it to Adobe Media Server. You can also build custom applications that capture and encode audio and video.*

Example: Hello World application

Overview

***Note:** The following sections do not apply to Adobe Media Streaming Server because you cannot write server-side code for that server edition.*

This sample shows simple communication from the client to the server and back again. When a user clicks a button, the client connects to the server. The client calls a server-side function that returns a string. When the server replies, the client displays the string sent from the server.

The sample files are in the *rootinstall*\documentation\samples\HelloWorld folder.

Create the user interface

1 Start Flash and select Create New > Flash File (ActionScript 3.0).

2 In the Document Class field, enter **HelloWorld**. If you see an ActionScript Class Warning message about a missing definition—click OK. You will add the class file in the next section.

3 Choose Windows > Components. Click User Interface and double-click Button to add it to the Stage. On the Properties tab, enter the instance name **connectBtn**.

4 Add a Label component above the button, and give it the instance name **textLbl**.

5 Save the file as HelloWorld.fla.

You can save the client files to any location.

Write the client-side script

This script provides two button actions, either connecting to or disconnecting from the server. When connecting, the script calls the server with a string ("World"), which triggers a response that displays the returned string ("Hello, World!").

1 Choose File > New > ActionScript File. Check that the Target box has HelloWorld.fla.

2 Declare the package and import the required Flash classes:

```
package {
    import flash.display.MovieClip;
    import flash.net.Responder;
    import flash.net.NetConnection;
    import flash.events.NetStatusEvent;
    import flash.events.MouseEvent;
    public class HelloWorld extends MovieClip {
    }
}
```

3 Inside the HelloWorld class declaration, declare variables for the connection and the server responder:

```
private var nc:NetConnection;
private var myResponder:Responder = new Responder(onReply);
```

4 Define the class constructor. Set the label and button display values, and add an event listener to the button:

```
public function HelloWorld() {
    textLbl.text = "";
    connectBtn.label = "Connect";
    connectBtn.addEventListener(MouseEvent.CLICK, connectHandler);
}
```

5 Define the event listener actions, which depend on the button's current label:

```
public function connectHandler(event:MouseEvent):void {
        if (connectBtn.label == "Connect") {
            trace("Connecting...");
            nc = new NetConnection();
            // Connect to the server.
            nc.connect("rtmp://localhost/HelloWorld");
            // Call the server's client function serverHelloMsg, in HelloWorld.asc.
            nc.call("serverHelloMsg", myResponder, "World");
            connectBtn.label = "Disconnect";
        } else {
            trace("Disconnecting...");
            // Close the connection.
            nc.close();
            connectBtn.label = "Connect";
            textLbl.text = "";
        }
}
```

6 Define the responder function, which sets the label's display value:

```
private function onReply(result:Object):void {
    trace("onReply received value: " + result);
    textLbl.text = String(result);
}
```

7 Save the file as HelloWorld.as to the same folder as the HelloWorld.fla file.

Write the server-side script

1 Choose File > New > ActionScript Communications File.

2 Define the server-side function and the connection logic:

```
application.onConnect = function( client ) {
    client.serverHelloMsg = function( helloStr ) {
        return "Hello, " + helloStr + "!";
    }
    application.acceptConnection( client );
}
```

3 Save the file as HelloWorld.asc in the *rootinstall*/applications/HelloWorld folder. (Create the "HelloWorld" folder when you save the file.)

Compile and run the application

1 Verify that the server is running.

2 Select the HelloWorld.fla file tab.

3 Choose Control > Test Movie.

4 Click the Connect button.

"Hello, World!" is displayed, and the button label changes to Disconnect.

5 Click the Disconnect button.

The output of the `trace()` statements is displayed in the Flash Output window.

Overview of creating an application

Client-side code

A client has code written in ActionScript that connects to the server, handles events, and does other work. Working in Flash Professional, you can use ActionScript 3.0 or 2.0. Flash Builder and the Flex SDK use ActionScript 3.0.

For information about learning ActionScript and working with video, see the following resources:

- To simply stream media, you can use a pre-built media player. For more information, see "Build custom media players" on page 91.
- To build a social media application, see "Server-side code" on page 183.
- Flash Help Resource Center at www.adobe.com/devnet/flash.
- Flex Help Resource Center at www.adobe.com/devnet/flex.

More Help topics

"Copy client-side files to a web server" on page 187

Server-side code

In general, applications require server-side code written in Server-Side ActionScript if they need to do any of the following:

Authenticate clients By user name and password, or by credentials stored in an application server or database.

Implement connection logic By taking some action when a client connects or disconnects.

Update clients By calling remote methods on clients or updating shared objects that affect all connected clients.

Handle streams By allowing you to play, record, and manage streams sent to and from the server.

Connect to other servers By calling a web service or creating a network socket to an application server or database.

Place the server-side code in a file named main.asc or *yourApplicationName*.asc, where *yourApplicationName* is a folder in the *rootinstall*/applications folder. For example, to create an application called *skatingClips*, create the folder *rootinstall*/applications/skatingClips. The server-side code would be in a file called *main.asc* or *skatingClips.asc* in the skatingClips folder.

The server-side code goes at the top level of the application directory, or in its *scripts* subdirectory. For example, you can use either of these locations:

rootinstall/applications/*appName*

rootinstall/applications/*appName*/scripts

By default, the applications folder is in the root installation folder (C:\Program Files\Adobe\Adobe Media Server 5\applications, on Windows). To configure the location of the applications folder, edit the fms.ini or the Vhost.xml configuration file. In the fms.ini file, edit the following parameter: `VHOST.APPSDIR = C:\Program Files\Adobe\Adobe Media Server 5\applications`. In the Vhost.xml file, edit the `AppsDir` element.

More Help topics

"Copy server-side script files to the server" on page 186

Client and application objects

Server-side scripts have access to two special objects, the Client object and the application object. When a client connects to an application on Adobe Media Server, the server creates an instance of the server-side Client class to represent the client. An application can have thousands of clients connected. In your server-side code, you can use the Client object to send and receive messages to individual clients.

Each application also has a single application object, which is an instance of the server-side Application class. The application object represents the application instance. You can use it to accept clients, disconnect them, shut down the application, and so on.

Writing double-byte applications

If you use Server-Side ActionScript to develop an application that uses double-byte text (such as an Asian language character set), place your server-side code in a main.asc file that is UTF-8 encoded. Use a JavaScript editor, such as the Script window in Flash or Adobe® Dreamweaver®, that encodes files to the UTF-8 standard. Use built-in JavaScript methods, such as `Date.toLocaleString()`, to convert the string to the locale encoding for that system.

Some simple text editors might not encode files to the UTF-8 standard. However, some editors provide a Save As option to encode files in the UTF-8 standard.

Set UTF-8 encoding in Dreamweaver

1 Check the document encoding setting by selecting Modify > Page Properties, then Document Encoding. Choose Unicode (UTF-8).

2 Change the inline input setting by selecting Edit > Preferences (Windows) or Dreamweaver > Preferences (Mac OS), and then click General. Select Enable Double-Byte Online Input to enable double-byte text.

Use double-byte characters as method names

❖ Assign method names using the object array operator, not the dot operator:

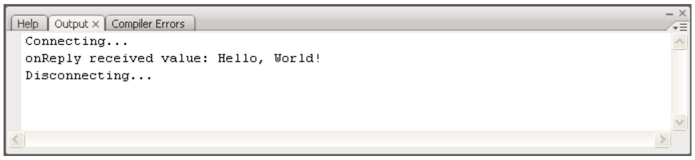
```
// This is the CORRECT way to create double-byte method names
obj["Any_hi_byte_name"] = function(){}

// This is the INCORRECT way to create double-byte method names.
obj.Any_hi_byte_name = function() {}
```

Test an application

Test and debug a client-side script

To help test a client-side script, use `trace()` statements to monitor each processing point. The output is shown in the Flash Output window (this example is from the "Example: Hello World application" on page 180):



```
Connecting...
onReply received value: Hello, World!
Disconnecting...
```

To debug a client-side script, use the Debug menu in Flash to set breakpoints, step into functions, and so forth. You can inspect the state of the script with Windows > Debug Panels.

Test and debug a server-side script

To test a server-side script, use `trace()` statements to monitor each processing point. View the output of `trace()` statements in the Live Log in the Administration Console.

To open the Administration Console, choose Start > All Programs > Adobe > Adobe Media Server 3.5 > Adobe Media Administration Console.

When a client connects to an application on the server, the application is loaded and can be seen in the Administration Console. To load an application directly from the Administration Console, select from the New Instance list of available application names. You can also stop an application or reload it—in either case, all clients are disconnected.

***Note:** When you edit and save an .asc file, the changes do not take effect until the application is restarted. Use the Administration Console to restart the application, then connect to the application again.*

For each application instance, you can observe its live log, clients, shared objects, if any, streams in use, and performance statistics.

View the output of a server-side script

The output of the `trace()` statements in a main.asc file is sent to the following log file:

rootinstall/logs/_defaultVHost_/*yourApplicationName*/*yourInstanceName*/application.*xx*.log

Where *yourInstanceName* is _definst_ by default and *xx* is the instance number, 00 for the most recent log file, 01 for the previous instance, and so on. You can view a log file with any text editor.

While an application is running, you can view the Live Log in the Administration Console. In the Administration Console opens, click View Applications, then Live Log.

Debug with the Administration Console

To playback streams and inspect data about shared objects, an application must make a special debug connection to the Administration Console.

The availability and number of debugging sessions is set in the `AllowDebugDefault` and `MaxPendingDebugConnections` elements of the Application.xml configuration file. By default, debugging is not allowed. To override the debug setting in the Application.xml file, add the following code to an application's server-side code:

```
application.allowDebug = true;
```

Note: *Set `allowDebug` to `false` before deploying the application.*

To start a debugging session:

1 Open the Administration Console.

2 Choose View Applications.

3 Select the application to debug from the list or choose New Instance and create a new instance of an application.

4 Press the Streams button to see the list of playing streams, if any.

5 Click on one of the streams.

6 Press the Play Stream button.

7 A pop-up window will open and the stream will play.

8 Press the Shared Objects button to see the application's shared objects, if any.

9 Select a shared object.

10 Press the Close Debug button to end the debug session.

Deploy an application

Register an application with the server

To connect to an application, the server must know that the application exists. This process is called registering the application with the server. To register an application with the server, create a folder for the application in the applications folder. For example, create the following folder to register an application called "myApplication":

rootinstall/applications/myApplication

The client-side code that connects to the application looks like the following:

```
myNetConnection.connect("rtmp://fms.examples.com/myApplication");
```

To create instances of an application, create subfolders. For example, the following folder creates an instance of myApplication called "room1":

rootinstall/applications/myApplication/room1.

The client-side code that connects to the application instance looks like the following:

```
myNetConnection.connect("rtmp://fms.examples.com/myApplication/room1");
```

Every application must have a folder in the applications folder. Usually, the application folder contains the server-side script and any media assets, but the folder can also be empty. It can also contain an application-specific Application.xml file.

By default, the applications folder is in the root installation folder (C:\Program Files\Adobe\Adobe Media Server 5\applications, on Windows). To configure the location of the applications folder, edit the fms.ini or the Vhost.xml configuration file. In the fms.ini file, edit the following parameter: `VHOST.APPSDIR = C:\Program Files\Adobe\Adobe Media Server 5\applications`. In the Vhost.xml file, edit the `AppsDir` element.

Copy server-side script files to the server

Copy server-side script files for an application to the folder you registered on the server. For example, for an application called "videoPlayer", copy the main.asc file to *rootinstall*/applications/videoPlayer. You can also place server-side scripts in "scripts" subfolder. For example, you can use either of these locations:

rootinstall/applications/*appName*

rootinstall/applications/*appName*/scripts

Note: *To replace a running application, copy the new files, then use the Administration Console to restart the application.*

Packaging server-side files

Adobe Media Server includes a command-line archive compiler utility, far.exe, which lets you package server-side scripts into a FAR file, which is an archive file like a ZIP file, to simplify deployment. You can also use the archive compiler utility to compile server-side script files to bytecode (with the file extension .ase) to speed the time required to load an application instance.

A large application can contain multiple server-side script files stored in different locations. Some files are located in the application directory and others are scattered in the script library paths that are defined in the server configuration file. To simplify deployment of your media application, you can package your server-side JS, ASC, and ASE files in a self-contained Adobe Media Server archive file (a FAR file).

The FAR file is a package that includes the main script file (which is either main.js, main.asc, main.ase, *applicationName*.js, *applicationName*.asc, or *applicationName*.ase) and any other script files that are referred to in the main script.

The syntax for running the archive compiler utility to create a script package is as follows:

```
c:\> far -package -archive <archive> -files <file1> [<file2> ... <fileN>]
```

The following table describes the command-line options available for `far -package`.

Option	Description
-archive *archive*	Specifies the name of the archive file, which has a .far extension.
-files *file1* [*file2* ... *fileN*]	Specifies the list of files to be included in the archive file. At least one file is required.

Note: *If the main script refers to scripts in a subdirectory, the hierarchy must be maintained in the archive file. To maintain this hierarchy, Adobe recommends that you run the FAR utility in the same directory where the main script is located.*

Copy media files to the server

Copy video and audio files to the streams/_definst_ folder in the application folder:

rootinstall/applications/*someapplication*/streams/_definst_/

If an application connects to an instance of the application, for example, `nc.connect("rtmp://`*`fms.example.com/someapplication/someinstance`*`")`, place the streams in the following folder:

rootinstall/applications/*someapplication*/streams/*someinstance*/

There are several ways to configure the server to look for media files stored in other locations. For example, the vod streaming service that installs with the server is configured to look for media files in the *rootinstall*/applications/vod/media folder.

More Help topics

"Stream on-demand media (RTMP)" on page 24

Copy client-side files to a web server

You can deploy SWF files and HTML files on any web server. The SWF file contains the `NetConnection.connect()` call that connects to the application on Adobe Media Server. The HTML file is a container for the SWF file. For more information, see the Flash documentation at www.adobe.com/support/flash.

For more information about deploying AIR applications, see the Adobe AIR documentation at www.adobe.com/support/air.

By default, Adobe Media Server 5.0 and later install with Apache HTTP Server. If you installed and enabled the web server, you can deploy SWF files and HTML files from the same computer on which Adobe Media Server is installed. You can also deploy JPG, GIF, and many other file types. For a complete list of the file types Apache serves by default, see the *rootinstall*/Apache2.2/conf/httpd.conf file.

By default, Apache is configured with the following aliases:

Alias	Path
/	*rootinstall*/webroot
/cgi-bin/	*rootinstall*/Apache2.2/cgi-bin

To serve files over HTTP, place the files in the *rootinstall*/webroot folder or an appropriate subfolder. To serve CGI programs, place the files in the *rootinstall*/Apache2.2/cgi-bin folder.

You can also add subfolders to these folders. Subfolders cannot use the following names from the fmshttpd.conf file: icons, error, SWFs, vod, or cgi-bin. In addition, subfolder names cannot use the following reserved words: open, send, idle, fcs, fms.

By default, Apache is configured to serve all HTTP content from the *rootinstall*/webroot folder. If you want to provision the server to serve files from application-specific directories, edit the httpd.conf Apache configuration file.

Chapter 4: Developing streaming media applications

Adobe® Flash® Media Server can stream live video and recorded video to and from Flash Player and AIR. Recorded video is called *on-demand*, or VOD (video on demand).

Live applications capture, encode, and stream live media. Live video is typically used for events such as corporate meetings, education, sports events, and concerts.

Connecting to the server

About the NetConnection class

Before a client can play or publish audio and video, it must connect to the server. The connection request is accepted or rejected by an application instance on the server. The server sends information messages back to the client. Once the application accepts the connection request, a connection is available to both the client and the server.

The NetConnection class connects a client to an application instance on the server. To connect, create an instance of the NetConnection class and call the `connect()` method. Pass an application instance URI to the `connect()` method:

```
var nc:NetConnection = new NetConnection();
nc.connect("rtmp://localhost/HelloServer");
```

The server sends back a `NetStatusEvent` that tells the client whether it has connected successfully.

A NetConnection object is like a pipe that streams audio, video, and data from client to server, or from server to client. Once you create the NetConnection object, you can attach one or more streams to it. Streams handle the flow of audio, video, and data over a network connection.

A stream can carry more than one type of content (audio, video, and data). However, a stream flows in only one direction, from server to client or client to server.

More Help topics

"Managing connections" on page 193

About the application URI

The application instance URI can be absolute or relative and has the following syntax (items in brackets are optional):

```
protocol:[//host][:port]/appname/[instanceName]
```

The parts of the URI are described in the following table.

Part	Example	Description
protocol:	rtmp:	The protocol used to connect to Adobe Media Server, Real-Time Messaging Protocol and its variations. For possible values, see the `NetConnection.connect()` entry in the ActionScript 3.0 Reference.
//host	//www.example.com //localhost	The host name of a local or remote computer. To connect to a server on the same host computer as the client, use //localhost or omit the //host identifier.
:port	:8080	The port number to connect to on the server. For RTMP and its variations, the default port is 1935. For HTTP and its variations, the default port is 80. To connect over the default port, do not specify the port number. Specify the port number only to connect to a port not configured as the default.
/appname/	/sudoku/	The name of a subdirectory of the *rootinstall*/applications folder. You can specify another location for the "applications" directory in the fms.ini configuration file (at *rootinstall*/conf/fms.ini).
instanceName	room1	An instance of the application to which the client connects. For example, a chat room application can have many chat rooms: chatroom/room1, chatroom/room2, and so on. If you do not specify an instance name, the client connects to the default application instance, named _definst_.

If the client and the application are on the same server, as they often are while developing an application, the only parts of the URI that are required are the protocol and the application name. You can also use //localhost or 127.0.0.1 for the domain name:

```
rtmp:/vod/
rtmp://localhost/vod
rtmp://127.0.0.1/vod
```

To test these connections, open *rootinstall*/samples/videoPlayer/videoplayer.html in a browser. Open the Adobe Media Administration Console (*rootinstall*/tools/fms_adminConsole.htm) in a browser. In the sample video player, enter each URL in the Stream URL box, and click Play Stream. In the Adobe Media Administration Console, choose View Applications > Live Log to see the connection.

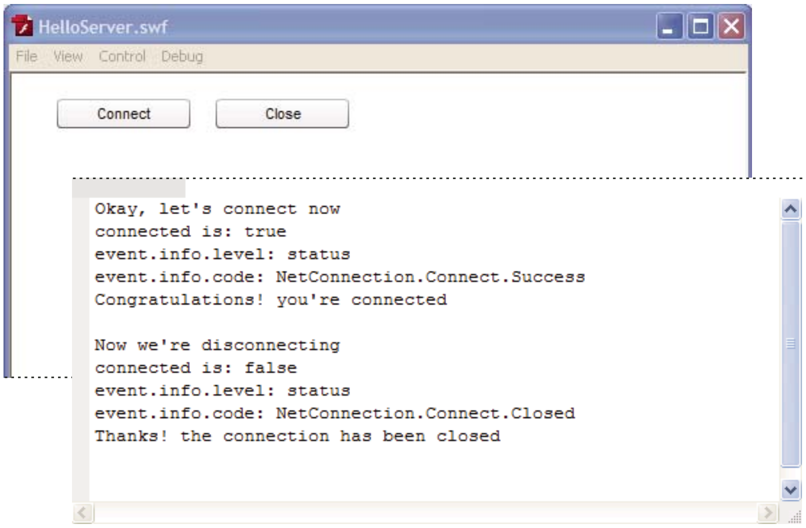
When the client and the application aren't on the same server, specify a host domain name or IP address, as in any of the following:

```
rtmp://www.fmsserver.com/vod
rtmp://www.fmsserver.com/vod/mediaplayer1
rtmp://www.fmsserver.com/vod/mediaplayer2
```

Example: Hello Server application

***Note:** Adobe Media Streaming Server does not support this example.*

You can find the HelloServer application in the *rootinstall*/documentation/samples/HelloServer folder. This simple Flash application displays two buttons that enable you to connect to the server and close the connection.



```
Okay, let's connect now
connected is: true
event.info.level: status
event.info.code: NetConnection.Connect.Success
Congratulations! you're connected

Now we're disconnecting
connected is: false
event.info.level: status
event.info.code: NetConnection.Connect.Closed
Thanks! the connection has been closed
```

The Output window displays messages about the connection status

Run the application

The easiest way to run the sample is to install it on the same computer as the server.

1 Register the application by creating a folder in the server's applications folder:

   rootinstall/applications/HelloServer

2 (Optional) To run the sample on a server installed on a different computer, open HelloServer.as and edit this line to add the URL to your server:

```
nc.connect("rtmp://localhost/HelloServer");
```

Design the Flash user interface

The sample is already built and included in the samples folder. However, these instructions show you how to recreate it, so that you can build it on your own and add to it.

1 In Flash Professional, choose File > New > Flash File (ActionScript 3.0), and click OK.

2 Choose Window > Components to open the Components panel.

3 Double-click the Button component to add it to the Stage.

4 In the Properties Inspector, click the Properties tab. Select MovieClip as the instance behavior, and enter the instance name **connectBtn**.

5 Click the Parameters tab, then Label. Enter **Connect** as the button label.

6 Drag a second button component to the Stage.

7 Give the second button the instance name **closeBtn** and the label **Close**.

8 Save the FLA file, naming it HelloServer.fla.

Write the client-side code

1 In Flash Professional, choose File > New > ActionScript File, and click OK.

2 Save the ActionScript file as HelloServer.as.

3 Return to the FLA file. Choose File > Publish Settings. Click the Flash tab, then Settings.

4 In the Document Class box, enter HelloServer. Click the green check mark to make sure the class file can be located. Click OK, then OK again.

5 In the ActionScript file, enter a package declaration. If you saved the file to the same directory as the FLA file, do not use a package name, for example:

```
package {
}
```

However, if you saved the file to a subdirectory below the FLA file, the package name must match the directory path to your ActionScript file, for example:

```
package samples {
}
```

6 Within the package, import the ActionScript classes you'll use in the script:

```
import flash.display.MovieClip;
import flash.net.NetConnection;
import flash.events.NetStatusEvent;
import flash.events.MouseEvent;
```

7 After the `import` statements, create a class declaration. Within the class, define a variable of type NetConnection:

```
public class HelloServer extends MovieClip {
    private var nc:NetConnection;
}
```

Because the class extends the MovieClip class, it inherits all the properties and methods of the MovieClip class.

8 Write the class constructor, registering an event listener on each button:

```
public function HelloServer() {
    // register listeners for mouse clicks on the two buttons
    connectBtn.addEventListener(MouseEvent.CLICK, connectHandler);
    closeBtn.addEventListener(MouseEvent.CLICK, closeHandler);
}
```

Use `addEventListener()` to call an event handler named `connectHandler()` when a `click` MouseEvent occurs on the Connect button. Likewise, call `closeHandler()` when a `click` MouseEvent occurs on the Close button.

9 Write the `connectHandler()` function to connect to the server when a user clicks the Connect button:

```
public function connectHandler(event:MouseEvent):void {
    trace("Okay, let's connect now");
    nc = new NetConnection();
    nc.addEventListener(NetStatusEvent.NET_STATUS, netStatusHandler);
    nc.connect("rtmp://localhost/HelloServer");
}
```

In `connectHandler()`, add an event listener to listen for a `netStatus` event returned by the NetConnection object. Then, connect to the application instance on the server by calling `NetConnection.connect()` with the correct URI. This URI connects to an application instance named *HelloServer*, where the server runs on the same computer as the client.

10 Write the `closeHandler()` function to define what happens when a user clicks the Close button:

```
public function closeHandler(event:MouseEvent):void {
    trace("Now we're disconnecting");
    nc.close();
}
```

It's a best practice to explicitly call `close()` to close the connection to the server.

11 Write the `netStatusHandler()` function to handle `netStatus` objects returned by the NetConnection object:

```
public function netStatusHandler(event:NetStatusEvent):void {
    trace("connected is: " + nc.connected);
    trace("event.info.level: " + event.info.level);
    trace("event.info.code: " + event.info.code);
    switch (event.info.code)
    {
        case "NetConnection.Connect.Success":
            trace("Congratulations! you're connected" + "\n");
            break;
        case "NetConnection.Connect.Rejected":
            trace ("Oops! the connection was rejected" + "\n");
            break;
        case "NetConnection.Connect.Closed":
            trace("Thanks! the connection has been closed" + "\n");
            break;
        }
}
```

A `netStatus` object contains an `info` object, which in turn contains a `level` and a `code` that describes the connection status.

Understand the connection messages

When you run the sample and click the Connect button, you see messages like this, as long as the connection is successful:

```
Okay, let's connect now
connected is: true
event.info.level: status
event.info.code: NetConnection.Connect.Success
Congratulations! you're connected
```

The line `connected is: true` shows the value of the `NetConnection.connected` property, meaning whether Flash Player is connected to the server over RTMP. The next two lines describe the `netStatus` event the NetConnection object sends to report its connection status:

```
event.info.level: status
event.info.code: NetConnection.Connect.Success
```

The `level` property can have two values, `status` or `error`. The `code` property describes the status of the connection. You can check for various `code` values in your `netStatusHandler` function and take action. Always check for a successful connection before you create streams or do other work in your application.

Likewise, when you click the Close button, you see the following:

```
Now we're disconnecting
connected is: false
event.info.level: status
event.info.code: NetConnection.Connect.Closed
Thanks! the connection has been closed
```

Managing connections

Connection status codes

After the connection between client and server is made, it can break for various reasons. The network might go down, the server might stop, or the connection might be closed from the server or the client. Any change in the connection status creates a `netStatus` event, which has both a `code` and a `level` property describing the change. This is one `code` and `level` combination:

Code	Level	Meaning
`NetConnection.Connect.Success`	`status`	A connection has been established successfully.

See `NetStatus.info` in the *ActionScript 3.0 Reference* for a complete list of all code and level values that can be returned in a `netStatus` event.

When the event is returned, you can access the connection code and level with `event.info.code` and `event.info.level`. You can also check the `NetConnection.connected` property (which has a value of `true` or `false`) to see if the connection still exists. If the connection can't be made or becomes unavailable, you need to take some action from the application client.

Managing connections in server-side code

An application can have server-side code in a main.asc or *applicationName*.asc file that manages clients trying to connect.

The server-side code has access to `Client` objects, which represent individual clients on the server side, and a single `application` object, which enables you to manage the application instance. In the server code, you use Server-Side ActionScript and the server-side information objects (see the *Server-Side ActionScript Language Reference*).

In the server-side code, the application can accept or reject connections from clients, shut down the application, and perform other tasks to manage the connection. When a client connects, the application receives an `application.onConnect` event. Likewise, when the client disconnects, the application receives an `application.onDisconnect` event.

To manage the connection from the server, start with `application.onConnect()` and `application.onDisconnect()` in Server-Side ActionScript.

More Help topics

"Server-side code" on page 183

Example: Managing connections

This example shows how to manage connections from both the application client and the server-side code.

Write the client code

In the client code, you need to check for specific connection codes and handle them. Create live streams or play recorded streams only when the client receives `NetConnection.Connect.Success`.

***Note:** See the SimpleConnectManage sample, SimpleConnectManage.as file.*

1 Create a NetConnection object and call the `connect()` method to connect to the server.

2 Write a `netStatus` event handler. In it, check for specific connection codes, and take an action for each:

```
public function netStatusHandler(event:NetStatusEvent):void
{
    trace("connected is: " + nc.connected );
    trace("event.info.level: " + event.info.level);
    trace("event.info.code: " + event.info.code);
    switch (event.info.code)
    {
        case "NetConnection.Connect.Success":
            trace("Congratulations! you're connected");
            // create live streams
            // play recorded streams
            break;
        case "NetConnection.Connect.Rejected":
            trace ("Oops! the connection was rejected");
            // try to connect again
            break;
        case "NetConnection.Connect.Failed":
            trace("The server may be down or unreachable");
            // display a message for the user
            break;
        case "NetConnection.Connect.Closed":
            trace("The connection was closed successfully - goodbye");
            // display a reconnect button
            break;
        }
    }
```

Run the code

***Note:** These instructions apply to any ActionScript 3.0 example without a Flash user interface. The ActionScript 3.0 examples are provided for your convenience.*

1 Check the client-side code to see which application it connects to:

```
nc.connect("rtmp://localhost/HelloServer");
```

2 Register the application on the server by creating an application instance directory for it in the applications directory, for example:

rootinstall/applications/HelloServer

3 (Optional) Or, to use an application you have registered with the server, change the URI used in the call to `connect()`:

```
nc.connect("rtmp://localhost/MyApplication");
```

4 In Flash Builder, create an ActionScript project named SimpleConnectManage (choose File > New > ActionScript Project, and follow the wizard).

5 Add the SimpleConnectManage sample files to the project.

6 Choose Run > Debug. In the Debug window, enter SimpleConnectManage for Project and SimpleConnectManage.as for Application file. Click Debug.

7 Close the empty application window that opens and return to Flash Builder. Check the messages in the Console window.

If the connection is successful, you should see output like this:

```
connected is: true
event.info.level: status
event.info.code: NetConnection.Connect.Success
Congratulations! you're connected
[SWF] C:\samples\SimpleConnectManage\bin\SimpleConnectManage-debug.swf - 2,377 bytes after
decompression
```

Get the server version number

Use the ActionScript 3.0 NetStatusEvent to get the server version number. The server version number is in the `event.info.data.version` property. The following code connects to the vod service and outputs the server version number:

```
import flash.net.NetConnection;
import flash.events.NetStatusEvent;
import flash.events.AsyncErrorEvent;
var nc:NetConnection = new NetConnection();
nc.client = this;
nc.addEventListener(NetStatusEvent.NET_STATUS, netStatusHandler);
nc.addEventListener(AsyncErrorEvent.ASYNC_ERROR, asyncErrorHandler);
function netStatusHandler(event:NetStatusEvent):void{
    trace("Adobe Media Server version: " + event.info.data.version);
    trace(event.info.code);
}
function asyncErrorHandler(event:AsyncErrorEvent):void{
    trace(event);
}
// The vod applicatin expects this function to check bandwidth.
function onBWDone(...data):Number{
    return 0;
}
nc.connect("rtmp://localhost/vod");
```

You can also add the following line of code inside the `netStatusHandler()` function in the HelloServer example application:

```
trace("Adobe Media Server version: " + event.info.data.version);
```

Streaming media files

Playing media files

To play a stream, use the NetStream class. Create a NetStream object and pass the contructor a NetConnection object:

```
ns:NetStream = new NetStream(nc);
```

Call the `NetStream.play()` method or the `NetStream.play2()` method and pass the URI of the media file:

```
ns.play("bikes");
ns.play2("bikes");
```

This code plays the recorded stream named "bikes.flv" within the application to which you are connected with `NetConnection.connect()`.

Different file types require prefixes and file extensions. For example, the following code plays the file "bikes.f4v",

```
ns.play("mp4:bikes.f4v");
```

For detailed information about `play()` and `play2()`, their parameters, and when to use each method, see the NetStream class in the *ActionScript 3.0 Reference*.

Playing streams nested in subfolders

To play a stream, specify the codec before you specify the path to the stream. FLV streams don't require a codec prefix, but F4V/MP4 files, MP3 files, and RAW files do.

Suppose you're using the default vod application. By default, the vod application is configured to look for streams in the applications/vod/media folder. Suppose a stream called "sample" is nested in the applications/vod/media/final folder. The following examples are for the client-side `NetStream.play()` method:

```
ns.play("mp4:final/sample.f4v",0,-1)  // F4V/MP4 files
ns.play("raw:final/sample",0,-1) // RAW files
ns.play("mp3:final/sample",0,-1) //MP3 files
ns.play("final/sample",0,-1). // FLV files
```

The following are examples for the Server-Side ActionScript `Stream.play()` method:

```
myStream.play("mp4:final/sample.f4v",0,-1)  // F4V/MP4 files
myStream.play("raw:final/sample",0,-1)  // RAW files
myStream.play("mp3:final/sample",0,-1)  // MP3 files
myStream.play("final/sample",0,-1)  // FLV files
```

Naming streams

Stream names cannot contain any of the following characters: `\ / : * ? " < > |`.

Managing the client buffer

- Very low frame rate H.264 videos can take a long time to start if the buffer is too short.

  H.264 video requires 64 messages before it begins playback. For example, at 15 fps, 2 seconds of buffer holds 30 samples. In this case, the server and the player wait for over 4 seconds for the 64 images to arrive, even if they're small.

- Server-side streams do not play if the file size is less than either the configured buffer time (`MinBufferTime` tag in the Application.xml configuration file), or 2 seconds.

- When playing a stream, set `NetStream.bufferTime` to at least `.1`. Set the value smaller for live applications. Set the value larger (3-5 seconds) for on-demand applications.

Mapping URIs to local and network drives

Use virtual directories to simplify mapping URIs to local and network drives. Virtual directories let you publish and store media files in different, predetermined locations, which can help you organize your media files. Configure virtual directories in the `VirtualDirectory/Streams` tag of the Vhost.xml file.

One way you can use directory mapping is to separate storage of different kinds of resources. For example, your application could allow users to view either high-bandwidth video or low-bandwidth video, and you might want to store high-bandwidth and low-bandwidth video in separate folders. You can create a mapping wherein all streams that start with *low* are stored in a specific directory, C:\low_bandwidth, and all streams that start with *high* are stored in a different directory:

```
<VirtualDirectory>
    <Streams>low;c:\low_bandwidth</Streams>
    <Streams>high;c:\high_bandwidth</Streams>
</VirtualDirectory>
```

When the client wants to access low-bandwidth video, the client calls `ns.play("low/sample")`. This call tells the server to look for the sample.flv file in the c:\low_bandwidth folder.

Similarly, a call to `ns.play("high/sample")` tells the server to look for the sample.flv file in the c:\high_bandwidth folder.

The following table shows three examples of different virtual directory configurations, including mapping to a local drive and a network drive, and how the configurations determine the directory to which a recorded stream is published. In the first case, because the URI specified (`"myStream"`) does not match the virtual directory name that is specified (`"low"`), the server publishes the stream to the default streams directory.

Mapping in Vhost.xml **<VirtualDirectory><Streams> tag**	**URI in NetStream call**	**Location of published stream**
low;e:\fmsstreams	"myStream"	c:\...*rootinstall*\applications\yourApp\streams_definst_\myStream.flv
low;e:\fmsstreams	"low/myStream"	e:\fmsstreams\myStream.flv
low;\\mynetworkDrive\share\fmsstreams	"low/myStream"	\\mynetworkDrive\share\fmsstreams\myStream.flv

More Help topics

"Getting started streaming media" on page 1

Capturing video snapshots

This feature enables you to get a thumbnail snapshot of a given video, including sound, for display purposes.

Flash Player clients are permitted to access data from streams in the directories specified by the `Client.audioSampleAccess` and `Client.videoSampleAccess` properties. See .

To access data, call `BitmapData.draw()` and `SoundMixer.computeSpectrum()` on the client. For more information about accessing raw audio, see Accessing raw sound data.

Handling metadata in streams

A recorded media file often has metadata encoded in it by the server or a tool. The Flash Video Exporter utility (version 1.1 or later) is a tool that embeds a video's duration, frame rate, and other information into the video file itself. Other video encoders embed different sets of metadata, or you can explicitly add your own metadata.

The NetStream object that plays the stream on the client dispatches an `onMetaData` event when the stream encounters the metadata. To read the metadata, you must handle the event and extract the `info` object that contains the metadata. For example, if a file is encoded with Flash Video Exporter, the `info` object contains these properties:

`duration`	The duration of the video.
`width`	The width of the video display.
`height`	The height of the video display.
`framerate`	The frame rate at which the video was encoded.

More Help topics

"Example: Add metadata to live video" on page 246

Using XMP metadata

You can deliver Adobe Extensible Metadata Platform (XMP) metadata embedded video streaming through Adobe Media Server to Flash Player. Adobe Media Server supports XMP metadata embedded in FLV and MP4/F4V formats. Adobe Media Server 3.5 supports one XMP metadata packet per MP4/F4V file.

With XMP metadata, you have a communication system that provides critical media information from media creation to the point where media is viewed. XMP information you add during the production process can add to the interactive experience of the media. In addition, speech-to-text metadata embedded within files and encoded from Adobe encoding tools such as Adobe Media Encoder can be delivered. AMF0 and AMF3 connections are supported. XMP metadata can be internal information about the file or information for end users.

For example, you could create a trailer in Adobe® Premiere® and transfer the metadata to the FLV file. When users view the file, they can use Flash Player 10 search to look for metadata and jump to a specific location in the file. When NetStream plays the content, an `onXMPData` message with the single field `data` is sent as a callback. The `data` field contains the entire XMP message from the media file.

For detailed information about XMP, see www.adobe.com/go/learn_fms_xmp_en.

Example: Media player

This tutorial uses ActionScript 3.0 to add a Video object to the Stage to display video. For more information about working with video, see the "Working with Video" chapter in *ActionScript 3.0 Developer's Guide* at www.adobe.com/go/learn_fms_video_en.

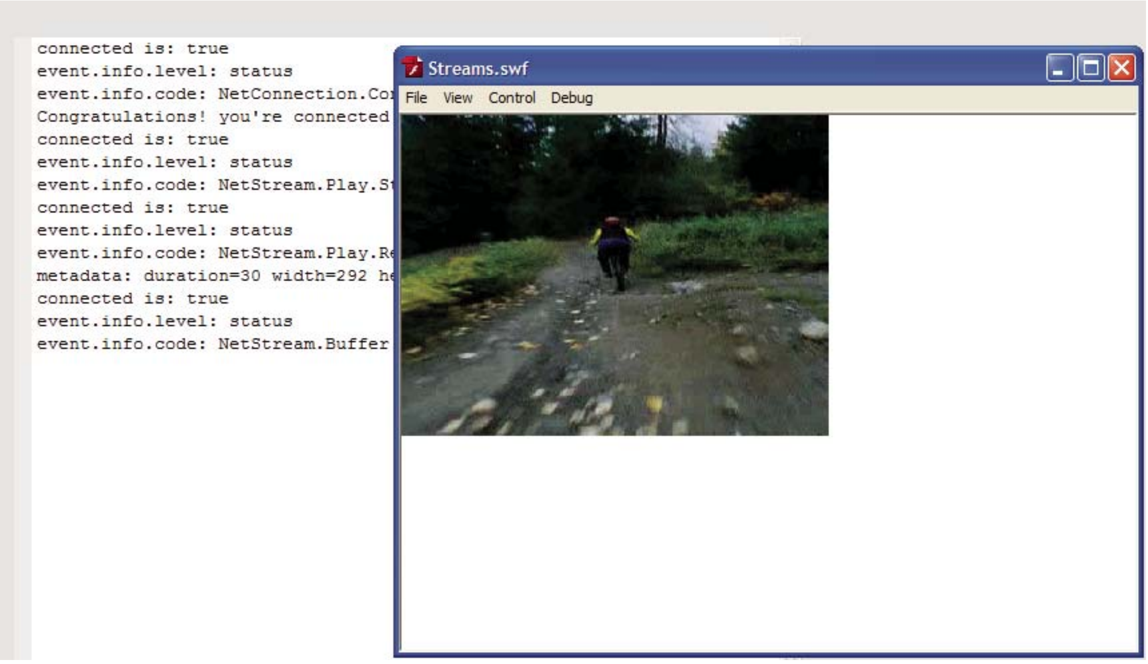
This tutorial provides the simplest example of displaying video for learning. To build a more robust video player, see the Open Source Media Framework.

Note: *This example uses the MediaPlayer sample, MediaPlayer.as, from the rootinstall/documentation/samples folder.*

Run the example in Flash

1. Create a *rootinstall*/applications/mediaplayer folder.
2. Copy the *rootinstall*/documentation/samples/MediaPlayer/streams folder to the *rootinstall*/applications/mediaplayer folder so you have the following:

   rootinstall/applications/mediaplayer/streams/_definst_/bikes.flv
3. In Flash, open the MediaPlayer.fla file from the *rootinstall*/documentation/samples/MediaPlayer folder.
4. Select Control > Test Movie. The video plays without sound and the Output window displays messages.



```
connected is: true
event.info.level: status
event.info.code: NetConnection.Co
Congratulations! you're connected
connected is: true
event.info.level: status
event.info.code: NetStream.Play.St
connected is: true
event.info.level: status
event.info.code: NetStream.Play.Re
metadata: duration=30 width=292 he
connected is: true
event.info.level: status
event.info.code: NetStream.Buffer
```

The Output window and the video in test-movie mode

You can watch the output as the stream plays and the connection status changes. The call to `NetStream.play()` triggers the call to `onMetaData`, which displays metadata in the console window, like this:

```
metadata: duration=30 width=292 height=292 framerate=30
```

Run the example in Flash Builder

1 Open MediaPlayer.as in Flash Builder.

2 Choose Run > Debug. For Project, choose MediaPlayer. For Application file, choose MediaPlayer.as.

3 Click Debug.

The video runs in an application window. Click the Flash Builder window to see the output messages.

Write the main client class

1 Create an ActionScript 3.0 class. Import NetConnection, NetStream, and any other classes you need:

```
package {
    import flash.display.Sprite;
    import flash.net.NetConnection;
    import flash.events.NetStatusEvent;
    import flash.net.NetStream;
    import flash.media.Video;
...
}
```

2 Create a new class, MediaPlayer, and declare the variables you'll need within it:

```
public class MediaPlayer extends Sprite
{
    var nc:NetConnection;
    var ns:NetStream;
    var video:Video;
...
}
```

3 Define the constructor: create a NetConnection object and add an event listener to it, and connect to the server:

```
public function MediaPlayer()
    {
        nc = new NetConnection();
        nc.addEventListener(NetStatusEvent.NET_STATUS, netStatusHandler);
        nc.connect("rtmp://localhost/mediaplayer");
    }
```

4 Create a `netStatusHandler` function that handles both NetConnection and NetStream events:

```
 private function netStatusHandler(event:NetStatusEvent):void{
    trace("event.info.level: " + event.info.level + "\n", "event.info.code: " +
event.info.code);
    switch (event.info.code){
        case "NetConnection.Connect.Success":
            // Call doPlaylist() or doVideo() here.
            doPlaylist(nc);
            break;
        case "NetConnection.Connect.Failed":
            // Handle this case here.
            break;
        case "NetConnection.Connect.Rejected":
            // Handle this case here.
            break;
        case "NetStream.Play.Stop":
            // Handle this case here.
            break;
        case "NetStream.Play.StreamNotFound":
            // Handle this case here.
            break;
        case "NetStream.Publish.BadName":
            trace("The stream name is already used");
            // Handle this case here.
            break;
     }
}
```

***Note:** To see the full list of event codes that are available, see `NetStatusEvent.info` in the ActionScript 3.0 Reference.*

1 Create a NetStream object and register a `netStatus` event listener:

```
private function connectStream(nc:NetConnection):void {
    ns = new NetStream(nc);
    ns.addEventListener(NetStatusEvent.NET_STATUS, netStatusHandler);
    ns.client = new CustomClient();
    ...
```

Notice that you set the `client` property to an instance of the CustomClient class. CustomClient is a separate class that defines some special event handlers.

2 Create a Video object and attach the stream to it:

```
    video = new Video();
    video.attachNetStream(ns);
```

In ActionScript 3.0, call `Video.attachNetStream()`—not `Video.attachVideo()` as in ActionScript 2.0—to attach the stream to the Video object.

3 Call `NetStream.play()` to play the stream and `addChild()` to add it to the Stage:

```
    ...
    ns.play("bikes", 0);
    addChild(video);
}
```

The URI of the stream you pass to `NetStream.play()` is relative to the URI of the application you pass to `NetConnection.connect()`.

Write the client event handler class

You also need to write the CustomClient class, which contains the `onMetaData` and `onPlayStatus` event handlers. You must handle these events when you call `NetStream.play()`, but you cannot use the `addEventListener()` method to register the event handlers.

1 In your main client class, attach the new class to the `NetStream.client` property:

```
ns.client = new CustomClient();
```

2 Create the new client class:

```
class CustomClient {
}
```

3 Write a function named `onMetaData()` to handle the `onMetaData` event:

```
public function onMetaData(info:Object):void {
    trace("metadata: duration=" + info.duration + " width=" + info.width +
        " height=" + info.height + " framerate=" + info.framerate);
}
```

4 Write a function named `onPlayStatus()` to handle the `onPlayStatus` event:

```
public function onPlayStatus(info:Object):void {
    trace("handling playstatus here");
}
```

Checking video files

Checking video files created or modified with third-party tools

Third-party tools are available to create and modify FLV and F4V files, but some of the tools create files that do not comply with the FLV and F4V standards. Common problems include bad timestamps, invalid `onMetaData` messages, bad message headers, and corrupted audio and video. The FLVCheck tool can be used to analyze FLV and F4V files before they are deployed on the server. In addition, the tool can also add or update metadata to reflect file duration correctly. The tool verifies that metadata is readable, specifies an accurate duration, and checks that the file is seekable by the server. The tool supports unicode filenames.

Note: *The FLVCheck tool does not correct file content corruption. The tool does fix metadata by scanning the Duration and Can Seek To End metadata fields. The tool can then merge the server metadata with the data present in the file.*

Checking other video files

Adobe Media Server supports playback and recording of H.264-encoded video and HE-AAC-encoded audio within an MPEG-4–based container format. A subset of the MPEG-4 standards are supported. All MP4 files and Adobe F4V files are part of the supported subset.

For MPEG-4–based container formats, use the FLVCheck tool to verify that the server can play back your files.

***Note:** The FLVCheck tool does not correct corrupted H.264-encoded files or make any other fixes to MP4/F4V files.*

Check a video file with the FLVCheck tool

The FLVCheck tool is a command line program; the executable is named flvcheck.

1 Open your operating system's command prompt and change directories to *rootinstall*/tools.

2 Use the following syntax to run the FLVCheck tool:

```
flvcheck [-option] <file ...>
```

For example, to check two files:

```
flvcheck -f abc.flv ../test/123.flv
```

The following table describes the command line options available.

Option	Description
`-f [ --file ]` file ...	Specifies the path to the video file(s) being checked. Relative paths may be used. (Avoid using the "\\xd3 character; try the "/" character instead.)
`-v [--verbose]`	Sets the verbose flag.
`-V [ --version ]`	Prints version information.
`-n [ --nobanner ]`	Turns off header.
`-h [ --help ]`	Provides a description of options and an example.
`-d [ --duration ]`	Specifies the margin of error, in seconds, that FLVCheck reports. (The default is 2 seconds.) When validating metadata, the absolute difference between `metadata_duration` and `actual_duration` is calculated and compared against the margin specified in this command. If the margin is exceeded, the server logs a warning that the metadata duration is incorrect. If the margin has not been exceeded, nothing will be logged. To get the exact duration, specify `-d 0`.
`-q [ --quiet ]`	Specifies that only the status code, not the text output, be returned. The `--help` option overrides this option.
`-w [ --warnings]`	Display warnings.
`-W [--warnings_as_errors]`	Treat warnings as errors.
`-s [ --fixvideostall ]`	Fix a stall in video playback (FLV only).
`-u [ --usage ]`	Displays an example and information about command-line parameters.
`-m [ --fixmeta ]`	(FLV files only) If a metadata tag is corrupted, creates a new copy of the original FLV file in the same directory as the original, with corrected metadata. (If the file contains no errors, a backup file is not created.) Only the Duration and Can Seek To End metadata fields are corrected.

3 If the FLVCheck tool finds no errors in the FLV file, the status code returned is 0. If there are one or more errors, a positive number indicating the total number of invalid files found is returned. If a return code of -1 is returned, an invalid command-line parameter was specified.

Errors and warnings are logged in a log file (stdout).

4 (FLV files only) If an error is returned from an FLV file due to a metadata error, you can use the tool to try to correct the problem. Try the following:

a Use the `-m` option to try to fix the metadata in the file:

```
flvcheck -m <file> [-quiet] [-help]
```

b Use the `-d` option to change the duration field margin of error. The duration field in the metadata may be inaccurate by a few seconds. For example, `flvcheck -f abc.flv -d 5` would allow the metadata duration to be inaccurate +/- 5 seconds.

Other types of errors cannot be fixed using the FLVCheck tool. MP4/F4V files cannot be fixed using the FLVCheck tool.

FLVCheck errors

If an error is found, the error is logged to the stdout file in the following format: Date, Time, ErrorNumber, ErrorMessage, and FileName. The possible error numbers, types of errors, and messages are as follows.

Error numbers	Error type	Error messages
-2	General	Invalid file system path specified.
-3	General	File not found.
-4	General	Cannot open file.
-5	General	File read error. Adobe Media Server cannot read the file, indicating that the encoding of part or all of the file is not compatible with the codecs that are supported.
-6	General	Cannot create corrected file. This error occurs if you run the tool with the `-m` option set, but the tool cannot create a file with corrected metadata.
-7	FLV	Invalid FLV signature.
-8	FLV	Invalid FLV data offset.
-9	FLV	Invalid FLV message footer.
-10	FLV	Unrecognized message type.
-11	FLV	Found backward timestamp.
-12	FLV	Unparsable data message.
-13	MP4	File does not contain a movie box. This error occurs if the MP4 file is empty.
-14	MP4	File does not contain any valid tracks. This error could occur if the MP4 file contains audio or video encoded with unsupported codecs.
-15	MP4	Too many tracks. Maximum allowed is 64.

Error numbers	Error type	Error messages
-16	MP4	Only one sample type allowed per track.
-17	MP4	Box is too large.
-18	MP4	Truncated box. The reported length of a box is longer than the remaining length of the file. The file may have been truncated, or the reported box length may be invalid.
-19	MP4	Duplicate box.
-20	MP4	Invalid box version.
-21	MP4	Invalid movie time scale.
-22	MP4	Invalid number of data entries in box.
-23	MP4	Invalid sample size.
-24	MP4	Invalid chapter time.
-25	MP4	Too many tag boxes. Max is 64.
-26	General	File appears to be FLV with wrong extension.
-27	MP4	Unsupported DRM scheme.
-28	MP4	Error reading MP4 tables.
-29	MP4	File contains unexpected movie fragments.
-30	MP4	File contains out-of-order movie fragments.
-32	RAW	Index or Contexts file missing or corrupted.
-33	RAW	Index File Version %x Not Supported. FMS Requires Version %x.
-34	RAW	Failed read in segment file %s. File missing or corrupted.
-35	RAW	Truncated message in segment file %s.
-36	RAW	Unrecognized message type in segment file %s.
-37	RAW	Invalid message footer in segment file %s.
-38	RAW	Segment file %s does not match index file.

FLVCheck warnings

Generally, warnings are informative and are not fatal errors; Adobe Media Server will ignore the error that caused the warning and continue to load and play back the video or audio file, but you may experience problems with playback. Warnings are logged to the stdout file in the following format: Date, Time, Warning Number, Warning Message, and File Name.

Warning number	Warning type	Message
-100	General	Metadata duration is missing or is incorrect.
-101	FLV	canSeekToEnd is false.
-102	MP4	Unrecognized box.
-103	MP4	Found incomplete track.
-104	MP4	Found duplicate video track. Ignoring...

Warning number	Warning type	Message
-105	MP4	Found duplicate audio track. Ignoring...
-106	MP4	Found duplicate data track. Ignoring...
-107	MP4	Track has unsupported sample type. Adobe Media Server ignores (will not play back) tracks that are encoded with unsupported codecs.
-108	MP4	Invalid video codec. This warning indicates that a track has an invalid video codec. Adobe Media Server cannot play back the track.
-109	MP4	Invalid audio codec. This warning indicates that a track has an invalid audio codec. Adobe Media Server cannot play back the track.
-110	FLV	Video may appear stalled due to lack of audio data.
-111	MP4	File has unsupported metadata format.
-112	MP4	Box has extraneous bytes at end.
-113	FLV	Video messages found but video flag not set.
-114	FLV	Audio messages found but audio flag not set.
-115	FLV	Video flag set but no video messages found.
-116	FLV	Audio flag set but no audio messages found.
-117	MP4	File is truncated. Will only be partially playable.
-118	MP4	Track contains unsupported edit list.
-119	FLV and RAW	Missing FLV metadata.
-120	MP4	Bad NellyMoser Frequency. Sample(s) skipped.
-121	MP4	Invalid Track Extends Box.
-122	MP4	Track contains unsupported sample flags.
-126	RAW	Found backward timestamp in segment file %s.

Handling errors

About error handling

As you build video applications, it is important to learn the art of managing connections and streams. In a networked environment, a connection attempt might fail for any of these reasons:

- Any section of the network between client and server might be down.
- The URI to which the client attempts to connect is incorrect.
- The application instance does not exist on the server.
- The server is down or busy.
- The maximum number of clients or maximum bandwidth threshold may have been exceeded.

If a connection is established successfully, you can then create a NetStream object and stream video. However, the stream might encounter problems. You might need to monitor the current frame rate, watch for buffer empty messages, downsample video and seek to the point of failure, or handle a stream that is not found. Inform customers of errors that occur during playback:

- The network connection fails during playback.
- The buffer empties before playback is complete.

To be resilient, your application needs to listen for and handle `netStatus` events that affect connections and streams. As you test and run your application, you can also use the Administration Console to troubleshoot various connection and stream events.

Handle a failed connection

If a connection cannot be made, handle the `netStatus` event *before* you create a NetStream object or any other objects. You may need to retry connecting to the server's URI, ask the user to reenter a user name or password, or take some other action.

The event codes to watch for and sample actions to take are as follows:

Event	Action
`NetConnection.Connect.Failed`	Display a message for the user that the server is down.
`NetConnection.Connect.Rejected`	Try to connect again.
`NetConnection.Connect.AppShutDown`	Disconnect all stream objects and close the connection.

***Note:** Use the SimpleConnectManage sample, SimpleConnectManage.as, written in ActionScript 3.0.*

Write client code to handle NetStatus events

❖ Create a NetConnection object and connect to the server. Then, write a `netStatus` event handler in which you detect each event and handle it appropriately for your application, for example:

```
public function netStatusHandler(event:NetStatusEvent):void
{
    trace("connected is: " + nc.connected );
    trace("event.info.level: " + event.info.level);
    trace("event.info.code: " + event.info.code);
    switch (event.info.code)
    {
         ...
         case "NetConnection.Connect.Rejected":
            trace ("Oops! the connection was rejected");
            // try to connect again
            break;
         case "NetConnection.Connect.Failed":
            trace("The server may be down or unreachable");
            break;
         case "NetConnection.Connect.AppShutDown":
            trace("The application is shutting down");
            // this method disconnects all stream objects
            nc.close();
            break;
        ...
    }
}
```

Handle a stream not found

If a stream your application attempts to play is not found, a `netStatus` event is triggered with a code of `NetStream.Play.StreamNotFound`. Your `netStatus` event handler should detect this code and take some action, such as displaying a message for the user or playing a standard stream in a default location.

Write the client code

❖ In your `netStatus` event handler, check for the `StreamNotFound` code and take some action:

```
private function onNetStatus(event:NetStatusEvent):void {
    switch (event.info.code) {
        case "NetStream.Play.StreamNotFound":
            trace("The server could not find the stream you specified");
            ns.play( "public/welcome");
            break;
        ...
    }
```

Working with playlists

About playlists

A *playlist* is a list of streams to play in a sequence. The server handles the list of streams as a continuous stream and provides buffering, so that the viewer experiences no interruption when the stream changes. You can use both client-side ActionScript and Server-Side ActionScript to create playlists.

Adobe Evangelist Jens Loeffler has written an Adobe DevNet article that uses client-side and server-side playlists to edit live streams into a highlight reel.

Create a client-side playlist

***Important:** There is a bug that impacts client-side playlists that are missing their first item. The first time a user plays such a playlist, the playlist skips the item and plays. Every subsequent time a user plays the playlist, the playlist does not play.*

This playlist uses the names of streams that are stored on the server. To change the playlist, you need to change the code in your application client.

***Note:** Use the MediaPlayer sample, MediaPlayer.as, written in ActionScript 3.0.*

1 Create a NetConnection object, connect to the server, and add a `netStatus` event handler.

2 Create a NetStream object and listen for `netStatus` events:

```
private function createPlayList(nc:NetConnection):void {
    stream = new NetStream(nc);
    stream.addEventListener(NetStatusEvent.NET_STATUS, netStatusHandler);
    stream.client = new CustomClient();
    ...
}
```

3 Attach the NetStream object to a Video object:

```
    video = new Video();
    video.attachNetStream(stream);
```

4 Define a series of `play()` methods on the NetStream object:

```
    stream.play( "advertisement", 0, 30 );
    stream.play( "myvideo", 10, -1, false );
    stream.play( "bikes", 0, -1, false );
    stream.play( "parade", 30, 120, false);
    addChild(video);
```

5 Listen for NetStream event codes in your `netStatus` event handler:

```
private function netStatusHandler(event:NetStatusEvent):void
{
    ...
    case "NetStream.Play.Stop":
        trace("The stream has finished playing");
        break;
    case "NetStream.Play.StreamNotFound":
        trace("The server could not find the stream");
        break;
}
```

This playlist plays these streams:

- A recorded stream named *advertisement.flv*, from the beginning, for 30 seconds
- The recorded stream *myvideo.flv*, starting 10 seconds in, until it ends
- The recorded stream *bikes.flv*, from start to end
- The recorded stream *parade.flv*, starting 30 seconds in and continuing for 2 minutes

Swap streams in a playlist

Flash Player 10 and Adobe Media Server 3.5

Swapping streams means to exchange one stream for another. While stream switching occurs at keyframes, swapping streams occurs at the stream boundary. Swapping streams is, therefore, useful with playlists. One use case is with playlists that contain content with advertising segments. After collecting statistics about usage patterns, you can swap one advertisement for another.

Use the `NetStream.play2()` method with the transition mode `SWAP` to swap streams in a playlist. The `NetStream.play2()`method takes a NetStreamPlayOptions object as a parameter. In the NetStreamPlayOptions object, specify the old stream, the stream to switch to, and the kind of transition to use—in this case, `NetStreamPlayTransitions.SWAP`.

For example, suppose a playlist is set to play Stream A, Stream B, and Stream C, in that order.

```
ns.play("streamA", 0, -1, true);
ns.play("streamB", 0, -1, false);
ns.play("streamC", 0, -1, false);
...
```

While Stream A plays, and before the server begins sending Stream C, you determine that you want to play Stream Z instead of Stream C. To perform this transition, use code like the following sample:

```
var param:NetStreamPlayOptions = new NetStreamPlayOptions();
param.oldStreamName = "streamC";
param.streamName = "streamZ";
param.transition = NetStreamPlayTransitions.SWAP
ns.play2(param);
```

The `SWAP` transition differs from the `SWITCH` transition. The call to swap streams must occur before the server delivers the old stream (in this example, `streamC`). If `streamC` is already in play, the server does not swap the content and sends a `NetStream.Play.Failed` event. If the server has not yet delivered `streamC,` the server swaps the content. The result is that `streamA`, `streamB`, and `streamZ` play.

When the server swaps to a stream with different content, the client application resets the buffer. The server swaps the stream at the start of the new stream, ensuring an uninterrupted experience.

Create a server-side playlist

A server-side playlist is a list of media played in sequence over a Server-Side ActionScript Stream object. A playlist can contain both live and recorded media.

A server-side playlist plays over a live stream (the Stream object). You can record the stream as an FLV or F4V file as it plays. If a playlist contains only FLV files, you can record it as an FLV file or as an F4V file. Otherwise , record it as an F4V file.

The following code creates a server-side playlist of two live streams and one recorded stream:

```
// Start the playlist when the application loads.
// This is a live playlist, it is not recorded.
application.onAppStart = function(){
    this.myStream = Stream.get("serverplaylist");
    // Play a live stream for 30 seconds.
    this.myStream.play("liveStream1", -1, 30);
    // Play a recorded stream in full after liveStream1 plays.
    this.myStream.play("mp4:recordedStream1.f4v", 0, -1, false);
    // Play another live stream for 30 seconds after recordedStream1 plays.
    this.myStream.play("liveStream2", -1, 30, false)
}
```

To play the playlist, call `NetStream.play("serverplaylist")` on the client. To play the playlist smoothly, set the client-side `NetStream.bufferTime` property to at least 1 second. (The default value is 0.1 seconds.)

To add media to a playlist, call `Stream.play()` and pass `false` for the `reset` parameter. When you pass `false`, the media doesn't start playing until the media currently playing has stopped.

The following server-side code plays two recorded media files consecutively and records them to the file "playlist.f4v". Each media file plays in its entirety. To play the playlist, call `NetStream.play("mp4:playlist.f4v")` on the client.

```
application.onAppStart = function(){
    this.clientStream = Stream.get("mp4:playlist.f4v");
    this.clientStream.record();
    this.clientStream.play("mp4:british.mp4", 0, -1);
    this.clientStream.play("mp4:shadows.mp4", 0, -1, false);
};
```

The following example was provided by Jens Loeffler on his blog flashstreamworks.com.

Use a server-side playlist to export a highlight clips during a DVR-enabled live event. Build the playlist (for example, an opening clip, then a selected part of the live event, then a section of an archived clip), and record it as an .f4v file. To play the highlight reel, just point your streaming player to the exported .f4v file.

```
application.myStream = Stream.get("mp4.highlights.f4v");
    if (application.myStream){
        application.myStream.record();
        application.myStream.play("mp4:titles.f4v", 0, 15);
        application.myStream.play("livesmith", -1, 30, false);
        application.myStream.play("mp4:smitharchive.mp4", 0, 30, false);
        application.myStream.play("mp4:closing.f4v", 0, 15, false);
    }
};
```

To play this example, the client calls `NetStream.play("mp4:highlights.f4v")`.

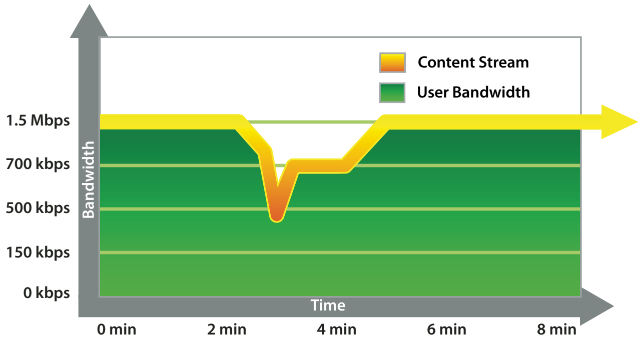
To play a server-side playlist, set the client-side `NetStream.bufferTime` property to at least 1 second. (The default value is 0.1 seconds.)

Dynamic streaming

About dynamic streaming

***Note:** The dynamic streaming API discussed is supported in Adobe Media Server 3.5 and later and Flash Player 10 and later.*

Adobe Media Server can receive commands to switch between versions of a content stream that are encoded at different bit rates. This feature lets your media application adapt to changing network conditions. It also lets your application adapt to clients with different capabilities, such as mobile devices with lower processing power and smaller screens. For example, suppose the server is streaming high-definition video to a client application but encounters poor network conditions. The server can switch to a standard-definition stream at a lower bit rate. If network conditions improve, the server can switch back to HD video. The transitions occur seamlessly in the client. Although network conditions have changed, the video streaming to the client is uninterrupted.



The server delivers streams that match bandwidth changes to maintain QoS

For optimal user experience, dynamic streaming requires the following:

- The different versions or pieces of content are synchronized: the video timelines must match.
- Audio or other data in each content stream is synchronized with the video data in that stream.

The server implements a transition between two pieces of recorded content in three possible ways, depending on the type of content that is being streamed:

- Video-only streams. Transitions occur at the nearest keyframe in the target timeline.
- Video and audio streams. Transitions occur at the audio sample that immediately precedes the nearest keyframe in the timeline of the target stream. The audio timelines of the initial and target streams must match, or an audio artifact results.
- Audio-only streams. Transitions occur at the nearest possible sample.

Implementing transitions between live video content is slightly more complicated. The streams must have timestamps that are synchronized closely enough, within 3-5 milliseconds, so that the server can select accurate transition points.

Using ActionScript to switch streams

Stream transitions occur on the server, but the command to switch streams and the determination to do so comes from the client application. The application developer includes logic to monitor download and playback statistics and to switch from the old stream to the new stream when appropriate.

Use the ActionScript 3.0 `NetStream.info` property and the NetStreamInfo class to monitor download and playback statistics. Use the `NetStream.play2()` method and the associated NetStreamPlayOptions class to change streams in mid-play.

Use the `NetStreamPlayOptions.offset` property to perform fast switching between streams. Use the default value of `NetStreamPlayOptions.offset` which is `-1`. The default value of `-1` indicates a fast switch time of `netstream.time + 3`. You can also specify a greater value, such as `netstream.offset + 4`, but do not use a lower value.

Important: *Do not use an value for* `NetStreamPlayOptions.offset` *that is lower than the default value.*

The application developer must also ensure that the client application has a playback buffer that is large enough to absorb any delay caused by the transition. Two factors that can cause a delay are the keyframe interval of the live stream and if the two streams being switched are not synchronized. For example, a 2-second buffer cannot accommodate a 3-second transition delay.

Important: *Set a value for* `NetStream.bufferTime` *of 10 seconds or greater.*

Adobe developed a new class called DynamicStream that extends the NetStream class. The DynamicStream class contains event listeners that monitor bandwidth, buffer usage, and dropped frames. The class switches streams based on that information. You can use the DynamicStream class to implement dynamic streaming, or you can use the DynamicStream class as a reference to write your own dynamic streaming algorithms. If you're migrating legacy code, it's a good idea to use the DynamicStream class.

Download the ActionScript class files and documentation for these classes from www.adobe.com/go/fms_dynamicstreaming.

Note: *These classes are not part of the ActionScript 3.0 library. These are custom classes developed by Adobe for Adobe Media Server users.*

Determining when to use dynamic streaming

Adobe recommends that you use dynamic streaming for content that meets some or all of the following criteria:

- Video with long duration
- Video with large file size
- HD video
- Video with larger dimensions, such as full screen video
- Content distributed to users who are more susceptible to bandwidth issues, such as home users, rather than corporate users

Encoding recommendations

To provide users with the best experience, when you encode the content, follow the recommendations in this DevNet article and in Will Law's presentation at Adobe MAX 2008.

The following is a summary of the recommendations:

- For the smoothest switching, encode audio with AAC.
- For streams with MP3 audio, keep the audio bit rates the same.
- Use the same audio sample rate when possible.
- To encode streams for lower bandwidth usage, encode the audio with a single channel (mono).
- Ensure that the video timelines of the streams are related and compatible.
- Use the same codecs and audio bitrates in all streams. If you don't, you may hear small audio pops when the streams switch.
- While not required, it is helpful if the keyframe interval (keyframe frequency) and frame rate (fps) are consistent across the different versions of content. A shorter keyframe interval lets the server switch streams more quickly, which means that the client can have a smaller playback buffer.

The following table shows various bit rates that you could use to encode a single piece of content:

Bit rate
150 kbps
300 kbps
500 kbps
700 kbps
1.5 Mbps (full web HD)

Determining when to switch streams

You can consider many factors when determining when to switch streams, such as the buffer length, number of bytes downloaded, and number of frames dropped. The DynamicStream and DynamicStreamItem classes, which you can download from adobe.com, are built with these factors in mind and contain the logic required for a dynamic streaming application.

If you prefer to develop your own application logic, it may be helpful to use the following strategy for streaming video:

1. For initial playback, select the lowest bit rate that is appropriate for the screen or device. For example, if the video plays in the web browser on a standard computer, an appropriate stream for initial playback is 300 kbps at 320 x 240.
2. To start playback quickly, select a small buffer length.
3. Once playback begins, increase the buffer length to at least 10 seconds.
4. Do not use a value for `NetStreamPlayOptions.offset` that is lower than the default.
5. Begin monitoring the client bandwidth (`NetStream.info.maxBytesPerSecond`) and buffer size (`NetStream.bufferLength`) as it fills.

   When current bandwidth is sufficient, the buffer fills quickly and stays steady. If the bandwidth begins to drop, the buffer starts to empty.
6. If the client bandwidth exceeds the requirements of the stream and the buffer is filling or is full, you can switch to higher-resolution content.

   Verify that client bandwidth is sufficient before switching. In addition to client bandwidth and buffer length, you can check additional statistics, such as the number of dropped frames (`NetStream.info.droppedFrames`).
7. After each transition to higher-resolution content, continue to monitor the buffer every 5 seconds, using a timer.

   If the buffer begins to empty, switch to lower-resolution content and monitor the buffer more frequently, such as after every 2 seconds.
8. Continue to upgrade while bandwidth is plentiful and the buffer is filling or full. For the best user experience, be conservative. Upgrade only when the reported bandwidth exceeds stream requirements by a solid margin.

More Help topics

Using the DynamicStream classes

ActionScript Guide to Dynamic Streaming

Check client bandwidth

Monitor the client bandwidth to help determine when switching streams is desirable. When client bandwidth is good, the client application can request the server to switch to a higher video bit rate. When client bandwidth is low, the client application can request the server to switch to a lower bit rate.

To measure bandwidth, use the NetStream.info property. A call to NetStream.info returns a NetStreamInfo object with properties that reflect the rate of incoming audio, video, and data bytes of the stream. With information about the incoming data rate, you can deduce the quality of the bandwidth.

Specifically, use the *byteCount and *bytesPerSecond properties in the NetStreamInfo class (or, in ActionScript 2.0, the object returned by NetStream.getInfo()). For details on these properties, see the ActionScript Language References.

One way to measure the client bandwidth is to measure the NetStreamInfo.byteCount property over a period of time to get the value of bytes per second and when a NetStream.Buffer.Full status event is received. This value approximates the maximum bandwidth available. Then compare the available bandwidth to the available bit rates and implement transitions as needed.

***Note:** The byteCount property does not return the same value as sc-stream-bytes in the server Access log. The byteCount property is a Quality of Service designed to provide data that can help you decide when to switch streams. Do not use the byteCount property for billing.*

Check for dropped frames

In addition to monitoring the buffer, check for dropped frames. Switch to a stream with a lower bit rate if too many frames are being dropped. Use the NetStreamInfo.droppedFrames property. This read-only property is a number and returns the number of video frames dropped in the current NetStream playback session.

One strategy to determine the rate of dropped frames is as follows: using a timer, calculate the difference between the current value of dropped frames and a previous value. Store that difference in a variable, droppedFPS. Monitor the current number of incoming frames per second in another variable, currentFPS. If droppedFPS exceeds 20% of the value of currentFPS, switch to a lower bit rate.

Switch streams

To request a transition between streams with the same content encoded at different bit rates, the client application uses the NetStream.play2() method. This method takes as a parameter a NetStreamPlayOptions object, which indicates how the server switches streams.

***Note:** The NetStream.play2() method extends the NetStream.play() method.*

The NetStreamPlayOptions object contains the following properties:

Property	Description
oldStreamName	The name of the stream currently being played (the old stream).
streamName	The name of the new stream to play; the stream to switch to.
start	The start time of the new stream to play. For most dynamic streaming purposes, the default value of -2 is appropriate. It tells the application to play the live stream specified in streamName. If a live stream of that name is not found, Flash Player plays the recorded stream specified in streamName. If a live or a recorded stream is not found, Flash Player opens a live stream named streamName, even though no one is publishing on it. When someone does begin publishing on that stream, the application begins playing it.
len	The duration (length) of the playback. For most dynamic streaming purposes, the default value of -1 is appropriate. This value means that the application plays a live stream until it is no longer available or plays a recorded stream until it ends.

Property	Description
`transition`	The transition mode. Possible values are constants in the NetStreamPlayTransition class. The one most applicable to switching between the same content at different bit rates is `SWITCH`. For information on other modes, see the NetStreamPlayTransition class in the ActionScript 3.0 Language Reference.

The following code example uses the `SWITCH` option to tell the server to switch to a higher bit rate stream. The example does not pass a value for `oldStreamName`, which tells the server to switch to the new stream at the next logical keyframe. This technique provides the smoothest viewing experience. (When using playlists, pass a value for `oldStreamName`; see "Swap streams in a playlist" on page 209.) In most dynamic stream-switching cases with a recorded video+audio stream, you can keep the default values of `start` and `len`, as in the example.

When the client requests a transition, the server sends a `NetStatusEvent.NET_STATUS` event with the code `NetStream.Play.Transition`. (In ActionScript 2.0, it sends an `onStatus` event with the same code.) The server sends this event to the client almost immediately, which indicates that the operation has succeeded. When the first frames of the new stream render, the server sends an `onPlayStatus` message with the code `NetStream.Play.TransitionComplete`. This event lets the client know exactly when the new stream has started to render.

If a client seeks after Flash Player sends a `NetStream.Play.Transition` message, the streams switch successfully, but Flash Player does not send a `NetStream.Play.TransitionComplete` message. The player doesn't send the message because after the seek it enters a new state and cannot send status events about the old state. The player behaves the same way for other callback methods such as `onMetaData()`.

The following example handles a stream transition:

```
var stream:NetStream = new NetStream(connection);
stream.addEventListener(NetStatusEvent.NET_STATUS, netStatusHandler);
stream.client = new CustomClient();
var video:Video = new Video();
video.attachNetStream(stream);
stream.play("streamA_150kbps");
addChild(video);
...
//Set up the transition to 700 kpbs content
var param:NetStreamPlayOptions = new NetStreamPlayOptions();
param.streamName = streamA_700kpbs;
param.transition = NetStreamPlayTransitions.SWITCH;
stream.play2(param);

//Handler function for the Transition event
class CustomClient{
    public function onPlayStatus(info:Object):void {
    trace("switch time: time=" + info.time + " name=" + info.name + " type=" + info.type);
    }
}
```

The NetStream.Play.Transition status event contains a reason field. Use this field to get additional information about the state of a transition request. The reason field usually contains the code "NetStream.Transition.Success", meaning the transition request succeeded and was processed normally. When switching between live streams, it is possible that the server could not find a synchronization point between the two streams. In this case, the server forces a transition to occur at an arbitrary frame and the reason field contains the code "NetStream.Transition.Forced". This situation can occur under the following circumstances:

- The two streams being switched don't have the same timeline, and therefore the server cannot select a time to perform the switch.
- The keyframe interval of the new stream is longer than the client's playback buffer, which is the maximum amount of time the server will wait for a transition. Since the server did not see a keyframe, it cannot select a frame for the switch.
- The live queue delay for the live streams is longer than the client's playback buffer, which creates a delay similar to a long keyframe interval.

Handling metadata during stream transitions

Adobe Media Server sends a NetStream.Play.TransitionComplete status event when the bits of a stream transition render to the client. When switching to a new stream, the onMetadata message for the new stream is sent immediately following the NetStream.Play.TransitionComplete status event. Listen for the TransitionComplete event before capturing the metadata. If the stream is live, all data keyframes associated with the stream are transmitted.

Setting the client buffer

Buffering manages fluctuations in bandwidth while a video is playing.

To create a good experience for users, set the buffer to a small value initially. A smaller value lets the stream begin playing relatively quickly. Once playback begins, increase the buffer to a larger value. A larger value ensures that the stream plays more smoothly regardless of noise or interruptions on the network.

To create the best experience for users, monitor the progress of a video and manage buffering as the video downloads. Consider setting different buffer sizes for different users, to ensure the best playback experience. One choice is to measure client bandwidth and set an initial buffer size based on it.

Monitor the buffer size in the client to determine when to switch streams. When client bandwidth is good, the amount of data in the buffer grows or the buffer is full. The client can request the server to switch to a higher video bit rate. When client bandwidth is low, the amount of data in the buffer shrinks or the buffer empties. The client can request the server to switch to a lower bit rate.

Call NetStream.info() to get a NetStreamInfoObject with properties that reflect the current statistics of the stream. The properties that deal with the buffer are the BufferLength and BufferByteLength properties. For details on these properties, see the ActionScript 3.0 Reference.

While the stream is playing, you can also detect and handle netStatus events. For example, when the buffer is full, the netStatus event returns an info.code value of NetStream.Buffer.Full. When the buffer is empty, another event fires with a code value of NetStream.Buffer.Empty. When the data has finished streaming, the NetStream.Buffer.Flush event is dispatched. You can listen for these events and set the buffer size smaller when empty and larger when full.

***Note:** Flash Player 9 Update 3 and later no longer clear the buffer when a stream is paused. This feature allows viewers to resume playback without experiencing any hesitation. You can also use `NetStream.pause()` in code to buffer data. You could buffer data while viewers are watching a commercial, for example, and then unpause the stream when the main video starts. For more information, see the `NetStream.pause()`.*

Set buffer time

To change the buffer time, in seconds, set the `NetStream.bufferTime` property:

```
ns.bufferTime(10);
```

The right size for the buffer varies depending on user bandwidth, and the following values are only suggestions. 5-10 seconds is a good initial buffer size for fast connections; 10 seconds is a good initial buffer size for slow connections. After playback starts, 30-60 seconds is a good buffer size.

Handle buffer events

This example shows how to detect buffer events and adjust the buffer time dynamically, as events occur. Highlights of the code are shown here; to see the complete sample, see the Buffer.as sample file.

1 In the constructor function of your main client class, create a NetConnection object and connect to the server (see Buffer.as in the documentation/samples/Buffer directory in the Adobe Media Server root install directory):

```
nc = new NetConnection();
nc.addEventListener(NetStatusEvent.NET_STATUS, netStatusHandler);
nc.connect("rtmp://localhost/Buffer");
```

2 Write a `netStatus` event handler, checking for success, failure, and full buffer and empty buffer events and changing buffer size accordingly:

```
private function netStatusHandler(event:NetStatusEvent):void {
    switch (event.info.code) {
        case "NetConnection.Connect.Success":
            trace("The connection was successful");
            connectStream(nc);
            break;
        case "NetConnection.Connect.Failed":
            trace("The connection failed");
            break;
        case "NetConnection.Connect.Rejected":
            trace("The connection was rejected");
            break;
        case "NetStream.Buffer.Full":
            ns.bufferTime = 30;
            trace("Expanded buffer to 30");
            break;
        case "NetStream.Buffer.Empty":
            ns.bufferTime = 8;
            trace("Reduced buffer to 8");
            break;
    }
}
```

3 Write a custom method to play a stream. In the method, set an initial buffer time, for example, 2 seconds:

```
private function connectStream(nc:NetConnection):void {
    ns:NetStream = new NetStream(nc);
    ns.addEventListener(NetStatusEvent.NET_STATUS, netStatusHandler);
    ns.client = new CustomClient();

    video = new Video();
    video.attachNetStream(ns);

    ns.play( "bikes", 0 );
    ns.bufferTime = 8;
    trace("Set an initial buffer time of 8 seconds");

    addChild(video);
}
```

4 Create the `onMetaData()` and `onPlayStatus()` event handlers:

```
public function onMetaData(info:Object):void {
        trace("Metadata: duration=" + info.duration + " width=" + info.width + " height=" +
info.height + " framerate=" + info.framerate);
}

public function onPlayStatus(info:Object):void {
    switch (info.code) {
        case "NetStream.Play.Complete":
            trace("The stream has completed");
            break;
    }
}
```

These event handlers are needed when you call `NetStream.play()`.

Recognizing transitions in log files

You can track stream events through access logs. Use the logs to differentiate a single stream play with transitions from multiple plays of a stream or different streams. When a transition for a single content stream occurs, the server tracks the status of the stream as a transition. The server logs a stop event for the original stream and a play event for the new stream. Normal stop and play events (that is, a stop or play without a transition) have a status code of 200. Stream transitions have a status code of 210. The access logs provide the following additional information:

Field	Description
`x-sid`	The ID of a stream. This ID is unique for the client session but not across sessions
`x-trans-sname`	The name of the stream that the server is transitioning from (the original stream)
`x-trans-sname-query`	The query stream portion of the stream name for the stream that the server is transitioning from
`x-trans-file-ext`	The filename extension portion of the stream name for the stream that the server is transitioning from

If you use a log processor, ensure that it looks at both the status codes and the x-sid values. Look at both values to recognize that a transition occurred on a single logical stream. As with normal streams, stream transitions occur in play/stop pairs. Your log processor can track stream transitions by recognizing stop events that have a 210 status code followed by play events with a 210 status code on the same stream. By looking at the status code, the log processor can also differentiate a stream transition from a play or stop event without a transition.

Reconnecting streams when a connection drops

Adobe Media Server 3.5.3, Flash Player 10.1

About reconnecting streams

You can build applications that support seamless playback when a connection is dropped or when a client switches from a wired to a wireless network connection. For an example of reconnecting streams, see the Stream Reconnect and Smart Seek example hosted by Adobe.

To provide seamless playback, use the ActionScript 3.0 `NetStream.attach()` method to attach the same NetStream object to a reconnected NetConnection object. You can also use this feature for load balancing.

Note: *If the Flash Player version is less than 10.1 or the Adobe Media Server version is less than 3.5.3, the stream closes when the connection drops.*

Stream Reconnect ActionScript API

Note: *The Stream Reconnect ActionScript API is ActionScript 3.0. These APIs are not supported in ActionScript 2.0.*

The following ActionScript 3.0 APIs enable you to reconnect a stream:

- `NetStream.attach(connection:NetConnection)`

  Attaches a stream to a NetConnection object.

- `NetStreamPlayTransitions.RESUME`

  The `RESUME` mode causes Flash Player 10.1 and greater to request data from a new connection at the same location where it dropped the previous connection. Flash Player aligns the stream across the two connections so no artifacts or jumps are observed in the video playback. Use this mode when you reconnect a stream that was dropped due to server issues or other connection problems.

- `NetStreamPlayTransitions.APPEND_AND_WAIT`

  The `APPEND_AND_WAIT` mode tells the server to build the playlist but not to stream it. Use this mode to rebuild a playlist after losing a connection and establishing a new connection.

- `NetConnection.Connect.Closed`

  Use this event to reconnect a stream, `NetConnection.Connect.Closed`.

- `NetConnection.Connect.NetworkChange`

  Notifies the client that a network connection has changed. Don't use this event to reconnect a stream, use `NetConnection.Connect.Closed`.

For detailed information about these APIs, see the ActionScript 3.0 Language Reference.

Using the Stream Reconnect ActionScript API

When a NetConnection closes due to a network change, the stream keeps playing using the existing buffer. Meanwhile, client-side ActionScript code reconnects to the server and resumes playing the stream.

Note: *If the Flash Player version is less than 10.1 or the Adobe Media Server version is less than 3.5.3, the stream closes as soon as the network connection is lost.*

Reconnecting a single stream

The following workflow reconnects a single stream:

1 Call `NetConnection.connect()` to connect to server A.

2 Create a NetStream. Set `NetStream.bufferTime` to at least a few seconds so there is enough data to play while the connection is down.

3 Call `NetStream.play2()` and use `NetStreamPlayTransitions.RESET` to play a stream called "myStream".

4 Monitor the NetConnection for the `"NetConnection.Connect.Closed"` event. Reconnect to server A if the connection drops.

5 Call the `NetStream.attach(connection:NetConnection)` method to attach the NetStream to the new connection.

6 Call `NetStream.play2()` and use `NetStreamPlayTransitions.RESUME` to play the stream "myStream".

Flash Player tells the server where to resume playing the stream.

7 Do not call `NetConnection.close()` or `NetStream.close()` if you intend to reconnect it. Calling close closes and cleans up the connection or the stream, so it will stop playback right away and future reconnects will not be possible.

After a attach is done to the new NetConnection, a close on the older NetConnection can be called.

Reconnecting a playlist

The following workflow reconnects a playlist:

1 Call `NetConnection.connect()` to connect to server A.

2 Create a NetStream. Set `NetStream.bufferTime` to at least a few seconds so there is enough data to play while the connection is down.

3 Call `NetStream.play2()`and use `NetStreamPlayTransitions.RESET` to play a stream called "myStream1".

4 Call `NetStream.play2()` and use `NetStreamPlayTransitions.APPEND` to play a second stream called "myStream2".

5 Monitor the NetConnection for the `"NetConnection.Connect.Closed"` event. Reconnect to server A if the connection drops.

6 Call the `NetStream.attach(connection:NetConnection)` method to attach the NetStream to the new connection.

7 Call `NetStream.play2()` and use `NetStreamPlayTransitions.APPEND_AND_WAIT` to add "myStream1" to the playlist.

8 Call `NetStream.play2()` and use `NetStreamPlayTransitions.RESUME` to add "myStream2" to the playlist and resume the stream.

Flash Player tells the server where to resume playing the stream.

Balancing a server load

The following workflow uses the smart reconnect API to balance a load:

1 Call `NetStream.attach(connection:NetConnection)` to attach a stream to a NetConnection object on another server.

2 After the stream is successfully attached to the new connection, call `NetConnection.close()` on the prior connection to prevent data leaks.

3 Call `NetStream.play2()` and use `NetStreamPlayOptions.transition` to `RESUME`. Set the rest of the NetStreamPlayOptions properties to the same values you used when you originally called `NetStream.play()` or `NetStream.play2()` to start the stream.

Monitoring a network interface change

To monitor changes to the network interface, monitor the NetConnection for the `"NetConnection.Connect.NetworkChange"` event. This event detects a network change (for example, connecting to a wireless network, disconnecting from a wireless network, disconnecting a network cable, and so on).

Note: *Do not use this event to monitor a dropped NetConnection or to write logic for to reconnect a stream. Use* `"NetConnection.Connect.Closed"`.

Monitoring a connection on a mobile device

Some mobile devices cannot receive a `"NetConnection.Connect.Closed"` message. In this case, you can monitor the `NetStream.bufferLength` and `NetStreamInfo.byteCount` properties in a timer to discover network issues. When `NetStream.bufferLength` is less than `NetStream.bufferTime`, and `NetStreamInfo.byteCount` is not increasing, there are probably network issues.

```
netStreamMonitorTimer.start();
netStreamMonitorTimer.addEventListener(TimerEvent.TIMER, timerHandler);
lastByteCount = 0;
private function timerHandler(e:TimerEvent):void{
    if(netstream.bufferLength < netstream.bufferTime && netstream.info.byteCount ==
lastByteCount) {
        // Network has issues.. reconnect to a new NetConnection
        netconnection2 = new NetConnection(); ///on NetConnection.Connect.Success:
        netStream.attach(netconnection2);
    }
    lastByteCount = netstream.info.byteCount;
}
```

Monitoring an RTMPS or RTMPT connection

For RTMPS and RTMPT connections, when you call `NetStream.attach()`, the `"NetConnection.Connect.Closed"` event may arrive late or may never arrive.

In this case, monitor the `NetStream.bufferLength` and `NetStreamInfo.byteCount` properties in a timer to discover network issues. When `NetStream.bufferLength` is less than `NetStream.bufferTime`, and `NetStreamInfo.byteCount` is not increasing, there are probably network issues.

```
netStreamMonitorTimer.start();
netStreamMonitorTimer.addEventListener(TimerEvent.TIMER, timerHandler);
lastByteCount = 0;
private function timerHandler(e:TimerEvent):void{
    if(netstream.bufferLength < netstream.bufferTime && netstream.info.byteCount ==
lastByteCount) {
        // Network has issues.. reconnect to a new NetConnection
        netconnection2 = new NetConnection(); ///on NetConnection.Connect.Success:
        netStream.attach(netconnection2);
    }
    lastByteCount = netstream.info.byteCount;
}
```

Authorization plug-in events and properties

Use the `E_PLAY` event of the Authorization plug-in to control streaming that occurs after a reconnection. The following table summarizes the events and properties for the Stream Reconnect feature:

Property	**Server version**	`E_PLAY` **Notification and Authorization**
`F_STREAM_OFFSET`	3.5.3	Read-only.
`F_STREAM_TRANSITION`	3.5	Read and write.

There is one new property: `F_STREAM_OFFSET`. The `F_STREAM_OFFSET` property indicates where to resume streaming after a reconnection, in seconds.

The `F_STREAM_TRANSITION` property indicates the transition mode sent by the client in the `NetStream.play2()` call. The values for Stream Reconnect are "resume" and "appendAndWait".

For more information about

More Help topics

"Developing an Authorization plug-in" on page 303

Server logging

The following events are written to the Adobe Media Server access log for the Stream Reconnect feature:

Event	**Category**	**Description**
connect	session	A client has connected to a Adobe Media Server application. This event is logged when you re-establish a connection after it drops.
play	stream	A stream has resumed playing.
stop	stream	A stream has stopped playing.

The following fields are new for the Stream Reconnect feature:

Field	**Description**
x-trans-mode	The transition mode sent by the client in the `NetStream.play2()` call. For Stream Reconnect, the transition modes are "resume" and "appendAndWait".
x-soffset	The offset value indicates where to resume streaming after you attach a NetStream.

These fields are disabled by default. You can optionally display these fields in the authEvent.log and in the access.log.

Fast switching between streams

Adobe Media Server 4.0, Flash Player 10.1

About fast switch

Use the `NetStreamPlayOptions.offset` property to tell the server when to switch between streams of different bitrates. The `offset` is the time, in seconds, at which the streams switch.

The default value of `offset` is -1, which is the fast switching mode. In this mode, switching occurs at the first available keyframe after `netstream.time + 3`, which is about 3 seconds later than the playback point.

Important: *Best practice for fast switching is to set* `NetStream.bufferLength` *to 10 seconds and* `NetStreamPlayOptions.offset` *to 3, 4, or 5 seconds.*

Any data buffered from a previous stream is flushed. Fast switch is faster than standard switch mode because clients don't have to wait for buffered data to play.

To use standard switch mode, set `offset` to a value greater than the buffer (`offset > NetStream.time + NetStream.bufferLength`).

Writing the fast switching code

Use the `NetStreamPlayTransitions.SWITCH` constant to trigger stream switches. Use the `NetStreamPlayOptions.offset` property to specify the absolute stream time at which the switch occurs. The `offset` property is absolute stream time, it is not an offset from the playback point. For example, to switch 5 seconds from the playback point, set the `offset` property to `netstream.time + 5`, not to `5`.

```
var ns:NetStream = new NetStream(nc);
var nso:NetStreamPlayOptions = new NetStreamPlayOptions();
nso.streamName = streamName;
nso.transition = NetStreamPlayTransitions.SWITCH;
nso.start = 10;
nso.len = -1; // play until the end of the file
nso.offset = ns.time + 5; // switch 5 secs. after the playback point.
ns.play2(nso);
```

Note: *The default value of* `offset` *is* `netStream.time + 3`*. The default value causes the server to switch the stream 3 seconds after the playback point.*

NetStatus events for fast switching

NetStatus Event	Description
`NetStream.Play.Failed`	The value of `offset` is less than the current playhead time. The stream cannot switch.
`NetStream.Play.Transition`	A `SWITCH` call was made successfully and the server starts streaming data for the new stream.

NetStatus Event	Description
`NetStream.Play.TransitionComplete`	A new stream starts playing. In fast switching mode, the time between the `NetStream.Play.Transition` event and the `NetStream.Play.TransitionComplete` event is much shorter than in standard switching mode.

Smart Seeking

Adobe Media Server 3.5.3, Flash Player 10.1

About Smart Seek

Adobe Media Server 3.5.3 and Flash Player 10.1 work together to support smart seeking in VOD streams and in live streams that have a buffer. Smart seeking uses back and forward buffers to seek without requesting data from the server. You can step forward and backward a specified number of frames. (Standard seeking flushes buffered data and asks the server to send new data based on the seek time.) Smart seeking reduces server load and improves seeking performance. Use smart seeking to create:

- Client-side DVR functionality. Seek a live stream within the client-side buffer instead of going to the server for delivery of new video.
- Trick modes. Create players that step through frames, fast-forward, fast-rewind, and advance in slow-motion.

For an example of smart seek, see the Stream Reconnect and Smart Seek example hosted by Adobe.

Note: *Smart seeking is not supported in peer-assisted networking applications or with progressive download.*

Smart Seek ActionScript API

Note: *The Smart Seek ActionScript API is ActionScript 3.0. These APIs are not supported in ActionScript 2.0.*

Flash Player maintains a back buffer and a forward buffer. The back buffer is a cache of data that has been displayed. The forward buffer is a cache of data that hasn't been displayed. Smart seeking retrieves data from within these buffers.

To turn on Smart Seek set `NetStream.inBufferSeek` to `true`.

To control the buffers, use the following APIs:

- `NetStream.backBufferLength`

  [read-only] The number of seconds of previously displayed data cached for rewinding and playback. This is the `bufferLength` property for the back buffer.

- `NetStream.backBufferTime`

  Specifies how much previously displayed data is cached for rewinding and playback, in seconds. The default value is 30 on the desktop and 3 on mobile.

- `NetStream.bufferLength`

  [read-only] The number of seconds of data currently in the buffer.

- `NetStream.bufferTime`

  Specifies how long to buffer messages before starting to display the stream, in seconds. The default value is 0.1.

To seek and step within the buffers, use the following APIs:

- `NetStream.seek()`

  Moves the playhead to the time specified in the call.

- `NetStream.step()`

  Steps the playhead forward or back the specified number of frames, relative to the currently displayed frame.

To detect a smart seek, use the following events:

- `NetStatusEvent.info.description` contains the string `"client-inBufferSeek"`.

  Dispatched when a call to `NetStream.seek()` is successful.

- `NetStream.Step.Notify`

  Dispatched when a call to `NetStream.step()` is successful.

For detailed information about these APIs, see the ActionScript 3.0 Language Reference.

Using the Smart Seek ActionScript API

Smart Seek is supported in Flash Player 10.1 and greater. Before running code, test for Flash Player version. For example, you could expose step back and forward buttons only to clients with Flash Player 10.1 and greater. The following code tests for Flash Player version and returns `true` for 10.1 and greater:

```
public var fp10_1:Boolean;
public function onStart():void{
    debug("Flash Player Version: " + Capabilities.version);
    fp10_1 = isFP10_1();
    debug("fp10.1: "+fp10_1);
}
public function isFP10_1():Boolean {
    var va:Array = Capabilities.version.split(" ")[1].toString().split(",");
    if(int(va[0]) > 10) { return true; }
    if(int(va[0]) < 10) { return false; }
    if(int(va[1]) > 1) { return true; }
    if(int(va[1]) < 1) { return false; }
    return true;
}
```

The following code sets `NetStream.inBufferSeek` to `true` to turn on Smart Seek if the Flash Player version is greater than 10.1:

```
// Call this function when you catch NetConnection.Connect.Success
public function createNetStream():void{

// Write code to create a NetStream object and a Video object...
//...
// Set the forward buffer, in seconds.
ns.bufferTime = 10;
try {
    if(fp10_1) {
// If Flash Player is greater than 10.1, turn on smart seeking
// and set the size of the back buffer, in seconds.
        ns.inBufferSeek = true;
        ns.backBufferTime = 30;
    }
} catch(e:Error) {}
```

When Smart Seek is turned on, calls to `NetStream.seek()` use the buffer. (Standard seeking flushes the buffer and sends a request for data to the server.) The following function can seek forward or backward to a number specified in a text field called `seekText`:

```
public function seekHandler():void {
    if(ns != null) {
        ns.seek(Number(seekText.text));
    }
}
```

To create players that step through frames, fast-forward, fast-rewind, and advance in slow-motion, adjust the number you pass to the `step()` and `seek()` functions.

To create client-side DVR functionality, set the `NetStream.backBufferTime` and `NetStream.bufferTime` properties. These properties specify the amount of data Flash Player stores in the client-side buffer. For example, to allow users to rewind 30 minutes before live, set `backBufferTime` to 1800 (60 seconds x 30 minutes). The cache is stored in memory. Set the buffer properties to lower values if the content is intended for netbooks or mobile devices.

Testing for a Smart Seek

A Smart Seek is a call to `NetStream.seek()` or a call to `NetStream.step()` when `NetStream.inBufferSeek` is `true`.

When a call to `NetStream.seek()` is successful, the NetStatusEvent `info.description` property contains the string `"client-inBufferSeek"`.

When a call to `NetStream.step()` is successful, the NetStatusEvent `info.code` property contains the string `"NetStream.Step.Notify"`. If a step has not completed, another call to step may return without executing. Get the `"NetStream.Step.Notify"` for the previous call before you call step again.

The following code tests for a call to `NetStream.seek()` and a call to `NetStream.step()`:

```
private function netStatusHandler(event:NetStatusEvent):void {
    switch (event.info.code) {
        case "NetStream.Seek.Notify":
            var desc:String = new String(event.info.description);
            if(desc.indexOf("client-inBufferSeek") >= 0)
                trace("A smart seek occured");
            else
                trace("A standard seek occured");
            break;
        case "NetStream.Step.Notify":
            trace("Successful NetStream.step() call");
            break;
    }
}
```

Smart Seek requires the following:

- `NetStream.inBufferSeek = true`

  The default value of `inBufferSeek` is `false`.
- Adobe Media Server 3.5.3
- Flash Player 10.1
- ActionScript 3.0
- The value of the buffers (`backBufferLength` and `bufferLength`) must be large enough to fulfill the seek request.

If any of the previous requirements are not met, Flash Player uses standard seeking but throws no compile-time or runtime errors.

Authorization plug-in events and properties

Use Authorization plug-in events and properties to log information about smart seeking and to block clients from sending Smart Seek commands to the server.

More Help topics

"Smart seeking" on page 309

Server logging

The following events are written to the Adobe Media Server access.log:

Event	Category	Description
client-seek	stream	The seek position when the client calls `NetStream.seek()` and `NetStream.inBufferSeek = true` which sends a `seekRaw` command to the server. The client sends a seekRaw command only for seeks inside the buffer. If a client seeks outside the buffer, the client sends a "seek" event.
start-transmit	stream	The server received a `startTransmit` command. This command asks the server to transmit more data because the buffer is running low.
stop-transmit	stream	The server received a `stopTransmit` command. This command asks the server to suspend transmission until the client sends a `startTransmit` event because there is enough data in the buffer.

Note: *If a user interface lets users step through frames, there could be thousands of calls to `NetStream.step()`. For performance reasons, the server does not write these calls to the log file. These calls do trigger a client-side `"NetStream.Step.Notify"` NetStatusEvent.*

Detecting bandwidth

ActionScript 3.0 native bandwidth detection

Adobe Media Server 3.0 and later support native bandwidth detection from the server to the client. After the client connects to the server, call `NetConnection.call("checkBandwidth", null)` to intiate bandwidth detection. The server sends chunks of data to the client and waits for a return value from the client. You do not need to write any server-side code.

Note: *This example is based on the rootinstall/documentation/samples/bandwidthcheck/Bandwidth.as sample.*

Enable bandwidth detection in the Application.xml file

❖ Verify that bandwidth detection is enabled in the Application.xml file:

```
<BandwidthDetection enabled="true">
    <MaxRate>-1</MaxRate>
    <DataSize>16384</DataSize>
    <MaxWait>2</MaxWait>
</BandwidthDetection>
```

Bandwidth detection is enabled by default. In addition to enabling and disabling bandwidth detection, you can configure the size of the data chunks the server sends to the client, the rate at which the data is sent, and the amount of time the server waits between data chunks.

You can edit the Application.xml file at the application level or at the vhost level. See Configuring a single application.

Write the client event handler class

❖ Create an ActionScript 3.0 class that handles events and calls bandwidth detection on the server. It must implement the `onBWCheck` and `onBWDone` functions:

```
class Client {
    public function onBWCheck(... rest):Number {
        return 0;
    }
    public function onBWDone(... rest):void {
        var bandwidthTotal:Number;
        if (rest.length > 0){
            bandwidthTotal = rest[0];
            // This code runs
            // when the bandwidth check is complete.
            trace("bandwidth = " + bandwidthTotal + " Kbps.");
        }
    }
}
```

The `onBWCheck()` function is required by native bandwidth detection. It takes an argument, `...rest`. The function must return a value, even if the value is 0, to indicate to the server that the client has received the data. You can call `onBWCheck()` multiple times.

The server calls the `onBWDone()` function when it finishes measuring the bandwidth. It takes four arguments. The first argument is the bandwidth measured in Kbps. The second and third arguments are not used. The fourth argument is the latency in milliseconds.

Write the main class

1 Create a main ActionScript 3.0 class, giving it a package and class name of your choice:

```
package {
    import flash.display.Sprite;
    import flash.net.NetConnection;
    import flash.events.NetStatusEvent;
    import flash.events.AsyncErrorEvent;

    public class Bandwidth extends Sprite
    {
    }
}
```

You can create the main and client classes in the same file, like the Bandwidth.as example file.

2 In the constructor of the main class, create a NetConnection object, set the `NetConnection.client` property to an instance of the client class, and connect to the server:

```
private var nc:NetConnection;

public function Bandwidth()
{
    nc = new NetConnection();
    nc.addEventListener(NetStatusEvent.NET_STATUS, netStatusHandler);
    nc.addEventListener(AsyncErrorEvent.ASYNC_ERROR, asyncErrorHandler);
    nc.client = new Client();
    nc.connect("rtmp://localhost/bandwidthcheck");
}
```

3 In the `netStatus` event handler, call `NetConnection.call()` if the connection is successful, passing `checkBandwidth` as the command to execute and `null` for the response object:

```
public function netStatusHandler(event:NetStatusEvent):void
{
    trace(event.info.code);
    switch (event.info.code)
    {
        case "NetConnection.Connect.Success":
            // Calls native bandwidth detection code on the server.
            // You don't need to write any server-side code.
            nc.call("checkBandwidth", null);
            break;
    }
}
```

***Note:** The `checkBandwidth()` method belongs to the Client class on the server.*

Run the sample

❖ Test the main class from Flash or Flash Builder. The following is the Flash Builder output showing the client's bandwidth:

```
[SWF] C:\samples\Bandwidth\bin\Bandwidth-debug.swf - 2,137 bytes after decompression
The connection was made successfully
Bandwidth from server to client is 17287 Kpbs
```

In this example, the Client class displays the bandwidth value. In your client, you can take some action, such as choosing a video to stream to the client based on the client's bandwidth.

ActionScript 2.0 native bandwidth detection

You can also use native bandwidth detection from ActionScript 2.0. Just as in ActionScript 3.0, define functions named `onBWCheck()` and `onBWDone()`. When the client connects to the server , call to `NetConnection.call("checkBandwidth")`.

***Note:** This example uses the rootinstall/documentation/samples/bandwidthcheck/BandwidthAS2.fla sample.*

1 On the server, create a *rootinstall*/applications/bandwidthcheck folder.

2 Do one of the following to verify that native bandwidth detection is enabled:

- Open the *rootinstall*/conf/_defaultRoot_/_defaultVHost_/Application.xml file and verify that `<BandwidthDetection enabled="true">`.

- Create an Application.xml with the following code and copy it to the *rootinstall*/applications/bandwidthcheck folder:

```
<Application>
    <Client>
        <BandwidthDetection enabled="true">
        </BandwidthDetection>
    </Client>
</Application>
```

3 In Flash Professional, choose File > New > ActionScript 2.0 to create a new ActionScript 2.0 file.

4 Open the Actions panel and paste the following code into frame 1:

```
nc = new NetConnection();
nc.onStatus = function(info){
    trace(info.code);
    if(info.code == "NetConnection.Connect.Success"){
        checkBandwidth();
    }
}
nc.onBWCheck = function(dataChunk){
    return 0;
}
nc.onBWDone = function(bandwidth){
    trace("Bandwidth from server to client is: " + bandwidth + " Kbps");
}
function checkBandwidth(){
    nc.call("checkBandwidth");
}
nc.connect("rtmp://localhost/bandwidthcheck");
```

5 Choose Control > Test Movie. The onStatus messages and the bandwidth measurement are traced to the Output panel.

Initiate native bandwidth detection from a server-side script

You can initiate native bandwidth detection from a server-side script. In this case, the server-side code calls the native `checkBandwidth()` function on the server:

```
application.onConnect = function (clientObj){
    this.acceptConnection(clientObj);
    clientObj.checkBandwidth();
}
```

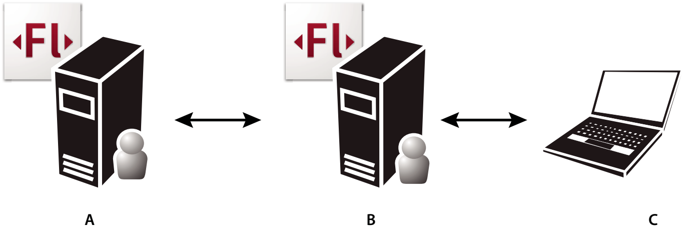
When initiating bandwidth detection from the server, do not call `checkBandwidth()` from the client.

The main.asc server-side script that Adobe provides with the vod application calls the `checkBandwidth` function. To see an example, open *rootinstall*/samples/applications/vod/main.asc.

Server-side script-based bandwidth detection

You can disable native bandwidth detection and use a server-side script that defines the `checkBandwidth` function. For example, if you have a legacy application that uses a server-side script to perform bandwidth detection, disable native bandwidth detection. You can disable native bandwidth detection at the application level or at the vhost level.

Script-based bandwidth detection is not as reliable as native bandwidth detection. If you use edge servers, native bandwidth detection is performed at the outermost edge server to reduce the load on the origin servers. Script-based bandwidth detection determines the bandwidth from the origin server to the client, not from the edge server to the client. If latency exists between the origin server and the edge server, it might affect the bandwidth calculation.



Latency between the origin server and the edge server can affect the bandwidth measurement.
***A.** Origin server **B.** Edge server **C.** Client*

To disable native bandwidth detection at the application level:

1 Open a text editor, create a new file, and save it as Application.xml to the *rootinstall*/applications/*applicationname* folder.

2 Copy the following XML into the file:

```
<Application>
    <Client>
        <BandwidthDetection enabled="false">
        </BandwidthDetection>
    </Client>
</Application>
```

3 Save the file. You do not need to restart the server.

To disable native bandwidth detection at the vhost level:

1 Open the *rootinstall*\conf_defaultRoot__defaultVHost_\Application.xml file in a text editor.

Note: *This path is for the default vhost. You can edit the Application.xml file for any vhost.*

2 Locate the BandwidthDetection tag and set the enabled attribute to false, as in the following:

```
<Application>
    ...
    <Client>
        ...
        <BandwidthDetection enabled="false">
        </BandwidthDetection>
        ...
    </Client>
    ...
</Application>
```

3 Save the file. Restart the server or vhost.

Detecting stream length

About detecting stream length

Call the server-side `Stream.length()` method to get the length, in seconds, of an audio or video stream. The length is measured by Adobe Media Server and differs from the duration that `onMetaData` returns, which is set by a user or a tool.

Pass the stream name to the `Stream.length()` method. You can pass a virtual stream name or a stream name in a URI relative to the application instance.

For example, the following code gets the length of a stream located in the streams/_definst_ folder of an application:

```
// for an FLV file
length = Stream.length("parade");
// for an MP3 file
length = Stream.length("mp3:parade.mp3");
// for an MP4 file
length = Stream.length("mp4:parade.mp4");
```

Get the length of a stream

This example uses Server-Side ActionScript to get the length of a stream.

***Note:** Use the StreamLength sample, main.asc (Server-Side ActionScript) and StreamLength.as (ActionScript 3.0). To run the sample, see "Deploy an application" on page 185.*

Write the server-side code

A client might need to retrieve the length of a stream stored on the server, for example, if a Flash presentation displays the length of a video to let the user decide whether to play it.

To do this, define a method in server-side code that calls `Stream.length()`, and then have the client call it using `NetConnection.call()`.

❖ In main.asc, define a function on the `client` object that calls `Stream.length()`. Do this within the `onConnect` handler:

```
application.onConnect = function( client ) {
    client.getStreamLength = function( streamName ) {
        trace("length is " + Stream.length( streamName ));
        return Stream.length( streamName );
    }
    application.acceptConnection( client );
}
```

Write the main client class

From the main client class, you call `getStreamLength()` in the server-side code. You need to create a Responder object to hold the response:

```
var responder:Responder = new Responder(onResult);
```

This line specifies that the `onResult()` function will handle the result. You also need to write `onResult()`, as shown in the following steps.

1 In your client code, create a package, import classes, and define variables as usual:

```
package {
    import flash.display.Sprite;
    import flash.net.NetConnection;
    import flash.events.NetStatusEvent;

    import flash.net.NetStream;
    import flash.net.Responder;
    import flash.media.Video;
    ...
```

2 Create a new class, StreamLength:

```
    public class StreamLength extends Sprite
    {
        var nc:NetConnection;
        var stream:NetStream;
        var video:Video;
        var responder:Responder;
    }
        ...
```

3 In the constructor for the StreamLength class, call `NetConnection.connect()` to connect to the server:

```
public function StreamLength()
{
    nc = new NetConnection();
    nc.addEventListener(NetStatusEvent.NET_STATUS, netStatusHandler);
    nc.connect("rtmp://localhost/StreamLength");
    }
```

4 Add a `netStatus` event handler to handle a successful connection, rejected connection, and failed connection:

```
private function netStatusHandler(event:NetStatusEvent):void
{
    trace("connected is: " + nc.connected );
    trace("event.info.level: " + event.info.level);
    trace("event.info.code: " + event.info.code);

    switch (event.info.code)
    {
        case "NetConnection.Connect.Success":
        trace("Congratulations! you're connected");
        connectStream(nc);
        break;

        case "NetConnection.Connect.Rejected":
        case "NetConnection.Connect.Failed":
        trace ("Oops! the connection was rejected");
        break;
     }
    }
```

5 Write a function to play the stream when a successful connection is made. In it, create a Responder object that handles its response in a function named `onResult()`. Then call `NetConnection.call()`, specifying `getStreamLength` as the function to call on the server, the Responder object, and the name of the stream:

```
// play a recorded stream on the server
private function connectStream(nc:NetConnection):void {
    stream = new NetStream(nc);
    stream.addEventListener(NetStatusEvent.NET_STATUS, netStatusHandler);
    stream.client = new CustomClient();

    responder = new Responder(onResult);
    nc.call("getStreamLength", responder, "bikes" );

}
```

6 Write the `onResult()` function to handle the stream length returned by `getStreamLength()` on the server:

```
private function onResult(result:Object):void {
                trace("The stream length is " + result + " seconds");
            output.text = "The stream length is " + result + " seconds";
    }
```

Write the client event handler class

❖ Write a separate class to handle the `onMetaData` and `onPlayStatus` events:

```
class CustomClient {
    public function onMetaData(info:Object):void {
        trace("metadata: duration=" + info.duration + " width=" + info.width +
        " height=" + info.height + " framerate=" + info.framerate);
    }
    public function onPlayStatus(info:Object):void {
        trace("handling playstatus here");
    }
}
```

Chapter 5: Working with live video

Adobe Media Server can broadcast live audio and video content to Flash Player, AIR, and Flash Lite clients. To capture and encode live content and stream it to Adobe Media Server, you can use Flash Media Live Encoder or build a custom Flash Player or AIR application.

You can capture live events in real time and stream them to large audiences or create social media applications that include live audio and video. For example, Adobe® Acrobat® Connect™ Pro is a web conferencing application that uses Adobe Media Server to capture and broadcast live audio and video.

Capturing live video

Using Flash Media Live Encoder to capture video

Flash Media Live Encoder is a free application that captures live video, encodes it, and streams it to Adobe Media Server. By default, Flash Media Live Encoder is configured to stream video to the live service at rtmp://localhost/live.

Adobe Media Server installs with a sample video player that can play streams from the live service. The video player is installed to the folder *rootinstall*\samples\videoPlayer.

The sample video player is based on Strobe Media Playback built on Open Source Media Framework. For more information, see osmf.org.

For more information about Flash Media Live Encoder, see www.adobe.com/go/fme.

More Help topics

"Stream live media (RTMP)" on page 17

Example: Custom video capture application

The following steps build an application in ActionScript 3.0 that:

- Captures and encodes video.
- Displays the video as it's captured.
- Streams video from the client to Adobe Media Server.
- Streams video from Adobe Media Server back to the client.
- Displays the video streamed from the server.

Note: *To test this code, create a rootinstall/applications/publishlive folder on the server. Open the rootinstall/documentation/samples/publishlive/PublishLive.swf file to connect to the application.*

1. In a new .as file, create a NetConnection object. To connect to the server, pass the URI of the application to the `NetConnection.connect()` method.

```
var nc:NetConnection;
var ns:NetStream;
var nsPlayer:NetStream;
var vid:Video;
var vidPlayer:Video;
var cam:Camera;
var mic:Microphone;

nc = new NetConnection();
nc.addEventListener(NetStatusEvent.NET_STATUS, onNetStatus);
nc.connect("rtmp://localhost/publishlive");
```

2. Define a function to handle `NetStatusEvent` events. If the client makes a successful connect to the server, the code calls functions that run the application.

```
function onNetStatus(event:NetStatusEvent):void{
    trace(event.info.code);
    if(event.info.code == "NetConnection.Connect.Success"){
        publishCamera();
        displayPublishingVideo();
        displayPlaybackVideo();
    }
}
```

3. Publish the video being captured by the camera and the audio being captured by the microphone. First, get references to the camera and microphone data. Create a NetStream object on the NetConnection you made to the server. Then call `NetStream.attachCamera()` to attach the captured video to the NetStream object. Call `NetStream.attachAudio()` to attach the live audio. Finally, call `NetStream.publish("`*`streamName`*`", "live")`to send the audio and video to the server.

```
function publishCamera() {
    cam = Camera.getCamera();
    mic = Microphone.getMicrophone();
    ns = new NetStream(nc);
    ns.attachCamera(cam);
    ns.attachAudio(mic);
    ns.publish("myCamera", "live");
}
```

4. Display the video the client is streaming to the server. Create a Video object. The Video object is a display object. Call `Video.attachCamera(cam)` to attach the camera video feed to the video object. Call `addChild(vid)` to add the Video object to the display list so that it appears in Flash Player.

```
function displayPublishingVideo():void {
    vid = new Video();
    vid.x = 10;
    vid.y = 10;
    vid.attachCamera(cam);
    addChild(vid);
}
```

5. Display the video the server is streaming to the client. A client that creates a stream to send to a server is *publishing*, while a client playing a stream is *subscribing*. The client in this example publishes a stream and subscribes to a stream. The client must create two streams, an outgoing (publishing) stream and an incoming (subscribing) stream. In this example, the outgoing stream is `ns` and the incoming stream is `nsPlayer`. You can use the same NetConnection object for both streams.

To play the audio and video as it returns from the server, call `NetStream.play()` and pass it the name of the stream you published. To display the incoming video, call `Video.attachNetStream()`:

```
function displayPlaybackVideo():void{
    nsPlayer = new NetStream(nc);
    nsPlayer.play("myCamera");
    vidPlayer = new Video();
    vidPlayer.x = cam.width + 20;
    vidPlayer.y = 10;
    vidPlayer.attachNetStream(nsPlayer);
    addChild(vidPlayer);
}
```

Use ActionScript to allow users to control playback, display video fullscreen, use cuepoints, and use metadata. For more information, see the "Working with video" chapter in ActionScript 3.0 Developer's Guide.

Adding DVR features to live video

About DVR support

Flash Media Server 3.5 or higehr

A DVR (digital video recorder) lets viewers pause live video and resumes playback from the paused location. Viewers can also rewind a live event, play the recorded section, and seek to the live section again. Just write a few lines of code and a video player can act like a DVR. Examples of DVR applications are instant replay and "catch-up" services.

Note: *The FLVPlayback component version 2.5 supports DVR. Download the component from the Adobe Media Server Tools*

Using DVR with dynamic streaming

Flash Media Server 4.0 or higher, Flash Player 10.1

To use DVR with a dynamic streaming application, enable absolute time code on the server. Absolute time code keeps DVR-enabled, multi-bitrate streams synchronized at the server.

The encoder must use absolute time code. Flash Media Live Encoder supports absolute time code, Flash Player does not. Third-party encoders also use absolute time code. Contact your encoder vendor to learn how to enable absoulte time code for your encoder.

Important: *The streams must be in sync, the server does not sync the streams. Absolute time code ensures that the server captures all the information required to keep encoder-synchronized streams in sync.*

Enable absolute time code

Enable absolute time code in the Application.xml configuration file at the application level. When enabled, the server assumes that incoming live streams have timestamps that are based on an absolute clock, such as a SMPTE time signal contained within the encoder's input source. The default value is false.

1 Create a new text or XML file and save it to the application's folder as "Application.xml".

   Note: *For an example of an application-level Application.xml file, open rootinstall/applications/vod/Application.xml in a text editor.*

2 Add the following to the XML file:

```
<Application>
    <StreamManager>
        <Live>
            <AssumeAbsoluteTime>true</AssumeAbsoluteTime>
        </Live>
    </StreamManager>
</Application
```

3 Save the file.

For more information about working with configuration files, see Configuring a single application.

Publish and record streams for DVR with dynamic streaming

To publish and record DVR streams with dynamic streaming, pass `"appendWithGap"` to the `NetStream.publish()` or `Stream.record()` methods. When you publish or record a stream in `"appendWithGap"` mode, the recorded stream preserves gaps created when a stream stops and restarts. In `"append"` mode, the server eliminates the gaps which can cause streams to lose sync. Gaps can occur when an encoder goes offline and comes back online. Gaps are visible to the client when a recorded file is played.

Publishing, playing, and seeking DVR video

To publish a stream for a DVR video player from a client, use `"record"` or `"append"` flags, as in the following:

```
NetStream.publish("myvideo", "record")
NetStream.publish("myvideo", "append")
```

To publish a stream for a DVR video player from the server, call `Stream.record()`. The `Stream.record()` method has two new parameters, `maxDuration` and `maxSize`, that let you specify the maximum length and file size of a stream. The following code publishes a stream with a maximum recording length of 10 mins (600 seconds) and an unlimited file size:

```
Stream.record("record", 600, -1)
```

When a client plays a stream published for a DVR video player, they don't play a live stream, they play a recorded stream. When a client views the stream "live", they're really viewing the recorded stream just after it was recorded.

To subscribe to a stream published for a DVR video player, use the following code:

```
NetStream.play("myvideo", 0, -1)
```

The previous code allows viewers joining an event late to view from the beginning.

To return to the beginning of a stream at any time, call the following:

```
NetStream.seek(0)
```

To start recording in the middle of an event, call the Server-Side ActionScript `Stream.record()` method. Calling this method lets you start and stop recording at any time.

To create a button that seeks to the latest available part of the recording (which is considered "live"), seek to the duration of the stream (the length of the data recorded on the server) minus the value of `NetStream.bufferTime`. Subtract `bufferTime` to make playback as close to instanteous as possible. Flash Player doesn't resume playback until the buffer is full.

To calculate a stream's duration, set a time stamp in the `onMetaData` callback function.

```
public function onMetaData(info:Object):void{
    trace("metadata:duration=" + info.duration);
    duration = info.duration;
    trace("stamp: " + stamp);
    stamp = getTimer();
}
```

To calculate the value to pass to the `seek()` method, calculate the current duration, and subtract the `bufferTime` minus an additional 2 seconds for safety.

```
private function getSeekToLiveValue():uint{
    currentDuration = Number((getTimer()-stamp)/1000) + duration;
    trace("currentDuration: " + currentDuration);
    seekVal = (currentDuration - nsPlayer.bufferLength) - 2;
    return seekVal;
}
```

To seek to "live", call the `getSeekToLiveValue()` function on the playback NetStream object:

```
private function onClick(event:MouseEvent):void {
    switch(event.currentTarget){
        case rewindBtn:
            nsPlayer.seek(nsPlayer.time - 5);
            break;
        case seekBtn:
            trace("seekToEndValue " + getSeekToEndValue());
            nsPlayer.seek(getSeekToLiveValue());
            break;
    }
}
```

***Note:** The previous code is in the DVR sample in the rootinstall/documentation/samples/dvr folder on the server.*

Using Flash Media Live Encoder to capture video for DVR playback

You can use Flash Media Live Encoder 3 to capture video for DVR playback. Earlier versions of Flash Media Live Encoder do not support recording to the server. For more information, see http://www.adobe.com/go/fme.

Example: Custom capture, publish, and DVR playback

This example is a client application that does the following:

- Captures and encodes video.
- Displays the video as it's captured.
- Streams video from the client to Adobe Media Server.
- Streams video from Adobe Media Server back to the client.
- Displays the video streamed from the server in a player that lets you rewind and pause live video.

***Note:** To test this code, create a RootInstall/applications/dvr folder on the server. Open the RootInstall/documentation/samples/dvr/DVR.swf file to connect to the application.*

1 On Adobe Media Server, create a *RootInstall*/applications/dvr folder.

2 In Flash, create an ActionScript file and save it as DVR.as.

3 Copy and paste the following code into the Script window:

```
package {
    import flash.display.MovieClip;
    import flash.utils.getTimer;
    import flash.net.NetConnection;
    import flash.events.*;
    import flash.net.NetStream;
    import flash.media.Video;
    import flash.media.Camera;
    import flash.media.Microphone;
    import fl.controls.Button;
    public class DVR extends MovieClip
    {
        private var nc:NetConnection;
        private var ns:NetStream;
        private var nsPlayer:NetStream;
        private var vid:Video;
        private var vidPlayer:Video;
        private var cam:Camera;
        private var mic:Microphone;
        private var pauseBtn:Button;
        private var rewindBtn:Button;
        private var playBtn:Button;
        private var seekBtn:Button;
        private var dvrFlag:Boolean;
        private var stamp:uint;
        private var duration:uint;
        private var currentDuration:uint;
        private var seekVal;uint;
        public function DVR()
        {
            nc = new NetConnection();
            nc.addEventListener(NetStatusEvent.NET_STATUS, onNetStatus);
            nc.connect("rtmp://localhost/dvr");
            setupButtons();
            dvrFlag = true;
        }
        private function onNetStatus(event:NetStatusEvent):void{
            trace(event.info.code);
            switch(event.info.code){
                case "NetConnection.Connect.Success":
                    publishCamera();
                    displayPublishingVideo();
                    displayPlaybackVideo();
                    break;
                case "NetStream.Play.Start":
                    trace("dvrFlag " + dvrFlag);
                    if(dvrFlag){
                        nsPlayer.seek(getSeekToLiveValue());
                        dvrFlag = false;
                    }
                    break;
            }
        }
        private function onAsyncError(event:AsyncErrorEvent):void{
                trace(event.text);
        }
        private function onClick(event:MouseEvent):void {
            switch(event.currentTarget){
                case rewindBtn:
                    nsPlayer.seek(nsPlayer.time - 5);
                    break;
                case pauseBtn:
                    nsPlayer.pause();
                    break;
                case playBtn:
                    nsPlayer.resume();
                    break;
                case seekBtn:
                    nsPlayer.seek(getSeekToLiveValue());
                    break;
            }
        }
        private function publishCamera() {
            cam = Camera.getCamera();
            mic = Microphone.getMicrophone();
            ns = new NetStream(nc);
            ns.client = this;
            ns.addEventListener(NetStatusEvent.NET_STATUS, onNetStatus);
            ns.addEventListener(AsyncErrorEvent.ASYNC_ERROR, onAsyncError);
            ns.attachCamera(cam);
            ns.attachAudio(mic);
            ns.publish("video", "record");
        }
        private function displayPublishingVideo():void {
            vid = new Video(cam.width, cam.height);
            vid.x = 10;
            vid.y = 10;
            vid.attachCamera(cam);
            addChild(vid);
        }
        private function displayPlaybackVideo():void{
            nsPlayer = new NetStream(nc);
            nsPlayer.client = this;
            nsPlayer.addEventListener(NetStatusEvent.NET_STATUS, onNetStatus);
            nsPlayer.addEventListener(AsyncErrorEvent.ASYNC_ERROR, onAsyncError);
            nsPlayer.play("video", 0, -1);
            vidPlayer = new Video(cam.width, cam.height);
            vidPlayer.x = cam.width + 20;
            vidPlayer.y = 10;
            vidPlayer.attachNetStream(nsPlayer);
            addChild(vidPlayer);
        }

        private function getSeekToLiveValue():uint{
            currentDuration = Number((getTimer()-stamp)/1000) + duration;
            trace("currentDuration: " + currentDuration);
            seekVal = (currentDuration - nsPlayer.bufferTime) - 2;
            trace("seekVal: " + seekVal);
            return seekVal;
        }
        private function setupButtons():void {
            rewindBtn = new Button();
            pauseBtn = new Button();
            playBtn = new Button();
            seekBtn = new Button();
            rewindBtn.width = 52;
            pauseBtn.width = 52;
            playBtn.width = 52;
            seekBtn.width = 100;
            rewindBtn.move(180,150);
            pauseBtn.move(235,150);
            playBtn.move(290,150);
            seekBtn.move(345, 150);
            rewindBtn.label = "Rew 5s";
            pauseBtn.label = "Pause";
            playBtn.label = "Play";
            seekBtn.label = "Seek to Live";
            rewindBtn.addEventListener(MouseEvent.CLICK, onClick);
            pauseBtn.addEventListener(MouseEvent.CLICK, onClick);
            playBtn.addEventListener(MouseEvent.CLICK, onClick);
            seekBtn.addEventListener(MouseEvent.CLICK, onClick);
            addChild(rewindBtn);
            addChild(pauseBtn);
            addChild(playBtn);
            addChild(seekBtn);
        }

        public function onMetaData(info:Object):void {
        trace("metadata:duration = " + info.duration);
            stamp = getTimer();
            trace("stamp: " + stamp);
            duration = info.duration;
    }
    }
}
```

4 Save the DVR.as file.

5 Choose File > New > Flash File (ActionScript 3.0) and click OK.

6 Save the file as DVR.fla in the same folder as the DVR.as file.

7 Open the Components Panel, drag a Button to the Stage, and delete it.

This action adds the button to the Library. The button is added to the application at runtime.

8 Choose File > Publish Settings. Click the Flash tab. Click Script Settings and enter DVR as the Document class. Click the checkmark to validate the path.

9 Save the file and choose Control > Test Movie to run the application.

Limiting the size and duration of recordings

You can limit the maximum size and duration of recordings using parameters in the Application.xml configuration file, Server-Side ActionScript, and the Authorization plug-in. Set these values to prevent using excessive disk space. The following are the Application.xml file parameters:

XML element	Description
`Application/StreamManager/Recording/MaxDuration`	The maximum duration of a recording, in seconds. The default value is -1, which enforces no maximum duration.
`Application/StreamManager/Recording/MaxDurationCap`	The maximum duration of a recording cap, in seconds. The default value is -1, which enforces no cap on maximum duration. The server-side `Stream.record()` method cannot override this value. The Authorization plug-in can override this value.
`Application/StreamManager/Recording/MaxSize`	The maximum size of a recording, in kilobytes. The default value is -1, which enforces no maximum size.
`Application/StreamManager/Recording/MaxSizeCap`	The maximum size of a recording cap, in kilobytes. The default value is -1, which enforces no cap on maximum size. The server-side `Stream.record()` method cannot override this value. The Authorization plug-in can override this value.

In addition to setting maximum values, in the Application.xml file you can set maximum cap values. Server-side scripts cannot override these caps. CDNs can use these caps to set a limit that clients cannot override.

***Note:** The Authorization plug-in can override any values set in the Application.xml file.*

To set values in Server-Side ActionScript, call `Stream.record()` and pass values for the `maxDuration` and `maxSize` parameters. The following code limits the recording to 5 minutes and sets an unlimited maximum file size (up to the value of `MaxSizeCap`):

```
s.record("record", 300, -1);
```

The server truncates recordings greater than `MaxCapSize` and `MaxCapDuration`.

More Help topics

"Developing an Authorization plug-in" on page 303

Scaling DVR applications

To build large-scale applications, use the server-side NetConnection class to chain multiple servers together. In this scenario, a client can request a stream that does not reside on the server to which it is connected. Use the server-side ProxyStream class to create a look-up mechanism for finding streams in the server chain.

You can set values in the Vhost.xml configuration file to configure the disk cache that holds the streams:

XML element	Attribute	Description
`VirtualHost/Proxy/CacheDir`	`enabled`	Determines whether the disk cache is enabled.
	`useAppDir`	Specifies whether to separate the cache subdirectories by application.
`VirtualHost/Proxy/CacheDir/Path`		The root directory of the disk cache.
`VirtualHost/Proxy/CacheDir/MaxSize`		The maximum size of the disk in gigabytes. The default value is 32. The value 0 disables the disk cache. The value -1 specifies no maximum value.
`VirtualHost/Proxy/RequestTimeout`		The maximum amount of time to wait for a response to a request (for metadata, content, and so on) from an upstream server., in seconds. The default value is 2 seconds.

If a server has multiple virtual hosts, point each virtual host to its own cache directory.

If the server runs out of disk space on an intermediate or edge server while writing to the `CacheDir`, it logs the following warning message in the core.xx.log for each segment that fails to write to disk:I/O Failed on cached stream file C:\Program Files\Adobe\Adobe Media Server 3.5\cache\streams\00\proxyapp\<IP>\C\Program Files\Adobe\Adobe Media Server 3.5_361\applications\primaryapp\streams_definst_\sample1_1500kbps.f4v\0000000000000000 during write: 28 No space left on device.

Logging

Streams played in a DVR video player are played as recorded streams. These streams log the same events in the log files as every recorded stream.

More Help topics

"Dynamic streaming" on page 210

Adding metadata to a live stream

About metadata

See Adobe Evangelist Jens Loeffler's article Working with metadata for live Flash video streaming.

Metadata in streaming media gives subscribers the opportunity to get information about the media they are viewing. Metadata can contain information about the video, such as title, copyright information, duration of the video, or creation date. A client can use the metadata to set the width and height of the video player.

In a recorded stream, a special data message is inserted at the beginning of the media file that provides metadata. Any client that connects to Adobe Media Server receives the metadata when it plays the recorded stream. However, when a client connects to a live stream during the broadcast, they miss receiving the data keyframe.

You can write code that tells Adobe Media Server to send metadata to clients whenever they connect to a live stream. Any client connecting to the server, even if they connect late, receives the metadata when it plays the live video.

You can also use this feature to add metadata to a live stream at any time during a broadcast.

***Note:** Because DVR applications use recorded streams, you do not need to use data keyframes to push metadata to the client. In DVR applications (and in all recorded video applications) the `onMetaData()` method is called at the beginning of the stream and during events such as seek and pause.*

Sending metadata to a live stream

To send metadata to clients when they connect to a live stream, pass a special command, `@setDataFrame,` to the client-side `NetStream.send()` method or to the server-side `Stream.send()` method. Whether the command is sent from the client or the server, handle the data on the client the same way you would handle any message from the `send()` method. Pass a handler name to the `send()` method and define a function with that name to handle the data.

To send the command from the client, call the `NetStream.send()` method:

```
NetStream.send(@setDataFrame, onMetaData [,metadata ])
```

The *`onMetaData`* parameter specifies a function that handles the metadata when it's received. You can create multiple data keyframes. Each data keyframe must use a unique handler (for example, `onMetaData1`, `onMetaData2`, and so on).

The *metadata* parameter is an Object or Array (or any subclass) that contains the metadata to set in the stream. Each metadata item is a property with a name and a value set in the metadata object. You can use any name, but Adobe recommends that you use common names, so that the metadata you set can be easily read.

To add metadata to a live stream in a client-side script, use the following code:

```
var metaData:Object = new Object();
metaData.title = "myStream";
metaData.width = 400;
metaData.height = 200;
ns.send("@setDataFrame", "onMetaData", metaData);
```

To clear metadata from a live stream in a client-side script, use the following code:

```
ns.send("@clearDataFrame", "onMetaData");
```

To add metadata to a live stream in a server-side script, use the following code:

```
s = Stream.get("myStream");
metaData = new Object();
metaData.title = "myStream";
metaData.width = 400;
metaData.height = 200;
s.send("@setDataFrame", "onMetaData", metaData);
```

To clear metadata from a live stream in a server-side script, use the following code:

```
s.send("@clearDataFrame", "onMetaData");
```

Retrieving metadata

You can retrieve metadata from client-side code only. You cannot retreive metadata from server-side code. Even if you send the @setDataFrame message from the server, use client-side code to retrieve it.

To retrieve metadata, assign the NetStream.client property to an object. Define an onMetaData function on that object, as in the following:

```
netstream.client = this;

function onMetaData(info:Object):void {
    var key:String;
    for (key in info) {
        trace(key + ": " + info[key]);
    }
}
```

This function outputs the metadata added by the server and the metadata sent with @setDataFrame.

To have the server add metadata in addition to any messages you send with the @setDataFrame message, publish the video with the "record" flag or the "append" flag, as in the following:

```
netstream.publish("mycamera", "record");
netstream.publish("mycamera", "append");
```

***Note:** If you publish a live video to the server with the "live" flag or without a type parameter, the server does not record the video. In this case, the server does not add standard metadata to the file.*

To play the stream, call NetStream.play() and pass a value for the start parameter to indicate that the stream is a recorded live stream:

```
netstream.play("mycamera", 0); // Plays a recorded live stream.
```

If you pass 0 or greater for the `start` parameter, the client plays the recorded stream starting from the time given. The recorded stream includes the standard metadata added by the server. If the server doesn't find a recorded stream, it ignores the `play()` method.

If you pass -2 or -1 for the `start` parameter, the client plays the live video and does not receive the standard metadata:

```
netstream.play("mycamera", -2); // Looks for a live stream first.
netstream.play("mycamera", -1); // Plays a live stream.
```

Example: Add metadata to live video

This example is a client application that does the following:

- Captures and encodes video.
- Displays the video as it's captured.
- Streams video from the client to Adobe Media Server.
- Sends metadata to the server that the server sends to clients when they play the live stream.
- Streams video from Adobe Media Server back to the client when you click a button.
- Displays the video streamed from the server.
- Displays the metadata sent from the server in a TextArea component.

***Note:** To test this code, create a RootInstall/applications/publishlive folder on the server. Open the RootInstall/documentation/samples/metadata/Metadata.swf file to connect to the application.*

1 On Adobe Media Server, create a *RootInstall*/applications/publishlive folder.

2 In Flash, create an ActionScript file and save it as Metadata.as.

3 Copy and paste the following code into the Script window:

```
package {
    import flash.display.MovieClip;
    import flash.net.NetConnection;
    import flash.events.NetStatusEvent;
    import flash.events.MouseEvent;
    import flash.events.AsyncErrorEvent;
    import flash.net.NetStream;
    import flash.media.Video;
    import flash.media.Camera;
    import flash.media.Microphone;
    import fl.controls.Button;
    import fl.controls.Label;
    import fl.controls.TextArea;
    public class Metadata extends MovieClip {
        private var nc:NetConnection;
        private var ns:NetStream;
        private var nsPlayer:NetStream;
        private var vid:Video;
        private var vidPlayer:Video;
        private var cam:Camera;
        private var mic:Microphone;
        private var clearBtn:Button;
        private var startPlaybackBtn:Button;
        private var outgoingLbl:Label;
        private var incomingLbl:Label;
        private var myMetadata:Object;
        private var outputWindow:TextArea;

        public function Metadata(){
            setupUI();
            nc = new NetConnection();
            nc.addEventListener(NetStatusEvent.NET_STATUS, onNetStatus);
            nc.connect("rtmp://localhost/publishlive");
        }

        /*
         *  Clear the MetaData associated with the stream
         */
        private function clearHandler(event:MouseEvent):void {
            if (ns){
                trace("Clearing MetaData");
                ns.send("@clearDataFrame", "onMetaData");
            }
        }

        private function startHandler(event:MouseEvent):void {
            displayPlaybackVideo();
        }

        private function onNetStatus(event:NetStatusEvent):void {
            trace(event.target + ": " + event.info.code);
            switch (event.info.code)
            {
                case "NetConnection.Connect.Success":
                    publishCamera();
                    displayPublishingVideo();
                    break;
                case "NetStream.Publish.Start":
                    sendMetadata();
                    break;
            }
        }

        private function asyncErrorHandler(event:AsyncErrorEvent):void {
            trace(event.text);
        }
        private function sendMetadata():void {
            trace("sendMetaData() called")
            myMetadata = new Object();
         myMetadata.customProp = "Welcome to the Live feed of YOUR LIFE, already in progress.";
            ns.send("@setDataFrame", "onMetaData", myMetadata);
        }
        private function publishCamera():void {
            cam = Camera.getCamera();
            mic = Microphone.getMicrophone();
            ns = new NetStream(nc);
            ns.client = this;
            ns.addEventListener(NetStatusEvent.NET_STATUS, onNetStatus);
            ns.addEventListener(AsyncErrorEvent.ASYNC_ERROR, asyncErrorHandler);
            ns.attachCamera(cam);
            ns.attachAudio(mic);
            ns.publish("myCamera", "record");
        }
```

```
private function displayPublishingVideo():void {
    vid = new Video(cam.width, cam.height);
    vid.x = 10;
    vid.y = 10;
    vid.attachCamera(cam);
    addChild(vid);
}
private function displayPlaybackVideo():void {
    nsPlayer = new NetStream(nc);
    nsPlayer.client = this;
    nsPlayer.addEventListener(NetStatusEvent.NET_STATUS, onNetStatus);
    nsPlayer.addEventListener(AsyncErrorEvent.ASYNC_ERROR, asyncErrorHandler);
    nsPlayer.play("myCamera", 0);
    vidPlayer = new Video(cam.width, cam.height);
    vidPlayer.x = cam.width + 100;
    vidPlayer.y = 10;
    vidPlayer.attachNetStream(nsPlayer);
    addChild(vidPlayer);
}

private function setupUI():void {
    outputWindow = new TextArea();
    outputWindow.move(250, 175);
    outputWindow.width = 250;
    outputWindow.height = 150;

    outgoingLbl = new Label();
    incomingLbl = new Label();
    outgoingLbl.width = 150;
    incomingLbl.width = 150;
    outgoingLbl.text = "Publishing Stream";
    incomingLbl.text = "Playback Stream";
    outgoingLbl.move(30, 150);
    incomingLbl.move(300, 150);

    startPlaybackBtn = new Button();
    startPlaybackBtn.width = 150;
    startPlaybackBtn.move(250, 345)
    startPlaybackBtn.label = "View Live Event";
    startPlaybackBtn.addEventListener(MouseEvent.CLICK, startHandler);

    clearBtn = new Button();
```

```
            clearBtn.width = 100;
            clearBtn.move(135,345);
            clearBtn.label = "Clear Metadata";
            clearBtn.addEventListener(MouseEvent.CLICK, clearHandler);

            addChild(clearBtn);
            addChild(outgoingLbl);
            addChild(incomingLbl);
            addChild(startPlaybackBtn);
            addChild(outputWindow);
        }

        public function onMetaData(info:Object):void {
            var key:String;
            for (key in info){
                outputWindow.appendText(key + ": " + info[key] + "\n");
            }
        }
    }
}
```

4 Save the file.

5 Choose File > New > Flash File (ActionScript 3.0) and click OK.

6 Save the file as Metadata.fla in the same folder as the Metadata.as file.

7 Open the Components Panel, drag a Button and a TextArea component to the Stage, and delete them.

This action adds the components to the Library. The components are added to the application at runtime.

8 Choose File > Publish Settings. Click the Flash tab. Click Script Settings and enter Metadata as the Document class. Click the checkmark to validate the path.

9 Save the file and choose Control > Test Movie to run the application.

Flash Media Live Encoder metadata properties

Flash Media Live Encoder sets the following metadata properties and values. You do not need to add this metadata to live streams:

Metadata property name	**Data type**	**Description**
`lastkeyframetimestamp`	Number	The timestamp of the last video keyframe recorded.
`width`	Number	The width of the video, in pixels.
`height`	Number	The height of the video, in pixels.
`videodatarate`	Number	The video bit rate.
`audiodatarate`	Number	The audio bit rate.
`framerate`	Number	The frames per second at which the video was recorded.
`creationdate`	String	The creation date of the file.
`createdby`	String	The creator of the file.

Metadata property name	Data type	Description
audiocodecid	Number	The audio codec ID used in the file. Values are: 0 Uncompressed 1 ADPCM 2 MP3 5 Nellymoser 8 kHz Mono 6 Nellymoser 10 HE-AAC 11 Speex
videocodecid	Number	The video codec ID used in the file. Values are: 2 Sorenson H.263 3 Screen video 4 On2 VP6 5 On2 VP6 with transparency 7 H.264
audiodelay	Number	The delay introduced by the audio codec, in seconds.

Metadata properties for recorded live streams

If you record the file as you stream it, Adobe Media Server adds the metadata listed in the following table. To record a file as you publish it to the server, use the "record" parameter, as in the following:

```
ns.publish("myCamera", "record");
```

Metadata property name	Data type	Description
audiocodecid	Number	The audio codec ID used in the file. Values are: 0 Uncompressed 1 ADPCM 2 MP3 5 Nellymoser 8kHz Mono 6 Nellymoser 10 HE-AAC 11 Speex
canSeekToEnd	Boolean	Whether the last video frame is a keyframe (`true` if yes, `false` if no).
createdby	String	The name of the file creator.

Metadata property name	Data type	Description
`duration`	Number	The length of the file, in seconds.
`creationdate`	String	The date the file was created.
`videocodecid`	Number	The video codec ID used in the file. Values are: 2 Sorenson H.263 3 Screen video 4 On2 VP6 5 On2 VP6 with transparency 7 H.264

When you use the `"record"` flag with the `NetStream.publish()` call, the server attempts to merge your metadata properties with the standard metadata properties. If there is a conflict between the two, the server uses the standard metadata properties. For example, suppose you add these metadata properties:

```
duration=5
x=200
y=300
```

When the server starts to record the video, it begins to write its own metadata properties to the file, including `duration`. If the recording is 20 seconds long, the server adds `duration=20` to the metadata, overwriting the value you specified. However, `x=200` and `y=300` are still saved as metadata, because they create no conflict. The other properties the server sets, such as `audiocodecid`, `videocodecid`, `creationdate`, and so on, are also saved in the file.

When a stream is recorded, the recording starts at a keyframe. Recordings stop immediately on keyframes or I-frames.

Capturing timecode sent from Flash Media Live Encoder

Flash Media Live Encoder can embed SMPTE timecode, Vertical Interval Timecode (VITC), and Visual Timecode (BITC) in a stream and pass to Adobe Media Server. To capture the timecode, implement the OnFI method on the NetStream object.

Publishing live video in RAW file format

Flash Media Server 3.5.3

About the RAW file format

The RAW (Record and Watch) file format records media into configurable chunks that stream to any version of Flash Player. Use the RAW file format to serve long-length, multi-bitrate DVR streams without running into performance issues. The RAW file format records and plays back all streams that Adobe Media Server supports, including H.264 video, data-only, audio-only, and so on.

The RAW file format is a server feature; any version of Flash Player can publish or play a RAW stream. However, multi-bitrate stream support (also called dynamic streaming) requires Flash Player 10 and higher.

Important: *The RAW file format is internal to Adobe Media Server. At this time, you cannot edit these files with third-party tools or convert the files to FLV format or MP4 format.*

The RAW file format is an FLV file fragmented into the following files:

Filename	Description
index	Contains the list of segment files and their timestamp ranges.
context	Contains all the "context messages" for the stream.
A 16-digit hexadecimal number	There is one file for each segment of the stream. The number of files is the stream index of the first message in the segment. The first segment name is always 0. If the first segment contains 234 messages, the next segment name is EA, and so on.

The files are stored in a folder whose name is the name of the stream. Suppose the stream "foo" is stored in a folder named "foo". The "foo" folder contains the following files: index, context, 0000000000000000, 00000000000001C3, 0000000000000386, and so on.

The RAW file format enables Adobe Media Server to handle up to the following scenario:

Parameter	Value
Simultaneous DVR streams	25
Bit rate of each stream	2 Mbit
Codecs	H.264 and AAC
Stream duration	4 hours (x 2 Mbit = 7 GB size)
Number of clients	Depends on origin-edge configuration.

Note: *Raw streams with long durations create many files in a single directory. Depending on system resources, the file system may not be able to access the directory fast enough for recording or playback to keep up.*

More Help topics

"Dynamic streaming" on page 210

Streaming RAW files

Note: *Both ActionScript 2.0 and ActionScript 3.0 support the RAW file format. Flash Media Live Encoder 3.0 does not support the RAW file format.*

To record a live stream as a RAW stream, use the prefix `raw:` in the `NetStream.publish()` call or in the `Stream.get()` call.

The following client-side ActionScript uses the RAW file format to publish a live stream:

```
nc:NetConnection = new NetConnection();
nc.connect("rtmp://fms.example.com/live");
// In production code, test for a successful NetConnection here
ns:NetStream = new NetStream(nc);
ns.publish("raw:livestream", "record");
// You can use the "record" or the "append" flag.
```

The following Server-Side ActionScript records a live stream as a RAW stream:

```
s = Stream.get("raw:recordedStream");
s.record();
s.play("livestream", -1, -1);
```

The following client-side ActionScript plays the RAW stream:

```
ns.play("raw:livestream", 0, -1)
```

To use Flash Media Live Encoder, use the prefix `raw:` in the Stream field. Use a server-side script to record the stream. You can use the server-side script in the DVRCast application. Download the DVRCast application from www.adobe.com/go/ams_tools. The following example uses Flash Media Live Encoder and the DVRCast application.

Example: Publish and play a RAW stream

1. Download and install Flash Media Live Encoder from www.adobe.com/go/fmle.
2. Download the DVRCast application from www.adobe.com/go/ams_tools and do the following:
   a. Unzip the package.
   b. Copy the dvrcast1_1\DVRCast1.1\adobe\fms\samples\applications\dvrcast_origin folder and its contents to the Adobe Media Server *rootinstall*/applications folder.
   c. To verify that the application is registered with the server, open the Adobe Media Server Administration Console. Click View Applications and select dvrcast_origin from the New Instance menu at the bottom of the screen.
3. Connect a camera to your computer.
4. Open Flash Media Live Encoder and do the following:
   a. Choose a preset. The RAW format supports all the codecs that Flash Player supports.
   b. In the Output section, for the FMS URL, enter **rtmp://localhost/dvrcast_origin**.
   c. For stream, enter **raw:livestream**.
   d. Select DVR Auto Record
   e. Click Start
5. Browse to *rootinstall*\applications\dvrcast_origin\streams_definst_\livetream. The raw files are generated in this folder.
6. To view playback, do the following:
   a. Open rootinstall/samples/videoPlayer/videoplayer.html in a browser.
   b. For Stream URL, enter **rtmp://localhost/dvrcast_origin/raw:livestream**.
   c. Select VOD.
   d. Click Play Stream.

Validating RAW files and reading error messages

To check if a RAW stream is valid, pass a folder to the FLVCheck tool. For example, if you have a RAW stream named "foo" in the folder "C:\media\foo", pass the following command to the FLVCheck tool:

```
flvcheck -f C:\media\foo -v -w
```

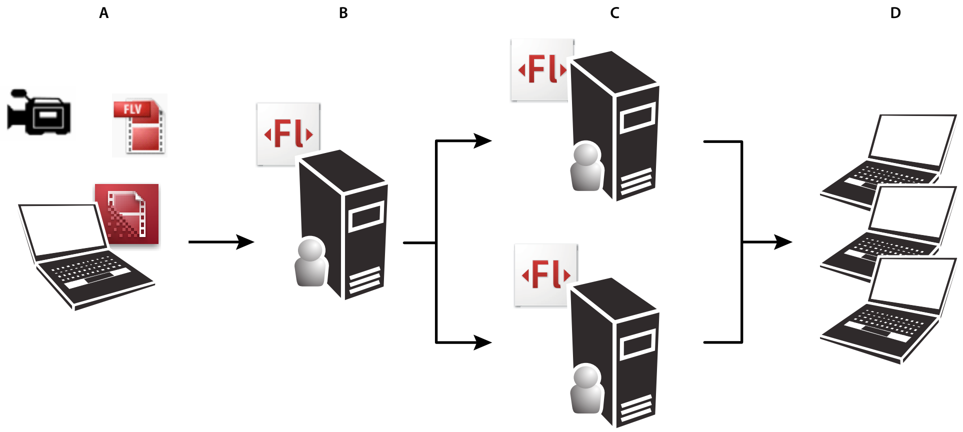
The following errors are reported by the FLVCheck tool version 2.0 and written to the log files:

Code	Error	Level	Message
-32	Missing/Corrupted Index or Context file	Error	Index or Contexts file missing or corrupted.
-33	Raw file index version doesn't match FMS version	Error	Index File Version %x Not Supported. FMS Requires Version %x.
-34	Segment file missing or corrupted	Error	Failed read in segment file %s. File missing or corrupted.
-35	Truncated message in segment file	Error	Truncated message in segment file %s.
-36	Bad message type	Error	Unrecognized message type in segment file %s.
-37	Message backtag does not match length	Error	Invalid message footer in segment file %s.
-38	Segment data does not match segment bounds	Error	Segment file %s does not match index file.
-119	Contexts file contains no `onMetadata` message (and there is at least one segment file)	Warning	Missing FLV metadata.
-126	Timestamp decreases from one message to the next	Warning	Found backward timestamp in segment file %s.

Multi-point publishing between servers

About multi-point publishing

Multi-point publishing allows clients to publish to servers with only one client-to-server connection. This feature enables you to build large-scale live broadcasting applications, even with servers or subscribers in different geographic locations.



Using multipoint publishing to publish content from server to server, even across geographic boundaries
***A.** Live Video **B.** Server 1 (New York City) **C.** Server 2 (Chicago) and Server 3 (Los Angeles) **D.** Users*

The application flow that corresponds to this illustration works like as follows:

1 A client connects to an application on Server 1 in New York City and calls `NetStream.publish()` to publish a live stream. This client can be a custom Flash Player or AIR application or Flash Media Live Encoder.

2 The server-side script on Server 1 receives an `application.onPublish()` event with the name of the published stream.

3 The `application.onPublish()` handler creates a NetStream object and calls `NetStream.publish()` to republish the live stream to Server 2 (Chicago) and Server 3 (Los Angeles).

4 Subscribers connecting to Server 2 and Server 3 receive the same live stream.

5 The applications receive `application.onUnpublish()` events when the clients stops publishing.

Example: Multipoint publishing

In this example, the client captures, encodes, and publishes the stream to the server. You can also use Flash Media Live Encoder for the same purpose.

***Note:** To test this code, create a RootInstall/applications/livestreams folder on the server. Open the Administration Console and create an instance of the livestreams application. Click Live Logs to see the server-side `trace()` statements as the application runs. Open the RootInstall/documentation/samples/livestreams/LiveStreams.swf file to connect to the application.*

1 In a client-side script, call the `NetStream.publish()` method to publish a live stream:

```
ns.publish("localnews", "live");
```

***Note:** To use Flash Media Live Encoder as a publishing client, enter the FMS URL `rtmp://localhost/livestreams` and the Stream `localnews`.*

2 In the server-side main.asc file, define an `application.onPublish()` event handler. This handler accepts the stream name that was published from the client, connects to the remote server, and republishes the stream to the remote server. (In this example, the remove server is another instance of the same application).

```
// Called when the client publishes
application.onPublish = function(client, myStream) {
    trace(myStream.name + " is publishing into application " + application.name);
    // This is an example of using the multi-point publish feature to republish
    // streams to another application instance on the local server.
    if (application.name == "livestreams/_definst_"){
        trace("Republishing the stream into livestreams/anotherinstance");
        nc = new NetConnection();
        nc.connect( "rtmp://localhost/livestreams/anotherinstance" );
        ns = new NetStream(nc);
        // called when the server NetStream object has a status
        ns.onStatus = function(info) {
            trace("Stream Status: " + info.code)
            if (info.code == "NetStream.Publish.Start") {
                trace("The stream is now publishing");
            }
        }
        ns.setBufferTime(2);
        ns.attach(myStream);
        ns.publish( myStream.name, "live" );
    }
}
```

Calling `NetStream.publish()` publishes the stream from your server to the remote server.

3 In the main.asc file, handle events that occur on the NetStream object you used to publish from your server to the remote server:

```
ns.onStatus = function(info) {
    trace("Stream Status: " + info.code)
    if (info.code == "NetStream.Publish.Start") {
        trace("The stream is now publishing");
    }
}
```

The server-side `NetStream.publish()` method triggers a `NetStatus` event with a `NetStream.Publish.Start` code, just like the client-side `NetStream.publish()` method.

4 Define what happens when the client stops publishing:

```
application.onUnpublish = function( client, myStream ) {
    trace(myStream.name + " is unpublishing"  );
}
```

Chapter 6: Building peer-assisted networking applications

Real-Time Media Flow Protocol (RTMFP)

Flash Player 10, AIR 1.5, Flash Media Server 3.5

Flash Player 10 and AIR 1.5 support Real-Time Media Flow Protocol (RTMFP). RTMFP is built on User Datagram Protocol (UDP). RTMP is built on Transmission Control Protocol (TCP). UDP provides lower latency than TCP. It also enables end-to-end peering—that is, direct data transmission between two clients. You can substitute RTMFP for RTMP in traditional unicast (one-to-many or one-to-one) applications to take advantage of the lower latency and lower bandwidth costs.

RTMFP provides the following features: NAT/firewall traversal, congestion control and prioritization, IP address mobility, and partial reliability.

RTMFP network traffic is encrypted with a 128-bit cipher. To play a stream over RTMFP, a client must know the name of the stream and know the peer ID of the publisher. The peer ID is a 256-bit value associated with the publisher's identity. The publisher must accept a peer request before a connection is made.

Adobe Computer Scientist Jozsef Vass explains RTMFP in an Adobe DevNet article, Best practices for real-time collaboration using Adobe Media Server.

About peer IDs

Each client has a peer ID, and the peer ID space is server-specific. The peer ID is the SHA256 of the client's Diffie-Hellman public key.

For a client-server connection, peer IDs are available in the ActionScript 3.0 `NetConnection.nearID` and `NetConnection.farID` properties. The `nearID` is the peer ID of the client, the `farID` is the peer ID of the server.

Peer IDs are also available in the Server-Side ActionScript `Client.nearID` and `Client.farID` properties. The `nearID` is the peer ID of the server, the `farID` is the peer ID of the client.

For a connection between Flash Player and Adobe Media Server, the client-side `NetConnection.nearID` and the server-side `Client.farID` properties share the same value. The client-side `NetConnection.farID` and the server-side `Client.nearID` properties share the same value.

Server-Side ActionScript also has `NetConnection.nearID` and `NetConnection.farID` properties that contain the peer IDs of both sides of a server-side RTMFP connection. In this case, the NetConnection is a client.

Unicast, broadcast, and multi-point publishing over RTMFP

Although RTMFP is often used for peer-assisted networking applications, you can use RTMFP in unicast, broadcast, and multi-point publishing applications as well. Simply replace the RTMP protocol in the `NetConnection.connect()` call with the RTMFP protocol:

```
netconnection.connect("rtmfp://fms.example.com/vod");
```

To use RTMFP with multi-point publishing, create a server-side NetConnection and use an RTMFP URL to connect to the target server (for example, `"rtmfp://localhost/myapp"`).

To stream audio, video, and data, create a NetStream just as you would with an RTMP connection. Do not pass a GroupSpecifier to the NetStream constructor as you would with a peer-assisted networking application.

When you use an RTMFP connection, updates to shared objects are sent over the RTMP chunk stream data channel within the RTMFP connection with full reliability. The RTMFP protocol supports a wide range of reliability settings (from zero to full) for audio, video, and data transfer, along with excellent retransmission and congestion handling. See `NetStream.audioReliable`, `NetStream.videoReliable`, and `NetStream.dataReliable` in the *ActionScript 3.0 Reference for Flash Platform*.

RTMFP groups

Flash Player 10.1, AIR 2, Flash Media Server 4

Flash Player 10.1, AIR 2, and Flash Media Server 4 support RTMFP groups. Clients make an RTMFP connection to Adobe Media Server and join a group.

Peer A member of a group, also called a "node". A peer is a Flash Player or AIR client that connects to Adobe Media Server over RTMFP and joins a group.

Group A group is a collection of one or more RTMFP nodes who agree on certain parameters and capabilities. There is a path between every peer in a group that forms a peer-to-peer mesh. There may not be a direct connection between two peers, but there is a always a path through other peers. A group is also called a "peer group" an "RTMFP group" and a "NetGroup".

Bootstrapping Connecting to at least one member of a group in order to join the group. You can write application logic to bootstrap a client to a group, or you can ask Adobe Media Server to bootstrap clients automatically. There is no random probing for peers and the server does not flood the network looking for peers.

Adobe Media Server handles introducing a client to a group. Flash Player handles peer communication within a group, organizing and optimizing a group for latency, and maintaining the full-connectedness of a group.

The group maintains the connections between peers automatically. Once peers are fully meshed into a group, then they can pass data within the group. Data does not need to be sent to the server to be distributed to each client. Peers can share data such as audio, video, and ActionScript objects. RTMFP and groups allow you to build peer-to-peer applications that can scale to millions of clients.

Use the ActionScript 3.0 GroupSpecifier class to define the parameters and capabilities of a group in a string called a "groupspec". Pass the groupspec to the NetGroup and NetStream constructors. Use the NetGroup class to manage a group and send ActionScript data within a group. Use the NetStream class to multicast audio and video data within a group.

Groupspecs are Strings that start with "G:" followed by hexadecimal digits, for example "G:01010b...". The first part of the string is the unchangeable identity of the group and contains the group permissions. If a peer changes any permissions or properties of the group, they start a new group. The string is case-sensitive.

Groups allow you to do the following things:

- "Multicasting" on page 264 streams

  Use multicasting for one sender or a few senders to deliver a streams of data to everyone in a group.

- "Post messages to a group" on page 262

  Use posting for lots of senders to deliver a small amount of data to everyone in a group.

- "Route messages directly to a peer" on page 263

Use directed routing to send a message through the overlay routing structure to one peer.

- "Replicate an object within a group" on page 264

  Use object replication to synchronize everyone in a group on a (potentially large) number of objects.

To multicast streams and post messages, call ActionScript methods and pass them the data to send. The data arrives at the other nodes in the group.

To route messages directly to a peer and replicate an object within a group, you must write ActionScript code to help deliver the data.

Create a group

1 Connect to Adobe Media Server. Pass an "rtmfp" URL to the `NetConnection.connect()` method:;

```
NetConnection.connect("rtmfp://fms.example.com/p2pexample/test1")
```

Note: If RTMFP is disabled in the Adaptor.xml file, it can take up to 2 minutes for the client to receive the `"NetConnection.Connect.Failed"` status. To prevent this delay, verify that `<RTMFP enable="true">` in the Adpator.xml file. It is enabled by default.

2 On `"NetConnection.Connect.Success"`, use the GroupSpecifier class to create a groupspec. Pass the groupspec to the NetGroup constructor.

```
 // Called in the "NetConnection.Connect.Success" case in the NetStatusEvent handler.
private function OnConnect():void{
    connected = true;
    // Create a GroupSpecifier object to pass to the NetGroup constructor.
    // The GroupSpecifier determines the properties of the group
    var groupSpecifier:GroupSpecifier;
    groupSpecifier = new GroupSpecifier("com.example.p2papp");
    groupSpecifier.postingEnabled = true;
    groupSpecifier.multicastEnabled = true;
    // The serverChannel lets the server do auto-bootstrapping
    groupSpecifier.serverChannelEnabled = true;
    netGroup = new NetGroup(netConnection, groupSpecifier.groupspecWithAuthorizations());
    netGroup.addEventListener(NetStatusEvent.NET_STATUS, NetStatusHandler);
}
```

3 On `"NetGroup.Connect.Success"`, you can manually bootstrap or automatically bootstrap peers to the group. The example enables the server channel to perform automatic bootstrapping. There are several additional options for bootstrapping described in the next section.

Additional information

Tom Krcha has written an in-depth article about the GroupSpecifier class on his blog Flash Realtime.

Bootstrap a peer to a group

After the initial connection to the server, a peer must be introduced to one or more peers of the same group, this technique is called "bootstrapping". Bootstrapping allows members of the same group to see each another. After bootstrapping, the peer self-organizes into the group and meshes with the other members.

To join a group, a client must know the GroupSpecifier that defines the group. If two sets of clients use the same GroupSpecifier, and never touch, they are in separate groups. If the two groups touch, they merge into a single larger group.

Each client has a peer ID, and the peer ID space is server-specific. The peer ID is the SHA256 of the client's Diffie-Hellman public key. Both client-side and server-side ActionScript can access the peer ID as a property of each RTMFP client. Peer IDs are available in the ActionScript 3.0 `NetConnection.nearID` and `NetConnection.farID` properties. Peer IDs are also available in the Server-Side ActionScript `Client.nearID` and `farID` properties. Client-side `NetConnection.nearID` and server-side `Client.farID` share the same value. Client-side `NetConnection.farID` and server-side `Client.nearID` share the same value.

Peers within an RTMFP group can be bootstrapped in the following ways:

- Server channel automatic bootstrapping.

  When clients connect over an RTMFP connection, the server bootstraps them with peers who are members of the same NetGroup. To enable automatic bootstrapping on the client-side, set `GroupSpecifier.serverChannelEnabled` to `true`.

- Manual bootstrapping.

  To manually bootstrap peers into a mesh, call the `NetGroup.addNeighbor()` method.

- LAN peer discovery

  Use the GroupSpecifier class to enable LAN peer discovery. LAN peer discovery allows an RTMFP NetConnection and its NetStream and NetGroup objects to automatically locate peers and join a group on the current subnet. Peers cannot discover each other unless they're in the same group on the same subnet of the LAN. If peers with matching groupspecs are on different subnets, no error or other event is dispatched if they fail to discover each other.

  The following code shows how to enable LAN peer discovery:

  ```
  var nc = new NetConnection();
  // Protocol must be RTMFP
  nc.connect("rtmfp://fms.example.com/appname/appinstance");
  var gs = new GroupSpecifier("com.example.discovery-test");
  // Must be enabled for LAN peer discovery to work
  gs.ipMulticastMemberUpdatesEnabled = true;
  // Multicast address over which to exchange peer discovery.
  gs.addIPMulticastAddress("224.0.0.255:30000");
  // Additional GroupSpecifier configuration...
  var ns = new NetStream(nc, gs.toString());
  ```

***Note:** Call the Administration API getServerStats()command to get RTMFP peer lookup statistics. For more information, see getServerStats().*

Server-side RTMFP groups

Use Server-Side ActionScript to create an RTMFP NetConnection. You can connect to an application on the same server or to an application on another Adobe Media Server. After you create an RTMFP NetConnection, define a GroupSpecifier to create a NetGroup and a NetStream. The server application joins the group and becomes a peer in the group mesh. A single client can join multiple groups. Once connected, you can interact with other peers in the RTMFP group. You can also publish streams into the group.

A server-side netConnection is a virtual Flash client. It has its own RTMFP stack. As far as the server is concerned, a server-side script that joins a group is the same as any client.

***Note:** Server-Side ActionScript doesn't support creating a connection directly between peers.*

Flash Player peer-assisted networking security dialog

When a NetStream or NetGroup object is constructed with a groupspec, Flash Player displays a "Peer-assisted Networking" dialog. The dialog asks users if Flash Player can use their connection to share data with their peers.

If the user clicks "Allow for this domain", the dialog is not displayed the next time the user connects to this application. If a user does not allow peer-assisted networking, all peer features within the group (posting, directed routing, and object replication, and multicast) are disabled. You can use RTMFP to subscribe to a pure native-IP multicast stream. In this case, the dialog is not displayed.

When using IP multicast with no peer-assisted functionality, you can disable the security dialog. Set the `GroupSpecifier.peerToPeerDisabled` property to `true`. By default, this property is `false` (peer-assisted connections are enabled). When `peerToPeerDisabled` is `true`, the security dialog does not appear.

ActionScript classes for working with RTMFP groups

Use the following ActionScript 3.0 classes and Server-Side ActionScript classes to create applications that use RTMFP groups:

- NetConnection

  Use the NetConnection class to create a two-way RTMFP connection between a Flash Player or AIR application and a Adobe Media Server application. Use the RTMFP protocol in the URL you pass to the `NetConnection.connect()` method.

- GroupSpecifier

  Use the GroupSpecifier class to define the capabilities, restrictions, and authorizations of an RTMFP peer-to-peer group. After you define the capabilities, pass a `groupspec` string or a GroupSpecifier object to the NetStream and NetGroup constructors. A `groupspec` is an opaque string that you pass to the NetStream and NetGroup constructors.

  In server-side code, you can pass the GroupSpecifier object or a `groupspec` string.

  In client-side code, you can only pass a `groupspec` string. To generate a `groupspec` string, call one of the following methods: `toString()`, `groupSpecWithAuthorizations()`, and `groupSpecWithoutAuthorizations()`.

- NetGroup

  Use the NetGroup class to manage an RTMFP group. The class properties provide information about group members. Call the class methods to post messages to the group, route messages to a group member, and to replicate objects among the group. To create a NetGroup, pass a GroupSpecifier object to the NetGroup constructor.

- NetGroupInfo

  The NetGroupInfo class specifies Quality of Service (QoS) statistics about a NetGroup object's RTMFP peer-to-peer data transport. The `NetGroup.info` property returns a NetGroupInfo object which is a snapshot of the current QoS state.

- NetStream

  Use the NetStream class to multicast audio and video data over a group. Pass a groupspec to the NetStream constructor and use the standard `publish()` and `play()` methods.

- NetStreamMulticastInfo

  The NetStreamMulticastInfo class specifies Quality of Service (QoS) statistics about a NetStream object's RTMFP peer-to-peer and IP multicast stream transport. The `NetStream.multicastInfo` property returns a NetStreamMulticastInfo object which is a snapshot of the current QoS state.

Post messages to a group

Flash Player 10.1, AIR 2, Flash Media Server 4

Call the NetGroup.post() method to broadcast an ActionScript message to all members of a NetGroup (also called a "group"). Use the "NetGroup.Posting.Notify" code to do something when a post has been received.

You can post messages to a group from the client-side ActionScript 3.0 NetGroup.post() method or from the Server-Side ActionScript NetGroup.post() method.

Use the post() method to broadcast non-stateful data. For example, use posting for text chat, opinion polls, and sensor reporting. Use posting to allow many clients to send small amounts of data.

This method is similar to the Server-Side ActionScript Application.broadcastMsg() method, but the post() method propagates messages from node to node in a group. Posting is not similar to using shared objects. Unlike shared objects, posting does not manage changes.

Understand the following about posting messages:

- The GroupSpecifier.postingEnabled property must be true in the groupspec passed to the NetGroup constructor.
- Receive a NetGroup.Neighbor.Connect event before you call post().
- Messages are serialized in AMF. A message can be any AMF object. A message cannot be a MovieClip.
- Messages must be unique to be considered new. Use a sequence number to make messages unique.
- Message delivery is not ordered. Message delivery is not guaranteed.
- The post() method returns the messageID for this message, or null on error. The messageID is the hexadecimal of the SHA256 of the raw bytes of the serialization of the message.
- The post() method sends a NetStatusEvent to the NetGroup's event listener with "NetGroup.Posting.Notify" in the info.code property.

The NetGroup.post() entry in the *ActionScript 3.0 Reference* contains a chat application example. The following is an excerpt from that example:

```
private function OnConnect():void{
    StatusMessage("Connected\n");
    connected = true;
    // Create a GroupSpecifier object to pass to the NetGroup constructor.
    // The GroupSpecifier determines the properties of the group
    var groupSpecifier:GroupSpecifier;
    groupSpecifier = new GroupSpecifier("com.aslrexample/" + groupNameText.text);
    groupSpecifier.postingEnabled = true;
    groupSpecifier.serverChannelEnabled = true;
    netGroup = new NetGroup(netConnection, groupSpecifier.groupspecWithAuthorizations());
    netGroup.addEventListener(NetStatusEvent.NET_STATUS, NetStatusHandler);
    StatusMessage("Join \"" + groupSpecifier.groupspecWithAuthorizations() + "\"\n");
}
// Called when you the chatText field has focus and you press Enter.
private function DoPost(e:ComponentEvent):void{
    if(joinedGroup){
    // Build the message to post.
        var message:Object = new Object;
        message.user = userNameText.text;
        message.text = chatText.text;
        message.sequence = sequenceNumber++;
        message.sender = netConnection.nearID;
    // Post the message to the group.
        netGroup.post(message);
        StatusMessage("==> " + chatText.text + "\n")
    } else {
        StatusMessage("Click Connect before sending a chat message");
    }
    ClearChatText();
}
```

Route messages directly to a peer

A client can send short ActionScript messages directly to a member of a peer-to-peer group without having a direct connection to that member. This feature is called directed routing. Directed routing leverages the full transitive connectivity and the geometric properties of the Group's self-organized structure to route messages through the group.

Use directed routing to send a message to a specific client, or to construct a DHT (distributed hash table).

Successful routing of the message requires the direct participation of all peers along the dynamic path between the sender and intended recipient. Each peer acts as a relay point for the message. Peers do not automatically relay received messages. Use ActionScript to handle the receipt and local processing or forwarding of a message. Message delivery is not guaranteed.

The bandwidth load of message routing is distributed among the members of the group, it is not concentrated at the server.

Adobe Evangelist Tom Krcha explains directed routing in an article on his blog, Flash Realtime: Directed routing explained.

The following are the Server-Side ActionScript directed routing APIs:

- `NetGroup.sendToAllNeighbors()`
- NetGroup.sendToNearest()
- `NetGroup.sendToNeighbor()`

The following are the client-side ActionScript 3.0 directed routing APIs:

- `NetGroup.sendToAllNeighbors()`
- `NetGroup.sendToNearest()`
- `NetGroup.sendToNeighbor()`

Replicate an object within a group

Clients can send ActionScript objects through the peer group reliably. This feature is called *object replication*. Object replication allows all members of an RTMFP group to have a consistent view of a set of objects. Use it to replicate workspaces, create whiteboards, and transfer files, synchronize nodes with a log of operations, and so on.

Object replication uses the full transitive connectivity of the group's self-organized structure. It replicates objects through the group from nodes that have the objects to nodes that need the objects. Each object is indexed by a number that must be unique across the entire group. The objects must remain immutable.

For a detailed explanation of object replication, including a sample application, see Adobe Evangelist Tom Krcha's blog post File Sharing over P2P in Flash Player 10.1 with Object Replication.

Use the following client-side ActionScript 3.0 APIs to replicate an object within a group:

- `NetGroup.addHaveObjects()`
- `NetGroup.addWantObjects()`
- `NetGroup.denyRequestedObject()`
- `NetGroup.removeHaveObjects()`
- `NetGroup.removeWantObjects()`
- `NetGroup.writeRequestedObject()`

Use the following Server-Side ActionScript APIs for object replication and other RTMFP group features:

- ByteArray class
- `File.readBytes()`
- `File.writeBytes()`

Multicasting

Flash Player 10.1, AIR 2, Flash Media Server 4

To complete a tutorial before learning more about the details of multicasting, see "Multicast media (RTMFP)" on page 36.

Multicasting is distributing audio and video data among members of a group. The server doesn't send data to each client—the data is distributed among peers. Multicasting allows few publishers to send a large amount of data. To multicast media, pass a groupspec to the NetStream constructor. Call `NetStream.publish()` or `NetStream.play()` to multicast the data. You can use the client-side NetStream class and the server-side NetStream class to multicast data.

Adobe Media Server supports both application-level multicast and IP multicast. You can also use application-level multicast and IP multicast cooperatively for a single stream, this use case is called *multicast fusion*. Multicast fusion combines both broadcast techniques to support higher quality of service. Within a firewall, use IP multicast. Outside the firewall, or on networks that don't support IP multicast, use application-level multicast. Multicast fusion lets your neighbor outside the firewall receive fragments of video that it cannot receive via IP multicast.

Adobe Media Server can multicast streams delivered to it over RTMP connections. For example, Adobe Media Live Encoder can deliver a stream to Adobe Media Server over RTMP and Adobe Media Server can multicast it to groups over RTMFP.

Understand the following about multicast:

- Any number of streams can be published into a group. However, this practice is not recommended because each group member consumes and relays all streams, even if the streams aren't playing at that specific client.
- Streams of the same name can be published into a group.

  When a client requests to play the stream name, it plays the first stream with that name that it can find. If the publisher for that instance of the named stream stops, the playing client resets to another instance of the named stream and a `"NetStream.MulticastStream.Reset"` event is dispatched. Within an RTMFP group, there is no single arbiter of stream state that prevents multiple clients from publishing streams with the same name. (In traditional, unicast stream publishing, Adobe Media Server is aware of stream state.) This is a significant reason for the improved scalability of multicast streams. However, to protect against stream name collisions or hijacking, define a publish password for the group and providing the `groupspec` with authorizations to trusted publishers only. Pass a `groupspec` without authorization to all other clients to prevent them from publishing competing streams under the same name.
- Publishers can call `NetStream.send()` to inject data into a group.
- When a client is in an RTMFP group in which a live multicast stream is playing, the client may act as a relay point for that stream to some number of direct neighbors. To control this number, use the `NetStream.multicastPushNeighborLimit` property. The default value is to 4. All the peers within a group work co-operatively to get the stream to each other. Each client is not pulling the stream from the server independently. For this reason, consider the expected average client uplink capacity when selecting the bitrate for the multicast stream you publish. Choosing a bitrate that's too high may result in peers not being able to relay the stream smoothly.

  ***Note:** This requirement is specific to P2P multicast. IP multicast, which can be used simultaneously with P2P multicast, can be used for higher bitrate, live multicast streams in a local area network.*

Additional information

Multicast explained in Flash Player 10.1, an article on Tom Krcha's Flash Realtime blog.

Application-level multicast

By default, the peer-to-peer mesh distributes streams published into an RTMFP group. This technology is known as "application-level multicast".

IP multicast

IP multicast uses routers to send data to a specified IP address. The routers send the data to any client registered to an IP multicast group. To set up IP multicast, work with your IT department. Ask them to set up an address to publish to and to configure the enterprise multicast routers to forward the traffic appropriately.

To publish streams to a IP multicast address, call the Server-Side ActionScript `NetStream.setIPMulticastPublishAddress()` method before you start publishing.

All subscribing (playing) peers in the group must add the IP multicast address to their GroupSpecifier. Call `GroupSpecifier.addIPMulticastAddress()` when you create a groupspec. Adding a IP multicast address allows a client to listen for and receive the IP multicast traffic.

By default, application-level multicast runs concurrently with IP multicast. This technology is called “fusion multicast”. To run IP multicast without application-level multicast, set up the following in the GroupSpecifier:

- `GroupSpecifier.peerToPeerDisabled=true`

  This setting turns off peer-to-peer multicast.

- `GroupSpecifier.multicastEnabled=true`
- Call `GroupSpecifier.addIPMulticastAddress()`

  Call this method in the client-side application only. To publish from the server, the stream publishes to the address you passed to the Server-Side ActionScript `NetStream.setIPMulticastPublishAddress()` method. That address is the same address the clients are using.

Source-specific IP multicast

Flash Media Server 4.5

Source-specific multicast (SSM) allows the client to specify an IP address from which it wants to receive data. The client receives data only from this source. SSM reduces the burden on the network. To configure SSM, use the Multicast Config tool installed at *rootinstall*/tools/multicast/configurator. The tutorial “Multicast media (RTMFP)” on page 36 uses the Multicast Config tool.

Source-specific multicast limits receipt of IP multicast data to data coming from the “source”. On a multi-homed system (a system with multiple IP addresses), the server-side publisher can use the following API to bind to the desired local interface IP address:

```
netConn.rtmfpBindAddresses = ["10.58.117.135"];
```

At the subscriber, indicate the desired source address when you add the IP multicast address to the GroupSpecifier:

```
gs.addIPMulticastAddress("{mcastAddr}", {port}, "10.58.117.135");
```

Not all host operating systems support SSM (for example, pre-Lion Mac OS). If the operating system doesn’t support SSL, the client does not receive data.

Create a client-side serverless RTMFP connection

Flash Media Server 4.0

You can create a network endpoint for RTMFP group and IP multicast communication without connecting to a server, but the functionality is limited. This mode is called “serverless mode”. Use severless mode to receive a pure IP multicast stream from a publisher without requiring clients to connect to a server or bootstrap into the peer-to-peer mesh.

***Note:** In Server-Side ActionScript, use serverless mode to create a robust server-only group for distributing peer introductions. See “Distribute peer introductions across servers” on page 270.*

Because clients don’t connect to a server, there is no automatic bootstrapping. In serverless mode, configure peers to discover each other on a LAN using IP multicast. After they discover each other, they can communicate within the group. In serverless mode, clients can do the following:

- Use IP multicast to discover peers on the LAN.
- Receive streams over IP multicast.
- Send and receive streams in a NetGroup. (The NetGroup consists only of peers discovered over IP multicast on the LAN.)

The following code creates a connection in serverless mode:

```
var nc:NetConnection = new NetConnection;
nc.connect("rtmfp:");
```

The NetConnection instance `nc` is a serverless RTMFP NetConnection. The following code uses the NetConnection to make a NetStream that can receive a pure IP multicast. It also suppresses the peer-assisted networking permission dialog

```
var gs:GroupSpecifier = new GroupSpecifier("com.adobe.pureIPMulticastGroup");
gs.multicastEnabled = true;
// Prevents the P2P permission dialog from appearing.
gs.peerToPeerDisabled = true;
// Receive multicast stream on 239.255.255.1 port 30000
gs.addIPMulticastAddress("239.255.255.1:30000");
var ns:NetStream = new NetStream(nc, gs.groupspecWithAuthorizations());
```

Wait for the `"NetStream.Connect.Success"` message, which is required before you can use a group NetStream. Attach the NetStream to a video display object and call `ns.play(streamName)`. Or, you can create an group on your LAN for any of the RTMFP group modes (including peer-to-peer multicast), as in the following:

```
var gs:GroupSpecifier = new GroupSpecifier("com.adobe.myAdHocGroup");
gs.multicastEnabled = true;
gs.postingEnabled = true;
gs.ipMulticastMemberUpdatesEnabled = true;
// Peers find each other on 239.255.255.11 port 30001
gs.addIPMulticastAddress("239.255.255.11:30001");
var ns:NetStream = new NetStream(nc, gs.groupspecWithAuthorizations());
var ng:NetGroup = new NetGroup(nc, gs.groupspecWithAuthorizations());
```

Wait for `NetStream.Connect.Success` and `NetGroup.Connect.Success`. After receiving them, you can publish or play a peer-to-peer multicast stream on the NetStream, and post on the NetGroup.

See Adobe Evangelist Tom Krcha's video tutorial, Controlling the desktop with your mobile device via P2P.

Fusion multicast

By default, when you run IP multicast, you are also running application-level multicast. This technique is called "fusion multicast". Both IP and application-level multicast run coordinated and concurrently. The application-level multicast runs slightly behind the IP multicast to take advantage of the greater efficiency of IP multicast.

Checking multicast quality of service

To check whether a client is multicasting, and to check on quality of service, use the ActionScript 3.0 `NetStream.multicastInfo` property. This property is an instance of the NetStreamMulticastInfo class which specifies various QoS metrics for statistics when multicasting. For example, the `NetStreamMulticastInfo.bytesReceivedFromIPMulticast` property tells much data a client is receiving over pure IP multicast (as opposed to application-level multicast).

In Server-Side ActionScript, use the NetStreamMulticastInfo class and the MulticastStreamIngest class.

Prevent the server from unloading an application

Adobe Media Server considers an application with no inbound connections idle. Eventually, the server unloads the application. To prevent the server from unloading the application, define an `Application.onAppStop()` handler that returns `false`.

For example, consider the following scenario involving two Adobe Media Server installations, Server A and Server B. Server A acts as the peer introducer and ingests live streams. An application on Server B pulls a stream from Server A and calls `Stream.play()` to publish it to an RTMFP group. The application on Server B has outbound connections to server A, but it doesn't have inbound client connections. When an application doesn't have incoming client connections, the server considers the application idle and unloads it.

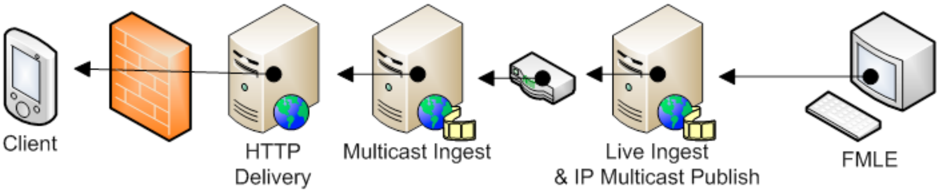
Ingest, convert, and record a multicast stream

Flash Media Server 4.5

Use Server-Side ActionScript to ingest a multicast RTMFP stream. After the server ingests the multicast stream, write a script to do the following:

- Convert the multicast stream to a Stream object.
- Deliver the Stream to clients over HTTP.

  Connect to the livepkgr application and package the Stream for delivery using HTTP Dynamic Streaming and HTTP Live Streaming.
- Deliver the Stream object to clients over RTMP/T/S/E.
- Record the Stream object.



Use multicast ingest to deliver live content across server tiers

***Note:** You cannot directly bridge an RTMFP multicast stream to another RTMFP group (by attaching the Stream to a multicast NetStream). You can bridge indirectly to another RTMFP group by publishing into the target group via a NetStream that has been attached to a MulticastStreamIngest created Stream.*

Multicast Stream Ingest API

- MulticastStreamIngest class

  Use the MulticastStreamIngest class to bind to a multicast stream in a group and transform the multicast messages to non-multicast messages. Use the class to access QoS information and to control the ingest.
- `MulticastStreamIngest.close()`

  Stops ingesting the source multicast stream.
- `MulticastStreamIngest.ingesting`

  Indicates whether the target multicast stream is bound and being ingested or not.
- `MulticastStreamIngest.multicastInfo`

  A MulticastStreamInfo object whose properties contain statistics about the stream quality of service.
- `MulticastStreamIngest.multicastPushNeighborLimit`

  The maximum number of peers to which to push multicast media.
- `MulticastStreamIngest.multicastWindowDuration`

The duration in seconds of the peer-to-peer multicast reassembly window.

- `NetGroup.getMulticastStreamIngest("livestream")`

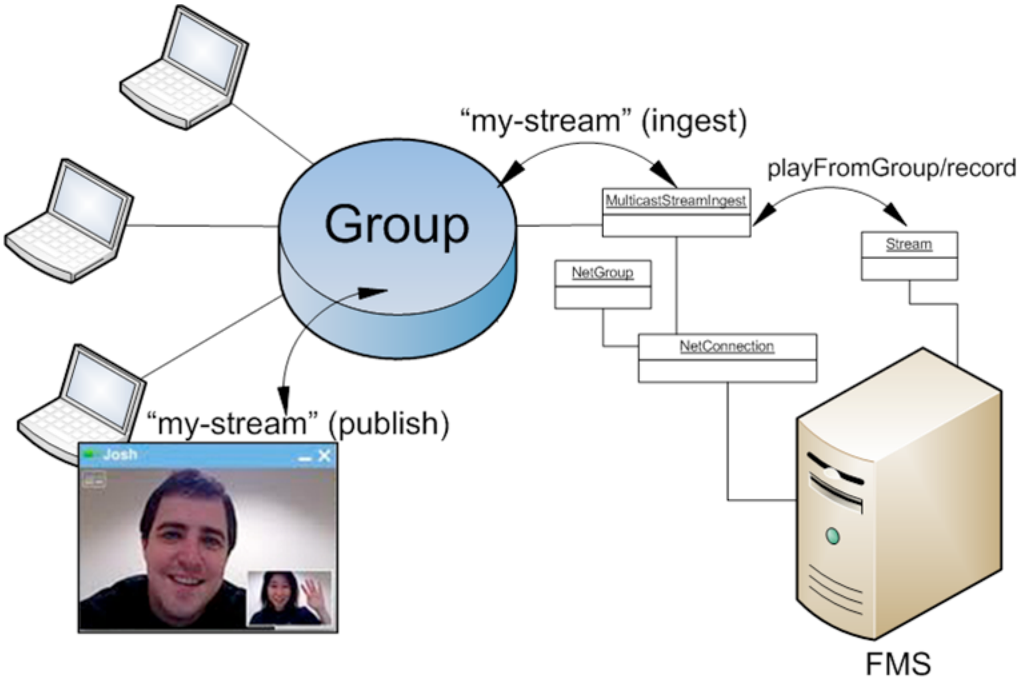
  A factory method that constructs and returns a MulticastStreamIngest object that is bound to the named stream being published into the group the NetGroup has joined.

- `Stream.playFromGroup(ingest)`

  Sets the data source for a Stream object. The `ingest` parameter is a MulticastStreamIngest object that is ingesting a multicast stream from a group.

Use the MulticastStreamIngest API

To create a MulticastStreamIngest instance and ingest a live stream, call `NetGroup.getMulticastStreamIngest()`. To play the ingested stream, call `Stream.playFromGroup()`. Use the MulticastStreamIngest class to check QoS and publishing status, stop ingesting a stream, and configure ingest settings. For more information about these APIs, see Server-Side ActionScript Reference.



Multicast ingest application flow

The following pseudocode provides a high-level description. First, set up a Stream instance that plays and records the multicast ingest:

```
var stream = Stream.get("mp4:multicast-ingest.f4v");
```

Next, set up a server-side NetConnection and NetGroup to join the group into which the multicast stream is being published:

```
var nc = new NetConnection();
nc.onStatus = function(info) {
    if (info.code == "NetConnection.Connect.Success") {
        ng = new NetGroup(nc, groupspec);
        ng.onStatus = ngStatusHandler;
    }
};
nc.connect("rtmfp://<fms-introduction-server>...");
```

Write code to handle NetGroup status events. When joining the group succeeds, indicated by a `"NetGroup.Connect.Success"` event, attempt to start ingesting the multicast stream.

```
function ngStatusHandler(info) {
    if (info.code == "NetGroup.Connect.Success") {
        ingest = ng.getMulticastStreamIngest(sourceStreamName);
    }
}
```

Play the ingested multicast stream, record the multicast stream, and stop playback:

```
stream.playFromGroup(ingest);

// The stream can be recorded locally.
stream.record();
...
stream.record(false); // And recording stopped.

// To stop playback of a multicast stream, pass the Boolean false.
stream.playFromGroup(false);
```

Note: *Server-side playlists aren't supported. However, there is a workaround. Create a remote Stream playback from an ingested Stream to a second Stream object. Use the second Stream object as part of a playlist.*

Peer-assisted networking application examples

The NetGroup entry in ActionScript 3.0 Reference contains a peer-assisted video and text chat example. The example is a Flex MXML file. To use Flash Pro instead of Flash Builder, see the example for the `NetGroup.post()` entry which is an AS file.

Adobe Evangelist Tom Krcha has several tutorials and video tutorials on his blog, FlashRealtime.com.

Distribute peer introductions across servers

Flash Media Server 4.5

Workflow for distributing introductions

Note: *You can distribute introductions only among origin servers. You cannot distribute introductions among edge servers. Also, you cannot use the Authorization Plug-in to distribute introductions.*

Adobe Media Server introduces RTMFP clients to each other so the clients can connect to each other directly and in a group. When Adobe Media Server plays this role, it's called an *introducer*. Flash Media Server 4.0 can introduce clients to each other and help them join a group only if the clients are connected to a single server. Use Server-Side ActionScript APIs added in Flash Media Server 4.5 to introduces clients to each other and let them join a group even if the clients are connected to separate servers. Distributing introductions across servers allows you to scale peer-assisted networking applications.

These steps describe the workflow for distributing peer introductions:

1 Configure the server to dispatch peer lookup events to the server-side script engine. See "Configure the server to dispatch events to the script engine" on page 274.

2 Write a server-side script that creates a robust Group of servers to act as introducers. See "Deploy servers in a robust server-only group" on page 274.

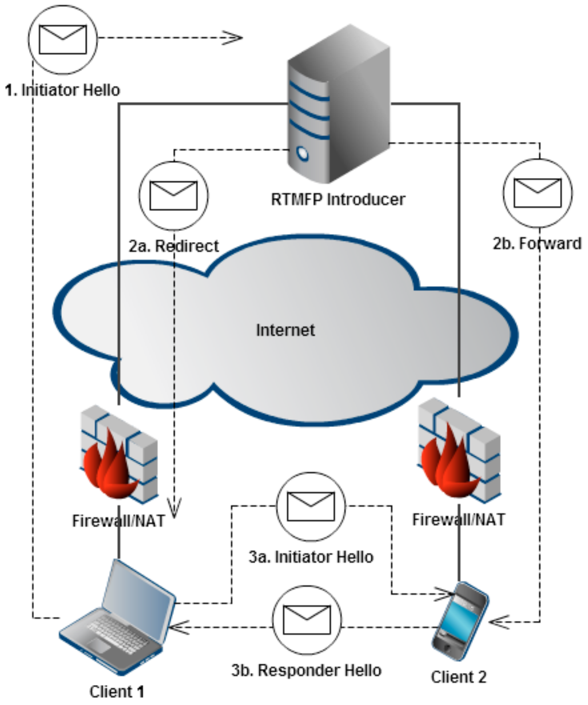
3 Write a server-side script that distributes peer introductions across servers. See "Distributed Introductions API" on page 279.

How peer introductions work

***Note:** For information about port and IP configuration and NAT traversal, see Configure ports and IP addresses for peer-assisted networking.*

Peer introduction flow on a single server

By default, the server introduces clients to each other if they connect to the same server.



The introduction flow for clients connected to a single RTMFP introducer.

Client 1 and Client 2 have established connections through their local NATs/firewalls to the same Adobe Media Server. Client 1 has discovered Client 2 either manually or through automatic peer discovery. Client 1 wants to connect directly to Client 2. The following steps describe the introduction flow:

1 Client 1 sends a peer lookup request to the introducer (Adobe Media Server). The peer lookup request contains the `NetConnection.nearID` of Client 2. The `nearID` is a globally unique fingerprint for the client and its active connection to an introducer.

2 The introducer does the following:

 a Replies to Client 1 with a redirect message that contains the IP address of Client 2.

 b Sends a forward message to Client 2. The message tells Client 2 that Client 1 wants to connect.

3 The clients do the following:

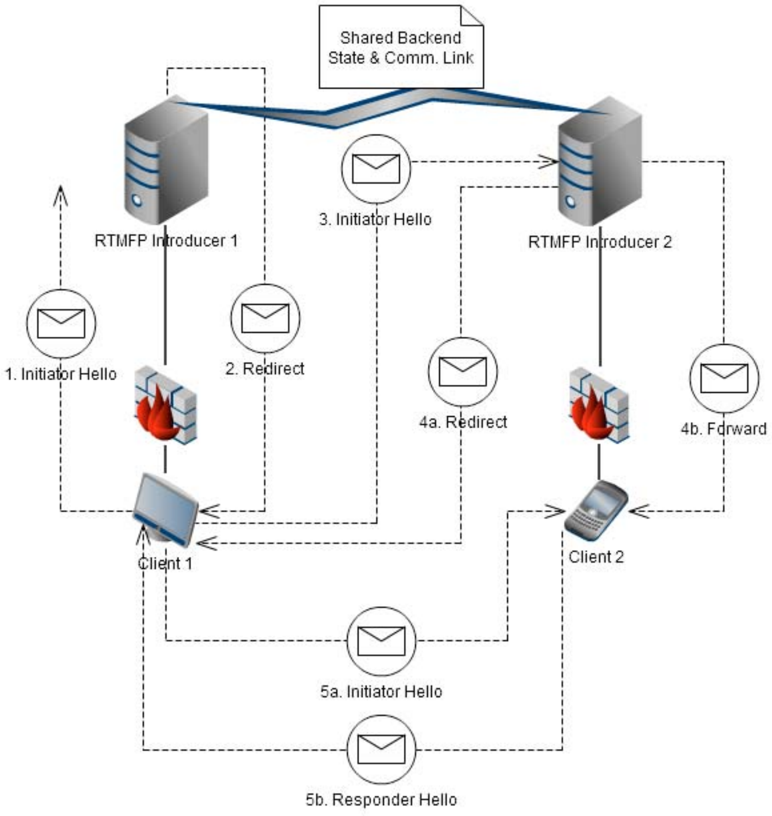
 a Client 1 resends its peer lookup request to the addresses it received in the redirect message.

 b Client 2 sends a responder hello message to the addresses of Client 1.

Because of the common introducer, the clients establish a direct connection even though they're behind NATs/firewalls.

Peer introduction application flow on multiple servers

Use Server-side ActionScript to distribute client introductions across multiple Adobe Media Server introducers.



The introduction flow for clients connected to different RTMFP introducers.

In this scenario, Client 1 is connected to Server A and Client 2 is connected to Server B. Client 1 knows the peer ID for Client 2 and wants to establish a direct connection. However, Client 2 is not connected to the same introducer as Client 1. Use Server-Side ActionScript to allow the servers to introduce these clients to each other. The following steps describe the introduction flow:

1 Client 1 sends a peer lookup request to Introducer 1.

The peer lookup request calls the `application.onPeerLookup()` event in a server-side script. The peer passes the IP address of Client 1, the peer ID of Client 2, a tag that identifies the request, and an ID for the RTMFP interface on which the request was received.

2 In the `application.onPeerLookup()` callback, Introducer 1 calls `application.sendPeerRedirect()` to reply to Client 1 with a redirect message. The redirect message includes the known addresses for Client 2 and the public address for Introducer 2.

***Note:** To prevent a client from establishing a peer connection, Introducer 1 does not reply to the peer lookup request. See "Filter introduction requests" on page 282.*

3 Client 1 resends a peer lookup request to the address for Introducer 2.

4 Introducer 2 does the following to handle the peer lookup request:

a Replies to Client 1 with a redirect message containing the IP addresses for Client 2.

b Sends a forward message to Client 2 informing it that Client 1 wants to connect.

5 The clients do the following:

a Client 1 resends its peer lookup request to the addresses for Client 2 that it received from Introducer 2.

b The server-script calls `Client.introducePeer()` on the Client 2 object to send a responder hello message to the addresses for Client 1.

Configure the server to dispatch events to the script engine

By default, the server handles peer lookup events internally. To write a script that distributes introductions across multiple servers, configure the server to dispatch peer lookup events to the script engine. In the Application.xml file, set the `mode` attribute of `PeerLookupEvents` to `"All"`.

By default, the server handles a peer joining and leaving a group internally. To write a script that uses the server channel to bootstrap peers within a group, configure the server. Set the `mode` attribute of `JoinLeaveEvents` to `"All"`to dispatch these events to the script engine.

```
<Application>
...
<RTMFP>
    <PeerLookupEvents mode="All"/>
    <GroupControl>
        <JoinLeaveEvents mode="All"/>
    </GroupControl>
</RTMFP>
...
</Application>
```

The following are possible values of the `mode` attributes:

Value	Element	Description
`"None"`	`PeerLookupEvents` and `JoinLeaveEvents`	The default value. The server handles all peer lookup events. You cannot use Server-Side ActionScript to distribute peer introductions or filter peer introductions.
`"Partial"`	`PeerLookupEvents`	The server handles peer lookup events for clients connected to the same fmscore process. Handle other peer lookup events with Server-Side ActionScript.
`"All"`	`PeerLookupEvents` and `JoinLeaveEvents`	Handle all lookup events with Server-Side ActionScript.

Deploy servers in a robust server-only group

To deploy servers to act as distributed RTMFP introducers, use the following techniques:

- Use a serverless RTMFP NetConnection and a NetGroup to join a server-only group (in this case, the groupspec is private and not shared with clients).
- Add permanent group members at known addresses.
- Use the server channel to distribute peer bootstrapping across servers in the group.

Use a serverless RTMFP NetConnection to create a NetGroup

To distribute introductions, the servers must share state. All the servers must know about each other and they must know the peer ID and near and far IP addresses of all connected clients. The easiest way to share state is to connect the servers in a NetGroup (also called a *group*).

To distribute introductions, the server group should be robust. When the servers are joined in a NetGroup, if an introducer fails, all the peers connected to that introducer fail. To create a robust group, use serverless mode to create the RTMFP NetConnections between servers. In serverless mode, if any of the peer servers fail, the other servers stay connected to each other and are informed about the failed server leaving the group.

Use a serverless connection between servers so that the server-side "peer" cannot loose connectivity and its peer ID never changes. The group can use `GroupSpecifier.ipMemberUpdatesEnabled` to discover neighbors on the same subnet.

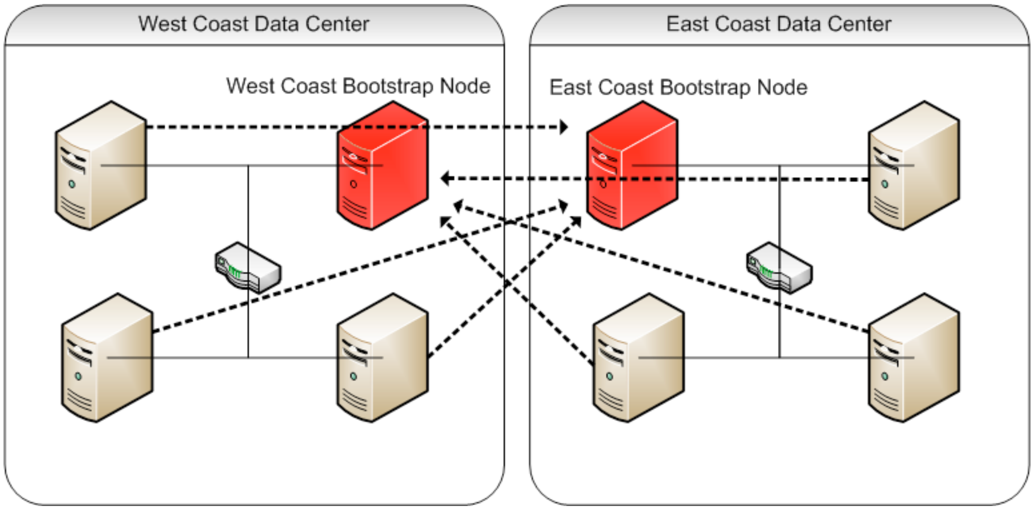
The following code creates a connection in serverless mode:

```
var nc:NetConnection = new NetConnection;
nc.connect("rtmfp:");
```

Add permanent members to a server-only group

Use this technique to join groups of peers across different subnets. Choose servers to act as permanent group members. For example, you can identify a known address for one server on each subnet, or for one server at each data center. Distribute the known address to the other servers using a configuration file. The server at the known address is a member of the group and opens peer-to-peer connections to other nodes that contact it over this known address. Once connected, this "bootstrap node" automatically informs its neighbors of other peers in the group. The built-in gossip functionality within groups creates a server-only group with full transitive connectivity.

Choose a server in each data center to act as a permanent group member. Assign these servers endpoint names and IP addresses. On the other servers, write application logic that reads the endpoint names and IP addresses from a configuration file. Use this information to add members directly.



Red servers are bootstrap nodes in server-only groups

In the previous diagram, red servers are bootstrap nodes. Solid lines are connections to permanent members. Dotted lines are peer connections created through the automatic gossip functionality of a group.

Use the following Server-Side ActionScript API to add permanent members to a server-only group:

- `NetConnection.rtmfpBindAddresses`

  An Array of Strings representing the specific address or addresses that the NetConnection binds locally when it opens its RTMFP protocol stack.

- `NetConnection.rtmfpEndpointName`

  The endpoint name for the local RTMFP protocol stack.

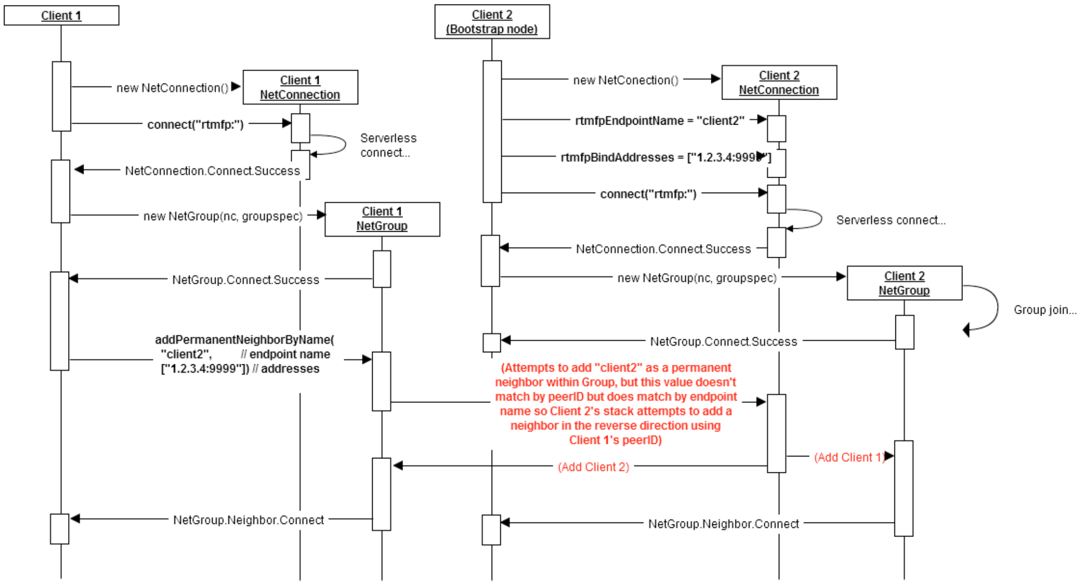
- `NetGroup.addPermanentNeighborByName()`

  Manually adds a neighbor by RTMFP endpoint name, instead of by peer ID. If a peer is disconnected from this neighbor, it automatically attempts to reconnect.

- `NetGroup.removePermanentNeighborByName()`

  Manually removes the "permanent" status for a neighbor by RTMFP endpoint name. Calling this method does not drop a connection. However, if either end chooses to drop the connection at a future time, the drop is allowed.

The following image describes the application flow for bootstrapping a server-only group:


Client 1
Client 2
(Bootstrap node)
new NetConnection()
Client 1
NetConnection
new NetConection()
Client 2
NetConnection
connect("rtmfp:")
rtmfpEndpointName = "client2"
Serverless
connect...
rtmfpBindAddresses = ["1.2.3.4:9999"]
NetConnection.Connect.Success
connect("rtmfp:")
new NetGroup(nc, groupspec)
Client 1
NetGroup
Serverless connect...
NetConnection.Connect.Success
NetGroup.Connect.Success
new NetGroup(nc, groupspec)
Client 2
NetGroup
Group join...
addPermanentNeighborByName(
"client2", // endpoint name
["1.2.3.4:9999"]) // addresses
NetGroup.Connect.Success
(Attempts to add "client2" as a permanent neighbor within Group, but this value doesn't match by peerID but does match by endpoint name so Client 2's stack attempts to add a neighbor in the reverse direction using Client 1's peerID)
(Add Client 1)
(Add Client 2)
NetGroup.Neighbor.Connect
NetGroup.Neighbor.Connect

***Note:** The calls in bold are the new Server-Side ActionScript APIs.*

This type of bootstrapping is unique because Client 1 initiates the interaction with Client 2 but doesn't know its peer ID. Client 1 knows only the IP address and the `rtmfpEndpointName`.

When Client 1 calls `addPermanentNeighborByName()` and passes the `rtmfpEndpointName` of Client 2, Client 2 retrieves the peer ID of Client 1. Client 2 uses that peer ID to add Client 1 as a neighbor in the reverse direction.

Once these two peers become neighbors, they both receive `NetGroup.Neighbor.Connect` events in Server-Side ActionScript. Their respective RTMFP stacks gossip to each other about other peers, which eventually results in a connected Group.

There isn't a security risk because the initiator must use the proper groupspec value to add a neighbor. You can distribute this value securely and make it unguessable. If Client 1 does not know the groupspec, the server ignores the request to add Client 2 as a permanent neighbor.

Unlike `NetGroup.addNeighbor()` or `NetGroup.addMemberHint()`, the `NetGroup.addPermanentNeighbor()` method causes the local RTMFP stack to automatically add this neighbor if it is removed. Permanent neighbors allow a group to maintain full connectedness across groups that have member nodes in separate subnets.

To determine whether the local node is a member of a connected Group, call `NetGroup.post()` from the bootstrap node on a set interval. Other nodes can listen for these messages to detect whether they are connected to the slice of the Group containing the bootstrap node. If they aren't receiving the messages, they could notify an admin or shutdown.

For detailed information about these APIs, see Server-Side ActionScript Language Reference.

Use the server channel to distribute peer bootstrapping across servers

Server-Side ActionScript provides the following techniques to bootstrap peers to a group:

- `GroupSpecifier.addBootstrapPeer(peerID)`

  Adds a peer ID to the group specifier before the peer joins the group.

- `NetGroup.addNeighbor(peerID)` and `NetGroup.addMemberHint(peerID)`

  Manually add a peer to a group. These methods require an external framework to distribute peer IDs.

- `GroupSpecifier.ipMulticastMemberUpdatesEnabled`

  Allows peers to find each other on a LAN.

- Use the server channel.

  This method does not require a LAN and does not require knowledge of peer IDs.

Use the following APIs to bootstrap a server to a group over the server channel:

- `Client.onGroupJoin(groupcontrol)`

  Called when a peer joins a group. By default, the server handles the join and leave events internally. To handle these events in a server-side script, configure the Application.xml file. See “Configure the server to dispatch events to the script engine” on page 274.

- `Client.onGroupLeave(groupspecDigest)`

  Called when a peer leaves a group.

- `GroupControl.addMemberHint(peerID)`

  Manually adds a record specifying that `peerID` is a member of the group. An immediate connection to this peer is attempted only if needed for the topology.

- `GroupControl.addNeighbor(peerID)`

  Manually adds a neighbor by immediately connecting directly to the specified peerID, which must already be in this group.

- `GroupControl.groupspecDigest`

  A digest of the canonical groupspec, which securely identifies the group the client has joined.

Use the GroupControl object to tell a peer to add a neighbor. In your application code, as you receive the `onGroupJoin()` and `onGroupLeave()` events, maintain a table of group names and peers in the group. Send a message to peers in the group telling them to add the new peer as a neighbor.

When a client joins a group with the `GroupSpecifier.serverChannelEnabled` flag set to true, a `Client.onGroupJoin()` callback event is sent to Server-Side ActionScript. The parameter to this method is a GroupControl object, which contains a digest of the canonical GroupSpecifier String for the group. In a script, this groupspecDigest String may be used as a key into a table that stores the list of connected Clients and the GroupControl object representing their membership in the group. When a Client joins a group, search in this table for other connected Clients who have joined the same group. If the search is successful, call the `groupControl.addNeighbor()` or `groupControl.addMemberHint()` methods to bootstrap the new client with peer connections to other neighbors within the group.

To maintain a mapping of groupspecDigests to a specific NetGroup object that joined a group, call the server-side `GroupSpecifier.encodeGroupspecDigest()` method. If you have a source GroupSpecifier, this method generates the groupspecDigest. Otherwise, the server must treat the groupspecDigest property in GroupControl objects as opaque strings. You cannot go from a digest to the starting groupspec String value.

The following server-side script uses the server channel bootstrapping API:

```
var groups = {};
Client.prototype.onGroupJoin = function(groupControl)
{
    groupControl["client"] = this; // Remember the associated Client.
    var groupControlArray = groups[groupControl.groupspecDigest];
    if (groupControlArray)
    {
        trace("Register Client in existing Group (by groupspec digest): " +
              groupControl.groupspecDigest +
              ", current Group size is: " +
              groupControlArray.length);

        // find a random member to bootstrap with
        r = Math.random();
        index = Math.floor(r * groupControlArray.length);

        var peerGroupControl = groupControlArray[index];
        groupControl.addNeighbor(peerGroupControl["client"].farID);
        groupControlArray.push(groupControl);
    }
    else
    {
        trace("Track client joining new Group (by groupspec digest): " +
groupControl.groupspecDigest);

        groupControlArray = [];
        groupControlArray.push(groupControl);
        groups[groupControl.groupspecDigest] = groupControlArray;
    }
}
```

For detailed information about these APIs, see Server-Side ActionScript Language Reference.

Distributed Introductions API

Use the distributed introductions Server-Side ActionScript API to control the peer introduction process and to create a shared peer registry.

Control the peer introduction process

- `Application.onPeerLookup()`

  Called when a client initiates a peer look-up. This event receives an object with the following properties:

Property	Data type	Description
`targetPeerID`	String	The peer ID of the target peer to which the initiating peer wants to connect.
`initiatorAddress`	String	The IP address of the peer initiating the request to connect. Send redirect information to this address.
`tag`	ByteArray	A value that uniquely identifies this lookup request.
`interfaceID`	Number	Identifies the RTMFP interface on which the request was received.

- `Application.sendPeerRedirect()`

  Call this method from the `application.onPeerLookup()` callback function to send the initiating peer an Array of addresses for the target peer.

- `Client.introducePeer()`

  Call on a target peer to open a connection with the peer that initiated the request to connect.

  For detailed information about these APIs, see Server-Side ActionScript Language Reference.

Create a peer registry

Use these APIs to create a peer registry. The peer registry lists the peer ID and IP addresses of every client that connects to an introducer. Every server acting as an introducer in a distributed environment shares this peer registry and uses it to locate peers when they receive a peer lookup. Write the application logic to create and share the peer registry.

- `Client.farAddress`

  The IP address and port number from which the server sees the client connection originate.

- `Client.nearAddress`

  The public address on the server to which the client connected.

- `Client.potentialNearAddresses`

  The public interfaces available for communication with the server.

- `Client.reportedAddresses`

  The addresses at which a client can receive RTMFP traffic. The client can update this value multiple times over the lifetime of its RTMFP connection to the server. This value can contain IP addresses and ports that the client has opened behind its NAT. If so, use these addresses and ports for interacting with other peers behind a common NAT.

- `Client.onFarAddressChange()`

  Called when a client's `farAddress` has changed. For example, an address changes when a client transitions from a LAN to a wireless connection.

- `Client.onReportedAddressChange()`

  Called when a client reports new addresses.

For detailed information about these APIs, see Server-Side ActionScript Language Reference.

Example: Distribute introductions across servers

This example explains at a high level how to distribute introductions across servers.

To distribute introductions across a group of servers, every server must know the peer ID and IP addresses for every connected client. Track these values in a peer registry. In the following example, assume the following functions exist to control the peer registry:

- `addClient(peerId, clientAddresses)`

  When a client connects to the server, add the peer ID and address to the peer registry.

- `removeClient(peerId)`

  When a client disconnects from the server, delete it from the peer registry.

- `updateClient(peerId, clientAddresses)`

  When a far address in the reported addresses changes, update the reported addresses in the peer registry.

- `getClientIfLocal(peerId)`

  Returns the Client object if the client with the given `peerId` is connected to the current server, otherwise returns null.

- `getRemoteClientAddrs(peerId)`

  Returns the client addresses from the peer registry for the given peer ID. If the client is not found, returns null.

To create the peer registry of client addresses, create a helper function that returns all the client addresses:

```
function getClientRedirectAddrs(client)
{
    var redirectAddresses = client.reportedAddresses.slice(0);
    redirectAddresses.push(client.farAddress);
    redirectAddresses.concat(client.potentialNearAddresses);
    return redirectAddresses;
}
```

When a client connects over RTMFP, add it to the peer registry:

```
application.onConnect = function(client) {
    if (client.protocol == "rtmfp")
    {
        addClient(client.farID, getClientRedirectAddrs(client));
    client.onFarAddressChange = function() {
        updateClient(this.farId, getClientRedirectAddrs(client));
    };
    client.onReportedAddressesChange = function() {
        updateClient(this.farId, getClientRedirectAddrs(client));
    };
    }
}

application.onDisconnect = function(client) {
    removeClient(client.farId);
}
```

Send redirect messages and responder hello messages in the `application.onPeerLookup()` callback function:

```
application.onPeerLookup = function(event) {
    // Check whether the target peer is local
    // (connected to the same server as the initiating peer).
    var targetPeer = getClientIfLocal(event.targetPeerID);
    if (targetPeer)
    {
    // Send the addresses of the target peer to the initiating peer.
    application.sendPeerRedirect(getClientRedirectAddrs(targetPeer), event);
    // Note: getClientIfLocal() must return the actual client object,
    // because that allows us to call introducePeer()
    // The target peer sends a responder hello to the initiating peer.
    targetPeer.introducePeer(event.initiatorAddress, event.tag);
    }
    else
    {
    // Client is not connected locally.
    targetPeerAddrs = getRemoteClientAddrs(event.targetPeerID);
    if (!targetPeerAddrs)
    {
        // Returning without calling application.sendPeerRedirect()
        // prevents the peer-to-peer connection.
        return;
    }
    // Send the addresses of the target peer to the initiating peer.
    application.sendPeerRedirect(targetPeerAddrs, event);
    }
}
```

Consider the following scenario in which the previous code is used on two Adobe Media Server introducers: Server A and Server B. Client 1 is connected to Server A and Client 2 is connected to Server B. Client 1 wants to connect to Client 2.

- Client 1 sends a peer lookup request to Server A.
- The peer lookup request calls the callback function `application.onPeerLookup()`in the server-side script.
- Because Client 2 is not connected to Server A, the `else` branch of code runs that does not have the local Client object. Server A calls `getRemoteClientAddrs()` to get the target addresses out of the shared store. These addresses include the addresses of Server B (to which Client 2 is directly connected). The script calls `application.sendPeerRedirect()`to pass the target addresses to Client 1.
- When the redirect reaches Client 1, Flash Player sends a peer lookup request to all the target addresses, including the address of Server B.
- When the request arrives at Server B, the `if` branch of the code runs that has the local target Client object. The code calls `sendPeerRedirect()` again to send information to Client 1. But now that the script has the Client object representing Client 2 at Server B, it can also call `introducePeer()` on Client 2. This call tells Client 2 to start sending packets to Client 1. (Client 1 has been trying to send packets toward Client 2 since it got the first `sendPeerRedirect()` address set from Server A).
- When both clients are sending traffic at each other, the necessary NAT hole-punching has happened. Both clients can see each other's traffic and establish a peer-to-peer connection. (Assuming that both clients are behind well-behaved NATs.)

Filter introduction requests

Flash Media Server 4.5

You can also use the distributed introduction API to filter introduction requests. The following sample script creates a map of peer IDs and client addresses. When the server receives a peer lookup request, the `application.onPeerLookup()` callback checks to see whether the target peer is in the map. If it isn't, the server denies the lookup request.

```
// A registry mapping peer ID values to Client objects
// for this application instance.
var peerIDToClientMap = {};

// On client connect, add RTMFP clients to the peer ID registry.
application.onConnect = function(client) {
    if (client.protocol == "rtmfp")
        peerIDToClientMap[client.farID] = client;
}

//
application.onPeerLookup = function(event) {
    var targetPeer = peerIDToClientMap[event.targetPeerID];
    if(!targetPeer){
        // Returning without calling application.sendPeerRedirect()
        // prevents the peer-to-peer connection.
        return;
    }
}
```

For detailed information about these APIs, see Server-Side ActionScript Language Reference.

Use the Administration API to monitor distributed introductions

Use the following properties returned in a call to the `getServerStats()` Administration API method to monitor distributed introductions:

- `rtmfp_forwards`
- `rtmfp_lookups`
- `rtmfp_lookups_denied`
- `rtmfp_redirects`

There are many ways to use these statistics. The number of lookups minus the number of redirects yields the number of lookup queries that were ignored or denied. The number of redirects minus the number of forwards yields the number of lookups in which the initiating client was redirected to a different Adobe Media Server node to connect with the target peer. The `rtmfp_lookups_denied` statistic is a counter that tracks the number of lookups that were explicitly denied (for example, because they were invalid). It is incremented when `application.denyPeerLookup()` is called. If there are too many denials, it could indicate a DOS attack.

Use the following properties returned in a call to the `getServerStats()` Administration API method to monitor peers joining and leaving a group:

- `group_join`
- `group_leave`

Chapter 7: Developing social applications

About social applications

***Important:** Adobe Media Server Standard does not support server-side scripting and therefore does not support social applications.*

In addition to streaming video applications, Adobe Media Server Extended, Adobe Media Server Professional, and Adobe Media Server Starter can host social and other real-time communication applications. Users can capture live audio and video, upload them to the server, record them, and share them with others. These server editions also provide access to remote shared objects that synchronize data between many users, and so is ideal for developing online games.

You can use Server-Side ActionScript to connect to other systems, including Java 2 Enterprise servers, web services, and Microsoft .NET servers. This connectivity allows applications to take advantage of services such as database authentication, real-time updates from web services, and e-mail.

About shared objects

Use shared objects to synchronize users and store data. Shared objects can do anything from holding the position of pieces on a game board to broadcasting chat text messages. Shared objects let you keep track of what users are doing in real time.

Create and use *remote shared objects*, which share data between multiple client applications. When one user makes a change that updates the shared object on the server, the shared object sends the change to all other users. The remote shared object acts as a hub to synchronize many users. In the section “SharedBall example” on page 285, when any user moves the ball, all users see it move.

***Note:** Adobe Media Server Standard does not support remote shared objects.*

All editions of the server support *local shared objects*, which are similar to browser cookies. Local shared objects are stored on the client computer and don’t require a server.

Shared objects, whether local or remote, can also be *temporary* or *persistent*:

- A temporary shared object is created by a server-side script or by a client connecting to the shared object. When the last client disconnects and the server-side script is no longer using the shared object, it is deleted.
- Persistent shared objects retain data after all clients disconnect and even after the application instance stops running. Persistent shared objects are available on the server for the next time the application instance starts. They maintain state between application sessions. Persistent objects are stored in files on the server or client.

  Persistent local shared objects To create persistent local shared objects, call the client-side `SharedObject.getLocal()` method. Persistent local shared objects have the extension .sol. You can specify a storage directory for the object by passing a value for the `localPath` parameter of the `SharedObject.getLocal()` command. By specifying a partial path for the location of a locally persistent remote shared object, you can let several applications from the same domain access the same shared objects.

  Remotely persistent shared objects To create remote shared objects that are persistent on the server, pass a value of `true` for the `persistence` parameter in the client-side `SharedObject.getRemote()` method or in the server-side `SharedObject.get()` method. These shared objects are named with the extension .fso and are stored on the server in a subdirectory of the application that created the shared object. Adobe Media Server creates these directories automatically; you don't have to create a directory for each instance name.

Remotely and locally persistent shared objects You create remote shared objects that are persistent on the client and the server by passing a local path for the persistence parameter in your client-side `SharedObject.getRemote()` command. The locally persistent shared object is named with the extension .sor and is stored on the client in the specified path. The remotely persistent .fso file is stored on the server in a subdirectory of the application that created the shared object.

Remote shared objects

Before you create a remote shared object, create a NetConnection object and connect to the server. Once you have the connection, use the methods in the SharedObject class to create and update the remote shared object. The general sequence of steps for using a remote shared object is outlined below:

1 Create a NetConnection object and connect to the server:

```
nc = new NetConnection();
nc.connect("rtmp://localhost/SharedBall");
```

This is the simplest way to connect to the server. In a real application, you would add event listeners on the NetConnection object and define event handler methods. For more information, see "SharedBall example" on page 285.

2 Create the remote shared object. When the connection is successful, call `SharedObject.getRemote()` to create a remote shared object on the server:

```
so = SharedObject.getRemote("ballPosition", nc.uri, false);
```

The first parameter is the name of the remote shared object. The second is the URI of the application you are connecting to and must be identical to the URI used in the `NetConnection.connect()` method. The easiest way to specify it is with the `nc.uri` property. The third parameter specifies whether the remote shared object is persistent. In this case, `false` is used to make the shared object temporary.

3 Connect to the remote shared object. Once the shared object is created, connect the client to the shared object using the NetConnection object you just created:

```
so.connect(nc);
```

You also need to add an event listener for `sync` events dispatched by the shared object:

```
so.addEventListener(SyncEvent.SYNC, syncHandler);
```

4 Synchronize the remote shared object with clients. Synchronizing the remote shared object requires two steps. First, when an individual client makes a change or sets a data value, you need to update the remote shared object. Next, update all other clients from the remote shared object.

a To update the remote shared object when a client makes a change, use `setProperty()`:

```
so.setProperty("x", sharedBall.x);
```

You must use `setProperty()` to update values in the shared object. The remote shared object has a `data` property that contains attributes and values. However, in ActionScript 3.0, you cannot write values directly to it, as in:

```
so.data.x = sharedBall.x; // you can't do this
```

b When the shared object is updated, it dispatches a `sync` event. Synchronize the change to the remaining clients by reading the value of the shared object's `data` property:

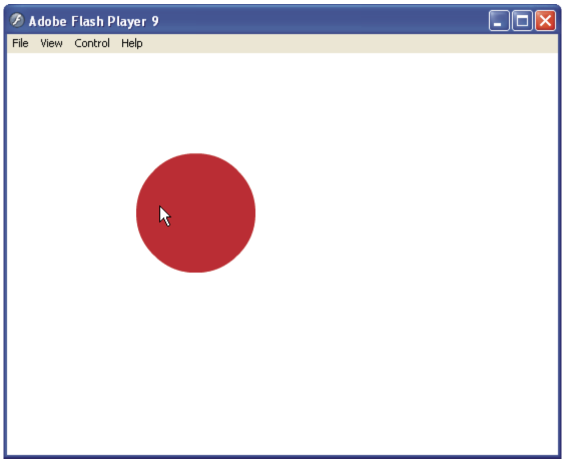
```
sharedBall.x = so.data.x;
```

This is usually done in a `sync` event handler, as shown in "SharedBall example" on page 285.

SharedBall example

The SharedBall sample creates a temporary remote shared object. It's similar to a multiplayer game. When one user moves the ball, it moves for all other users.

Note: *Use the SharedBall sample files (SharedBall.fla, SharedBall.as, and SharedBall.swf) in the documentation/samples/SharedBall directory in the Adobe Media Server root install directory.*



The SharedBall application running in Flash Player

Run the application

1 Register the application with your server by creating an application directory named SharedBall:

RootInstall/applications/SharedBall

2 Open the SharedBall samples files from the documentation/samples/SharedBall directory in the Adobe Media Server root install directory.

3 Open SharedBall.swf in a web browser.

4 Open a second instance of SharedBall.swf in a second browser window.

5 Move the ball in one window and watch it move in the other.

Design the user interface

1 In Flash, choose File > New > Flash File (ActionScript 3.0) and click OK.

2 From the toolbox, select the Rectangle tool. Drag to the lower-right corner, then select the Oval tool.

3 Draw a circle on the Stage. Give it any fill color you like.

4 Double-click the circle and choose Modify > Convert to Symbol.

5 In the Convert to Symbol dialog box, name the symbol **ball**, check that Movie Clip is selected, and click OK.

6 Select the ball symbol on the Stage and in the Property Inspector (Window > Properties) give it the instance name **sharedBall**.

7 Save the file as SharedBall.fla.

Write the client-side code

Be sure to look at the SharedBall.as sample file. These steps present only highlights.

1 In Flash Professional, create a new ActionScript file.

2 Create the class, extending MovieClip:

```
public class SharedBall extends MovieClip {...}
```

The class must extend MovieClip, because the sharedBall symbol in the FLA file is a Movie Clip symbol.

3 Create the constructor, in which you add event listeners and connect to the server:

```
    public function SharedBall()
    {
        nc = new NetConnection();
        addEventListeners();
        nc.connect("rtmp://localhost/SharedBall");
    }
```

4 Add event listeners for `netStatus`, `mouseDown`, `mouseUp`, and `mouseMove` events:

```
    private function addEventListeners() {
        nc.addEventListener(NetStatusEvent.NET_STATUS, netStatusHandler);
        // sharedBall is defined in the FLA file
        sharedBall.addEventListener(MouseEvent.MOUSE_DOWN, pickup);
        sharedBall.addEventListener(MouseEvent.MOUSE_UP, place);
        sharedBall.addEventListener(MouseEvent.MOUSE_MOVE, moveIt);
    }
```

5 In your `netStatus` handler, create a remote shared object when a connection is successful. (You'll also want to create error handlers for rejected and failed connections, shown in the sample AS file.) Connect to the shared object and add a `sync` event listener:

```
switch (event.info.code)
{
    case "NetConnection.Connect.Success":
        trace("Congratulations! you're connected");
            so = SharedObject.getRemote("ballPosition", nc.uri, false);
            so.connect(nc);
            so.addEventListener(SyncEvent.SYNC, syncHandler);
            break;
            ...
}
```

6 As a user moves the mouse, use `setProperty()` to set the changing ball location in the remote shared object:

```
function moveIt( event:MouseEvent ):void {
    if( so != null )
    {
    so.setProperty("x", sharedBall.x);
    so.setProperty("y", sharedBall.y);
    }
}
```

When the remote shared object is updated, it dispatches a `sync` event.

7 Write a `sync` event handler that updates all clients with the new ball position:

```
private function syncHandler(event:SyncEvent):void {
    sharedBall.x = so.data.x;
    sharedBall.y = so.data.y;
}
```

You can read the value of `so.data`, even though you can't write to it.

Broadcast messages to many users

A remote shared object allows either a client or server to send a message using `SharedObject.send()` to all clients connected to the shared object. The `send()` method can be used for text chat applications, for example, where all users subscribed to your shared object receive your message.

When you use `SharedObject.send()`, you, as broadcaster, also receive a copy of the message.

1 Write a method that `SharedObject.send()` will call:

```
private function doSomething(msg:String):void {
    trace("Here's the message: " + msg);
}
```

2 Call `send()` to broadcast the message:

```
so = SharedObject.getRemote("position", nc.uri, false);
so.connect(nc);
so.send("doSomething", msg);
```

Chapter 8: Securing applications

You can configure the server and your application to allow and deny access to assets such as streams and shared objects. You can also authenticate clients that connect to an application.

For more information about securing applications, see the “Developer Guidelines” section in the Adobe DevNet article, Hardening Guide for Adobe Media Server. The guidelines include modifying the Application.xml file, writing Server-Side ActionScript, securing communication with the server, and performing authentication.

Graeme Bull has several video tutorials about security on his site fmsguru.com:

•Taking advantage of the Application.xml file settings to help protect your server

- Using the client Agent and Referrer to help protect your server
- Using the client IP to help protect your server
- Using server side client permissions to help protect your server
- Using writeAccess to help protect your server

Allow or deny access to assets

About access control

When users access the server, by default, they have full access to all streams and shared objects. However, you can use Server-Side ActionScript to create a dynamic access control list (ACL) for shared objects and streams. You can control who has access to create, read, or update shared objects or streams.

When a client connects to the server, the server-side script (main.asc or *yourApplicationName*.asc) is passed a Client object. Each Client object has `readAccess` and `writeAccess` properties. You can use these properties to control access for each connection.

Implement dynamic access control

The `Client.readAccess` and `Client.writeAccess` properties take string values. These values can contain multiple strings separated by semicolons, like this:

```
client.readAccess = "appStreams;/appSO/";
client.writeAccess = "appStreams/public/;appSO/public/";
```

By default, `readAccess` and `writeAccess` are set to `/`, which means the client can access every stream and shared object on the server.

Allow access to streams

❖ In main.asc, add an `onConnect()` function that specifies a directory name on the server in your main.asc file:

```
application.onConnect = function(client, name) {
    // give this new client the same name as passed in
    client.name = name;

    // give write access
    client.writeAccess = "appStreams/public/";

    // accept the new client's connection
    application.acceptConnection(client);
}
```

This main.asc file grants access to all URIs that start with `appStreams/public`.

Deny access to streams

❖ In main.asc, add an `onConnect()` function that specifies a null value for `client.writeAccess`:

```
application.onConnect = function(client, name) {
    ...
    // deny write access to the server
    client.writeAccess = "";
}
```

Define access to shared objects

❖ In main.asc, add an `onConnect()` function that specifies shared object names, using the same URI naming conventions:

```
application.onConnect = function(client, name) {
    ...
    client.writeAccess = "appSO/public/";
}
```

This gives the client write access to all shared objects whose URIs begin with `appSO/public/`.

Authenticate clients

Use properties of the Client object

When a client connects to an application, the server creates a Client object that contains information about the client and passes it to the `application.onConnect()` handler in Server-Side ActionScript. You can write server-side code to access the properties of the Client object and use the values to verify the validity of the connecting client:

```
application.onConnect = function( pClient ) {
    for (var i in pClient) {
        trace( "key: " + i + ", value: " + pClient[i] );
    }
}
```

Check the client's IP address

❖ In main.asc, check the value of `client.ip` and, if needed, reject the client's connection to the application:

```
if (client.ip.indexOf("60.120") !=0) {
    application.rejectConnection(client, {"Access Denied"} );
}
```

Check an originating URL

❖ In main.asc, check the value of `client.referrer` against a list of URLs that should be denied access. Make sure that SWF files that are connecting to your application are coming from a location you expect. If you find a match, reject the client's connection:

```
referrerList = {};
referrerList["http://www.example.com"] = true;
referrerList["http://www.abc.com"] = true;

if (!referrerList[client.referrer]) {
    application.rejectConnection(client, {"Access Denied"} );
}
```

Use a unique key

1 In client-side ActionScript, create a unique key, as in the following code, which concatenates the local computer time with a random number:

```
var keyDate = String(new Date().getTime());
var keyNum = String(Math.random());
var uniqueKey = keyDate + keyNum;
```

2 Send the key to the server in the connection request:

```
nc.connect("rtmp://www.example.com/someApplication", uniqueKey);
```

3 The following code in the main.asc file looks for the unique key in the connection request. If the key is missing or has already been used, the connection is rejected. This way, if a connection is replayed by an imposter, the replay attempt fails.

```
clientKeyList = new Object();   // holds the list of clients by key

application.onConnect = function( pClient, uniqueKey ) {
    if ( uniqueKey != undefined ) {    // require a unique key with connection request
        if ( clientKeyList[uniqueKey] == undefined ) {   // first time -- allow connection
            pClient.uniqueKey = uniqueKey;
            clientKeyList[uniqueKey] = pClient;
            this.acceptConnection(pClient);
        } else {
            trace( "Connection rejected" );
            this.rejectConnection(pClient);
        }
    }
}

application.onDisconnect = function( pClient ) {
    delete clientKeyList[pClient.uniqueKey];
}
```

Use an Access plug-in

An Access plug-in intercepts incoming requests before passing them to the application. You can program an Access plug-in to use any form of authentication.

More Help topics

"Developing an Access plug-in" on page 301

Use Flash Player version

You can protect your content from clients that aren't running in Flash Player, based on the user agent string received from the connection. The user agent string identifies the platform and Flash Player version, for example:

```
WIN 8,0,0,0
MAC 9,0,45,0
```

There are two ways to access these strings:

Virtual keys Configure the server to remap the stream based on the Flash Player client.

Client.agent Challenge the connection using Server-Side ActionScript:

```
application.onConnect = function( pClient ) {
    var platform     = pClient.agent.split(" ");
    var versionMajor = platform[1].split(",")[0];
    var versionMinor = platform[1].split(",")[1];
    var versionBuild = platform[1].split(",")[2];
}

// output example
// Client.agent:  WIN 9,0,45,0
// platform[0]:   "WIN"
// versionMajor:  9
// versionMinor:  0
// versionBuild:  45
```

Verify connecting SWF files

You can configure the server to verify the authenticity of client SWF files before allowing them to connect to an application. Verifying SWF files prevents someone from creating their own SWF files that attempt to stream your resources. SWF verification is supported in Flash Player 9 Update 3 and later.

Allow or deny connections from specific domains

If you know the domains from which the legitimate clients will be connecting, you can whitelist those domains. Conversely, you can blacklist known bad domains.

You can enter a static list of domain names in the Adaptor.xml file.

You can also maintain these lists in your own server-side code and files. In the following example, a file named *bannedIPList.txt* contains a list of excluded IP addresses, which can be edited on the fly:

```
// bannedIPList.txt file contents:
// 192.168.0.1
// 128.493.33.0

function getBannedIPList() {
    var bannedIPFile = new File ("bannedIPList.txt") ;
    bannedIPFile.open("text","read");

    application.bannedIPList = bannedIPFile.readAll();

    bannedIPFile.close();
    delete bannedIPFile;
}

application.onConnect = function(pClient) {
    var isIPOK = true;
    getBannedIPList();
    for (var index=0; index<this.bannedIPList.length; index++) {
        var currentIP = this.bannedIPList[index];
        if (pClient.ip == currentIP) {
            isIPOK = false;
            trace("ip was rejected");
            break;
        }
    }

    if (isIPOK) {
        this.acceptConnection(pClient);
    } else {
        this.rejectConnection(pClient);
    }
}
```

In addition, you can create server-side code to check if requests are coming in too quickly from a particular domain:

```
application.VERIFY_TIMEOUT_VALUE = 2000;

Client.prototype.verifyTimeOut = function() {
    trace (">>>> Closing Connection")
    clearInterval(this.$verifyTimeOut);
    application.disconnect(this);
}

function VerifyClientHandler(pClient) {
    this.onResult = function (pClientRet) {
        // if the client returns the correct key, then clear timer
        if (pClientRet.key == pClient.verifyKey.key) {
            trace("Connection Passed");
            clearInterval(pClient.$verifyTimeOut);
        }
    }
}

application.onConnect = function(pClient) {
    this.acceptConnection(pClient);

    // create a random key and package within an Object
    pClient.verifyKey = ({key: Math.random()});

    // send the key to the client
    pClient.call("verifyClient",
        new VerifyClientHandler(pClient),
        pClient.verifyKey);

    // set a wait timer
    pClient.$verifyTimeOut = setInterval(pClient,
                                      $verifyTimeOut,
                                      this.VERIFY_TIMEOUT_VALUE,
                                      pClient);
}

application.onDisconnect = function(pClient) {
    clearInterval(pClient.$verifyTimeOut);
}
```

Authenticate users

Authenticate using an external resource

For a limited audience, it is feasible to request credentials (login and password) and challenge them using an external resource, such as a database, LDAP server, or other access-granting service.

1 The SWF supplies the user credentials in the connection request.

The client provides a token or username/password using client-side ActionScript:

```
var sUsername = "someUsername";
var sPassword = "somePassword";

nc.connect("rtmp://server/secure1/", sUsername, sPassword);
```

2 Adobe Media Server validates the credentials against a third-party system.

You can use the following classes to make calls from Server-Side ActionScript to external sources: WebService, LoadVars, XML classes, NetServices (connects to a Flash Remoting gateway). For more information about Flash Remoting, see www.adobe.com/go/learn_fms_flashremoting_en.

Chapter 9: Developing Plug-ins

Adobe® Media Server provides a plug-in architecture written in C++ that lets you extend the functionality of the server. Use the Access, Authorization, and File plug-ins to build unique deployments with expanded access, authorization, and file management solutions.

For example, you can use the plug-ins to accept, reject, or redirect clients before they reach your application-level code (Access plug-in), control access to streams and server events (Authorization plug-in), and create a file I/O mechanism (File plug-in).

For detailed information about each plug-in's API, see *Flash Media Server plug-in API Reference*.

For information about the Adobe Media Server Plug-in architecture, see Using plug-ins to configure and optimize Flash Media Server 4.5 by Denis Bulichenko.

What's new with plug-ins in Flash Media Server 4.5.1

Adobe Media Server 4.5.1 includes the following updates to the plug-ins:

- Protected RTMP streaming can be configured through the Authorization plug-in. See "Use the Authorization plug-in to configure protected RTMP" on page 32.

What's new with plug-ins in Flash Media Server 4.5

Flash Media Server 4.5 includes the following updates to the plug-ins:

- The HTTP Dynamic Streaming and HTTP Live Streaming file operations are routed through the File plug-in. See "Use the File plug-in to manage content for live HTTP streaming" on page 333.

Versioning, upgrading, and server editions

Server edition support

Flash Media Enterprise Server, Flash Media Interactive Server, and Flash Media Development Server support the following plug-ins:

- Access plug-in
- Authorization plug-in
- File plug-in

Flash Media Streaming Server supports the following plug-in:

- Access plug-in

About versioning

The Authorization plug-in and the File plug-in have versioning and accompanying create and destroy methods. The Access plug-in does not have versioning. The server detects the plug-in version number for the File and Authorization plug-ins based on the value returned by the create and destroy methods. To use the methods of a certain server version, use the correct create or destroy method and return the correct version number. For example, to use Flash Media Server 4.0 features with the Authorization plug-in, call `FmsCreateAuthAdaptor3()` and return 2.0.

***Note:** To limit a plug-in to Adobe Media Server 3.0 features, return version 0.0 from a call to* `FmsCreateXXXAdaptor2()`.

For more information on the versioning APIs, see *Adobe Media Server Plug-in API Reference*.

The File plug-in has the following versioning:

Server version	Plug-in version	Create and Destroy method

Adobe Media Server 3.0 and earlier	0.0	`FmsCreateFileAdaptor()`
		`FmsDestroyFileAdaptor()`

Adobe Media Server 3.5	1.0	`FmsCreateFileAdaptor2()`
		`FmsDestroyFileAdaptor2()`

Adobe Media Server 4.0	1.0	`FmsCreateFileAdaptor2()`
		`FmsDestroyFileAdaptor2()`

The Authorization plug-in has the following versioning:

Server version	Plug-in version	Create and Destroy method

Adobe Media Server 3.0 and earlier	0.0	`FmsCreateAuthAdaptor()`
		`FmsDestroyAuthAdaptor()`

Adobe Media Server 3.5	1.0	`FmsCreateAuthAdaptor2()`
		`FmsDestroyAuthAdaptor2()`

Adobe Media Server 4.0	2.0	`FmsCreateAuthAdaptor3()`
		`FmsDestroyAuthAdaptor3()`

Upgrading a plug-in

All Flash Media Server 3.5.x plug-ins work on 32-bit Flash Media Server 4. To use new features in Flash Media Server 4, including new APIs, upgrade your plug-in. If you do not intend to use the new features, you can continue to use your existing plug-in code.

To use 32-bit plug-ins on a 64-bit system, recompile the plug-ins with a 64-bit Flash Media Server library.

Recompile 32-bit plug-ins for a 32-bit server

1. Recompile the plug-in with Microsoft® Visual C++ 2003, Microsoft® Visual C++ 2005, or Microsoft Visual C++ 2008. On Linux, recompile the plug-in with GNU Compiler Collection 4.x, for example, GCC 4.1 which installs with CentOS.
2. For the File plug-in, use the `FmsCreateFileAdaptor2()` and `FmsDestroyFileAdaptor2()` methods and return version 1.0.

3 For the Authorization plug-in, use the `FmsCreateAuthAdaptor3()` and `FmsDestroyAuthAdaptor3()` methods and return version 2.0.

See the *Adobe Media Server plug-in API Reference*.

Recompile 32-bit plug-ins for a 64-bit server

- When porting 32-bit plug-in code to 64-bit code, be aware that the data types of some fields have changed.
- Use the include files and libraries in the *rootinstall*/samples/plugins/include folder to compile the plug-ins.
- On Linux, there are two makefiles: MakeFile.AuthModule64 and MakeFile.AuthModule. To compile for 64-bit, use MakeFile.AuthModule64.
- On Windows, install the 64-bit compiler for Visual C++ in Visual Studio 2008 or later. Choose Build > Configuration Manager and select Release x64. Build the 64-bit installation.

Working with plug-ins

Workflow for developing and deploying a plug-in

***Note:** For information about upgrading a plug-in, see "Upgrading a plug-in" on page 296.*

1 Modify the sample plug-in to meet your organization's needs or write your own plug-in.

See "Sample files" on page 297.

2 Build and compile the plug-in for your platform.

See "Compile a plug-in" on page 299.

3 Deploy the plug-in.

See "Deploy a plug-in" on page 299.

More Help topics

Flash Media Server Plug-In API Reference

Sample files

The server includes sample files for each plug-in. Use these samples to learn how to use the API or use them as starting points for your own plug-ins.

***Note:** To build the default plug-ins, use the make files. Use the make command with the `-f` option to run the make file.*

Access plug-in

The following sample files are installed to *rootinstall*\samples\plugins\access.

Filename	Description
adaptor.cpp sample.cpp adaptor.h	Sample Access plug-in C++ files and header file.
AccessModuleSample.sln AccessModuleSample.vcproj	Work files used with Microsoft Visual C++ for building an Access plug-in in a Windows environment. You can modify the files' values to reflect the practices in your environment.
Makefile.access	Use this file to build the default Access plug-in on Linux.
StdAfx.h StdAfx.cpp	Sample includes file declaration files.

Authorization plug-in

The following sample files are installed to *rootinstall*\samples\plugins\auth.

Filename	Description
AuthModule.cpp	Sample Authorization plug-in C++ file.
AuthModule.sln AuthModule.vcproj	Work files used with Microsoft Visual C++ for building an Authorization plug-in in a Windows environment. You can modify the files' values to reflect the practices in your environment.
Makefile.AuthModule	Use this file to build the default Authorization plug-in on Linux.
StdAfx.h StdAfx.cpp	Sample includes file declaration files.

File plug-in

The following sample files are installed to *rootinstall*\samples\plugins\file.

Filename	Description
SimpleFileAdaptor.cpp SimpleFileAdaptor.h	Sample File plug-in C++ file and header file.
FileModule.h	A header file that defines create and destroy functions for the plug-in.
FileUtil.cpp	Utility functions for working with files and directories.
FileModule.sln FileModule.vcproj	Work files used with Microsoft Visual C++ for building a File plug-in in a Windows environment. You can modify the files' values to reflect the practices in your environment.
Makefile.FileModule	Use this file to build the default File plug-in on Linux.
StdAfx.h StdAfx.cpp	Sample includes file declaration files.

Header files

The following header files define the plug-in API and are installed to *rootinstall*\samples\plugins\include: FmsAdaptor.h, FmsAuthActions.h, FmsAuthAdaptor.h, FmsAuthEvents.h, FmsFileAdaptor.h, FmsMedia.h, and IFCAccessAdaptor.h.

Compile a plug-in

Use Microsoft® Visual Studio .NET 2003, Microsoft Visual C++ 2005, or Microsoft Visual Studio .NET 2008 to compile plug-ins in Windows. Use GNU Compiler Collection 3.4.x to compile plug-ins in Linux.

***Note:** The Authorization plug-in and the Access plug-in have compilation warnings that are harmless. Ignore the warnings as you compile.*

Build the plug-in as a Release. If you build the plug-in as Debug the modules don't load with the server process.

Plug-ins are implemented as shared library (DLL) files in Windows systems and as shared object (SO) files in Linux systems. The filenames are: libconnect.dll/libconenct.so, AuthModule.dll/AuthModule.so, and FileModule.dll/FileModule.so.

To build on a 64-bit version of a linux GCC compiler, use the `-m32` switch and `-B` option for GCC. Point the `-B` switch to the /usr/lib32 libraries.

An Access plug-in must have the name libconnect.dll/libconnect.so. Authorization plug-ins and File plug-ins can have any name.

Deploy a plug-in

Once deployed, a plug-in plugs into a server process. The plug-in is loaded during the process start-up and unloaded during process shutdown. For example, when a vhost is restarted a new core process starts up and loads a copy of an Authorization plug-in. When the vhost stops, the core process unloads the Authorization plug-in.

1 Do one of the following to stop the server:

- In Windows, choose Start > Control Panel > Administrative Tools > Services. Select Adobe Media Server (FMS) from the Services list and click Stop.
- In Linux, open a shell window and go to the directory where the server is installed: **cd /opt/adobe/ams**. Enter the following: **./server stop**.

2 Copy the compiled plug-in DLL or SO files to one of the following folders:

- *rootinstall*/modules/access
- *rootinstall*/modules/auth
- *rootinstall*/modules/fileio

***Note:** Do not change the folder names. If you're deploying multiple Authorization plug-ins, copy all the plug-ins into the /auth folder.*

3 Do one of the following to start the server:

- In Windows, choose Start > Control Panel > Administrative Tools > Services. Select Adobe Media Server (FMS) from the Services list and click Start.
- In Linux, open a shell window and go to the directory where the server is installed: **cd /opt/adobe/ams**. Enter the following: **./server start**.

General development tasks

Sending data from a plug-in to a log file

Call the `log()` function of the IFmsServerContext class in a plug-in to send custom messages to log files from the Authorization and File plug-ins. Plug-in log files are located in the *RootInstall*/logs directory and are named fileio.NN.log and authMessage.NN.log, by default.

***Note:** In addition to client connections, the Authorization plug-in logs the connections made when a server-side `Stream.play()` method is called. These connections are distinguished by the IP address 127.0.0.1 in the access.log files and in the authEvent.log files.*

The `log()` function has an argument that specifies whether the log message should also be logged to the system log (Windows Event Viewer or Linux syslog). The default value is false. Excessive logging to the system log can cause performance problems.

Edit the `Logging` section of the Server.xml configuration file (located in *RootInstall*/conf) to enable or disable each log file. Edit the Logger.xml file (located in *RootInstall*/conf) to add or change the configuration information, including the location and name of the log files. For detailed information, see the comments in the XML files.

The Access plug-in is hosted by a process that closes standard out and standard error, therefore plug-in developers cannot rely on these to log messages. Instead, for the Access plug-in, log messages explicitly to a file. The sample Access plug-in demonstrates a workaround. Search for the line `FILE* logFile = fopen("SampleAdaptor.log", "a");` in the *RootInstall*/samples/plugins/access/adaptor.cpp file.

More Help topics

Adobe Media Server Plug-In API Reference

Retrieving data from a configuration file

Call the `getConfig()` function of the IFmsServerContext class in an Authorization plug-in or a File plug-in to retrieve custom data stored in the Server.xml configuration file. You can analyze this data in the plug-in to activate different functionality in the plug-in, depending on the server configuration.

Plug-ins can retrieve data stored in the `Plugins` XML element. To add custom data, open the Server.xml file (located in the *RootInstall*\conf folder) in a text editor and add data. Verify the XML and restart the server.

Calls to `getConfig()` can get only text node values. Calls to the method cannot get values added as attributes to elements, multiple elements, or nested elements.

For an example of a call to `IFmsServerContext->getConfig()`, see the AuthModule.cpp file.

Handling time-critical calls

Every call from the server to a File or Authorization plug-in—for example, `authorize()` in the Authorization plug-in—is time-critical. A call should be processed and returned back to the server as fast as possible, because the server remains in a pending state until the call is returned. If processing the call is time consuming and requires a wait or sleep operation, the passed arguments should be preserved and passed to a thread pool that is created by the plug-in. For an example, see the FileModule sample.

Developing an Access plug-in

Access plug-in overview

Important: *Every edition of the server supports Access plug-ins. However, the Access plug-in is not supported over RTMFP connections.*

For information about developing an Access Plug-in, see Using plug-ins to configure and optimize Flash Media Server 4.5 by Denis Bulichenko.

An Access plug-in adds another layer of security to the server; it intercepts connection requests and lets you examine the client and the server to determine whether requests should be accepted, rejected, or redirected before the requests reach the server's script layer. You can only use one Access plug-in.

You can code the plug-in to accept, reject, or redirect requests based on criteria like how many users are connected to the server and the amount of bandwidth being consumed.

You can query your organization's database of users and passwords to determine which connection requests should be allowed. Once the plug-in accepts the connection, you can update the database with a record of the user's access to the server.

You can set read and write access for files and folders on the server, set permissions to access audio and video bitmap data, and inspect client properties.

More Help topics

Adobe Media Server Plug-In API Reference

Access plug-in connection flow

Once you've installed an Access plug-in, it is initialized with a context pointer when the server starts. The context pointer and plug-in pointer provide two-way communication between the Access plug-in and the server.

When a client attempts to connect to the server, the server determines whether or not an Access plug-in exists. If the Access plug-in is available, it examines the connection request and either authorizes, rejects, or redirects the connection. If an Access plug-in is not available, connection requests proceed as usual.

Note: *There can only be one Access plug-in per server installation.*

Rewriting connections to the server

Both the Access and Authorization plug-ins let you authorize connections to the server. However, the plug-ins run at different places and are active in different stages of the connection process.

New connections arrive at the edge process (amsedge), and the data in the connection message is sent to the Access plug-in (which runs in the edge process). At this stage, the plug-in has information about the client such as IP address, the originating URL of the SWF file, the connection (or *target*) URI, user agent, and so on. Using this information, the Access plug-in can accept, reject, or redirect the connection, or rewrite the target URI. Rewriting the target URI forces the connection to a different vhost, application, or application instance. The plug-in can't rewrite a connection to a different adaptor because the socket-level connection to the adaptor has already been made.

You must use the Access plug-in to force incoming connections to a different vhost, application, or application instance; it is too late to rewrite the target URI by the time the Authorization plug-in is notified. Also, since the Access plug-in runs early in the connection process, it is the most efficient way to screen connections. If you want authorization to be as lightweight as possible, use the Access plug-in, even if you don't need to rewrite the connection URI.

The Authorization plug-in runs in the core process (fmscore) after the connection has been forwarded to the application. The Authorization plug-in can accept, reject, or redirect attempts by clients to connect to applications, but it cannot rewrite connections to different URIs.

Note: *Redirecting a connection is different than rewriting a connection URI. Redirecting a client sends a redirection message containing a new URI back to the client and terminates the current connection. The client then attempts to connect to the new URI.*

Writing the code in an Access plug-in

You can either modify the code in the sample Access plug-in file provided by Adobe, or you can write your own plug-in.

A client connection request triggers the `onAccess()` callback function in the Access plug-in. Write the code that examines connection requests, modifies client properties, and accepts, rejects, or redirects the requests in the `onAccess()` callback function.

Call `getValue()` to query client fields. Call `setValue()` to modify client fields. For a list of fields, see the `fms_access` namespace in the *Flash Media Server Plug-in API Reference*. The field `x-page-url`, the URL of the SWF file or the server in which a connection originated, was added in Adobe Media Server 3.5.

Call `getStats()` to query the following server statistics: `eTOTAL_CONNECTED`, `eBYTES_IN`, and `eBYTES_OUT`.

After you've queried the fields and written your connection logic, call `accept()`, `reject()`, or `redirect()` to allow a client to connect to an application on the server or not.

Assigning an application to a core process

You can dynamically assign an application to a core process. Use this feature to balance a load across core processes based on real-time performance counters, for example, CPU consumption and the size of the recorded media cache. You can also use this feature to provide a higher quality of service (QoS) to certain customers.

To specify which core process handles an application, pass the field `"coreIdNum"` and a core number to the `setValue()`function. The core number can be any positive integer. The server uses the following formula to determine which core process to use: core_number % number_of_ cores. For example, the following code assigns a core number of 3:

```
char* coreId = "3";
if(!setValue("coreIdNum", coreId))    {
    FILE * pFile = fopen ("error.log","a");
    fprintf(pFile, "Core id = %s", coreId);
}
```

You can call `getValue()` to find out which core process a client connection is assigned to, as in the following:

```
const char* coreId =  getValue("coreIdNum");
FILE * pFile = fopen ("output.txt","a");
fprintf(pFile, "Core id = %s", coreId);
```

Load balancing

You can code the Access plug-in to monitor performance metrics, such as CPU usage and the size of the recorded media cache, and balance loads accordingly. Distribution of applications across core processes is defined by `numprocs` and `scope` elements in Application.xml. For example, if `numprocs` is 3 and scope is set to `application`, each application gets three core processes to handle incoming connections. If a core process is overloaded, you can develop an Access plug-in that ensures new incoming connections are not sent to the overloaded core process. One strategy is to write plug-in code to do the following:

1. Track statistics for each core process. Use the Administration APIs: `getFileCacheStats()`, `getAppStats()`, `getInstanceStats()`.
2. Whenever a new client connects, the plug-in code must determine if the core process is overloaded. If it is, the plug-in code determines the core ID of a different process (one that isn't overloaded) and sends the new connection to that process.

Quality of Service (QoS)

To provide higher QoS to certain clients, you can use certain core processes exclusively for certain clients. For example, suppose you want to provide the best possible performance to certain clients that are paying a premium subscription fee to access content. You can develop an Access plug-in that distributes an application among six core processes and monitors the statistics for each process. Different core processes handle the connections to the application, depending on the client. Three core processes are reserved for the top clients; the remaining three are available to all other clients. The plug-in monitors statistics for each process. If a particular process becomes overloaded, the plug-in ensures that the overloaded process does not handle any new connections. The plug-in can route new connections to processes that can handle them.

Configuring folder permissions on the server

The `Access` section of the Application.xml configuration file (located in each virtual host directory and possibly in application directories) lets you configure folder-level permissions for the Access plug-in. If the `FolderAccess` element is set to `true`, you cannot use the `readAccess` and `writeAccess` fields in the Access plug-in to set permissions for individual files, you can only set permissions at the folder level. The default value is `false`, which lets you set permissions for individual files, as in the following:

```
<!-- Controls libconnect.dll access configurations -->
<Application>
    ...
    <Client>
        ...
        <Access>
            <FolderAccess>false</FolderAccess>
        </Access>
        ...
    ...
</Application>
```

Developing an Authorization plug-in

Authorization plug-in overview

For information about developing an Authorization Plug-in, see Using plug-ins to configure and optimize Flash Media Server 4.5 by Denis Bulichenko.

The Authorization plug-in authorizes client access to server events and fields associated with those events. Use the Authorization plug-in to do the following:

- Authorize connections to the server

  ***Note:** The Access plug-in can also authorize connections and is more lightweight. For more information, see "Access plug-in overview" on page 301.*

- Authorize playing a stream or seeking in a stream
- Authorize a dynamic stream transition
- Authorize publishing a stream or recording a stream
- Map logical URLs to physical locations

  For example, if the video player plays a stream "foo" —`ns.play("myvideos/foo")`— when the request is processed by the server, this virtual name could map to c:\apps\vidapp\myvideos\. The Authorization plug-in lets you remap this to a different physical location; for example, c:\myvideos\.

- Disconnect clients from the server
- Call a method in Server-Side ActionScript
- Access client statistics
- Identify when a new codec is discovered in a publishing stream
- Restrict the size of a recording by size or duration

Use the Authorization plug-in to deliver content to clients according to one or more of the following criteria:

- Geographic location
- Subscription level
- Stream origin
- Time and duration of a user's access to specific streams

You can also use the Authorization plug-in to monitor stream quality of service (QoS). The plug-in reports live stream QoS information to an external log file, which could then be read to another custom built tool.

More Help topics

Adobe Media Server Plug-In API Reference

Using multiple Authorization plug-ins

You can use a chain of plug-ins to sequentially perform actions on the incoming event. Allocate specific types of events to individual plug-ins: for example, auth1.dll (or auth1.so) could authorize playing a stream; auth2.dll (or auth2.so) could authorize publishing a stream, and so on.

***Note:** For cross-platform compatibility, use lowercase naming.*

The server loads the plug-ins in alphabetical order. When the server processes client requests for events, it follows alphabetical order. Each plug-in filters the incoming requests.

Writing the code in an Authorization plug-in

When the server loads an Authorization plug-in, it expects one of the following entry points:

```
FmsCreateAuthAdaptor() // Creates an Authorization plug-in.
FmsDestroyAuthAdaptor() // Destroys an Authorization plug-in.
FmsCreateAuthAdaptor2() // Creates an Authorization plug-in with versioning information.
FmsDestroyAuthAdaptor2() // Destroys an Authorization plug-in with versioning.
FmsCreateAuthAdaptor3() // Creates an Authorization plug-in with versioning information.
FmsDestroyAuthAdaptor3() // Destroys an Authorization plug-in with versioning.
```

Use the correct method for the version of the server you're using. For more information, see "About versioning" on page 296.

Implement the IFmsAuthAdaptor interface to process events as they occur. This class includes the functions `authorize()`, `notify()`, and `getEvents()`.

Once the plug-in is created, the server calls the `getEvents()` function to find out which events the plug-in wants to process. The server calls `getEvents()` once, and the events are good for the lifetime of the plug-in.

***Important:** If the `E_CODEC_CHANGE` event is not excluded, the plug-in scans all messages to detect a codec change. Subscribe to this event only as needed. This event is disabled in the Authorization plug-in sample code installed with the server.*

Some events must be authorized by the plug-in before the server executes them; other events only require the plug-in to notify the server that the plug-in received notification of the event. When authorization events occur, the server calls the `authorize()` function on the plug-in. When notification events occur, the server calls the `notify()` function on the plug-in. You write code in these functions that runs when the events occur.

The server suspends operations until the plug-in calls the `onAuthorize()` function of the IFmsAuthServerContext class. Call `onAuthorize()` in the last line of the `authorize()` function. Pass it a pointer to the event and a boolean value indicating whether the event is authorized or not. This call completes any pending operations on the server.

The server continues operating when it calls the `notify()` function of the IFmsAuthServerContext class. Call `onNotify()` in the last line of the `notify()` function. Pass it a pointer to the event. This call notifies the server that notification is complete and the event is no longer needed.

There are two ways to process events: assign an action to an event, and write code in the `authorize()` and `notify()` functions that processes the event.

Assigning actions to events

There are two actions you can assign to an event: `IFmsDisconnectAction` and `IFmsNotifyAction`. The `IFmsDisconnectAction` disconnects one or more clients when the event occurs and the `IFmsNotifyAction` calls a function on the Client object or the application object in a server-side script when the event occurs. You can add as many actions as needed. Actions are executed in the order they are assigned right before the event is disposed.

To assign an action to an event, call `addDisconnectAction()` or `addNotifyAction()` from an `IFmsAuthEvent` instance. Actions execute right before the event is disposed. If multiple actions are attached to an event, the actions execute in the order in which they were attached.

The following code adds the `IFmsNotifyAction` instance `pAction` to the `IFmsAuthEvent` instance `m_pAev`. The action calls `method()` in a server-side script and passes it the parameter `12345`.

```
FmsVariant field;
//action to notify SSAS by calling "method" with U16 variable = 12345
//and I64 variable client id
if (m_pAev->getField(IFmsAuthEvent::F_CLIENT_ID, field) == IFmsAuthEvent::S_SUCCESS)
{
    I64 clientId = field.i64;
    IFmsNotifyAction* pAction = m_pAev->addNotifyAction("Notified by adaptor");
    pAction->setClientId(field);
    const char mtd[] = "method";
    field.setString(reinterpret_cast<I8*>(const_cast<char*>(mtd)));
    pAction->setMethodName(field);
    field.setU16(12345);
///script does not work with I64 and returns invalid type
//should be preset to double or string
//field.setI64(clientId);  //wrong!!!
    field.setDouble((double)clientId);
    pAction->addParam(field);
}
```

***Note:** The previous sample code is excerpted from the AuthModule.cpp file in the rootinstall/samples/plugins/auth folder.*

You can call `addParam()` on a `IFmsNotifyAction` instance to pass parameters to a method. You must define the method in a server-side script. If you pass a value to `setClientId()`, the method is called on the server-side Client object. If you don't pass a value to `setClientId()`, the method is called on the server-side application object. For example, the following Server-Side ActionScript code defines `someMethod()` on the Client object:

```
application.onConnect = function(client){
    client.someMethod = function(msg){
        trace("inside someMethod");
    }
}
```

Mapping stream paths

***Important:** Use the characters "%3A" to escape a colon in a URL parameter. For example, if the URL is "rtmp://ip-addr/clientid/filename?foo=Smith:Joe", change it to "rtmp://ip-addr/clientid/filename?foo=Smith%3AJoe".*

Use the `authorize()` function to remap virtual stream paths to different physical locations. For example, if client 1 and client 2 both request to play the stream foo.flv, two `E_FILENAME_TRANSFORM` events are called, and you can remap the stream differently for each client. For example, client 1 could remap stream foo to c:\yourpath1\foo.flv and client 2 could remap stream foo to c:\yourpath2\foo.flv.

To map a logical stream to a different physical path, set a parameter in the Application.xml file, as follows:

```
<Application>
    ...
    <StreamManager>
        ...
        <QualifiedStreamsMapping enable="true" />
```

Every time a client plays or publishes a stream or executes a Stream call in Server-Side ActionScript, an `E_FILENAME_TRANSFORM` event occurs. The `E_FILENAME_TRANSFORM` event lets you modify the `F_STREAM_PATH` property to change the path to a stream's physical location.

The `E_FILENAME_TRANSFORM` event is a link between the Authorization plug-in and the File plug-in. The path to the physical location of a stream that you specify in this event is passed to the File plug-in for use in certain file operations.

Events occur in the following order:

1. `E_PLAY` or `E_PUBLISH` The logical stream name may be changed here.
2. `E_FILENAME_TRANSFORM` The resulting physical stream name may be changed here.
3. Playback occurs.
4. `E_STOP` or `E_UNPUBLISH`

About the E_FILENAME_TRANSFORM event

The behavior of the E_FILENAME_TRANSFORM event has improved with each release of Adobe Media Server.

Version2.5.2 the `E_FILENAME_TRANSFORM` event is called once for each stream in a application instance. Client fields are not available. For example, if client1 plays the file foo.flv from the application streamtest/inst1, the `E_FILENAME_TRANSFORM` event is called with the physical path to foo.flv (c:\defaultpath\foo.flv). Suppose you remap the physical path to c:\yourpath\yourfoo.flv. The `E_FILENAME_TRANSFORM` for the application streamtest/inst1 foo.flv is not called again and the new path, c:\yourpath\yourfoo.flv, is used for all clients.

Version 2.5.3 The `E_FILENAME_TRANSFORM` event is called for each client every time they play or publish a stream. Client fields are available. For example, suppose client1 and client2 both play the file foo.flv. Two `E_FILENAME_TRANSFORM` events are called and you can remap them for each client.

Version 3.0 The behavior of the `E_FILENAME_TRANSFORM` event was improved to handle playlists. In the case of a playlist, a client plays many clips in a row. The `E_FILENAME_TRANSFORM` event is called every time a clip is played, as in the following sequence:

```
play("clip0") / E_FILENAME_TRANSFORM / stop()
play("clip1") / E_FILENAME_TRANSFORM / stop()
```

The `E_FILENAME_TRANSFORM` event is not called when a client plays the same clip multiple times in row in a playlist, as in the following sequence:

```
play("clip0") / E_FILENAME_TRANSFORM / stop()
play("clip1") / E_FILENAME_TRANSFORM / stop()
play("clip0") / E_FILENAME_TRANSFORM / stop()
play("clip0") / stop()
play("clip1")/ E_FILENAME_TRANSFORM / stop()
```

Version 3.5 The `E_FILENAME_TRANSFORM` event is called every time a client plays a new clip in a playlist, as in the following sequence:

```
play("clip0") / E_FILENAME_TRANSFORM / stop()
play("clip1") / E_FILENAME_TRANSFORM / stop()
play("clip0") / stop()
play("clip0") / stop()
play("clip1") / stop()
```

To use the `E_FILENAME_TRANSFORM` event to remap a path when a client plays the same clip multiple times in a playlist, set `<QualifiedStreamsMapping enable="true"/>` in the Application.xml file. The following is an example of the sequence:

```
play("clip0") / E_FILENAME_TRANSFORM - Path 1/ stop()
play("clip1") / E_FILENAME_TRANSFORM - Path 1/ stop()
play("clip0") / E_FILENAME_TRANSFORM - Path 2/ stop()
play("clip0") / E_FILENAME_TRANSFORM - Path 3/ stop()
play("clip1") / E_FILENAME_TRANSFORM - Path 2/ stop()
```

When `<QualifiedStreamsMapping enable="false"/>`, the `E_FILENAME_TRANSFORM` event is called only the first time a clip plays.

Differences between edge and origin deployments

Authorization plug-ins can be deployed on origin servers and edge servers; the functionality is different in each case.

Origin When an Authorization plug-in is installed on an origin server, the E_PLAY (for a live stream) or E_LOADSEGMENT (for recorded stream) event is called to play the clip. To block play for a recorded stream, process the E_LOADSEGMENT event rather than E_PLAY.

Edge (Core) Server-Side ActionScript is not executed on edge servers, so the NOTIFY action cannot be used and should always return a failure status.

Handling errors

Authorization plug-ins are loaded by the FMSCore process. Exceptions inside a plug-in crash FMSCore.

Important: *Authorization plug-ins must handle exceptions to prevent FMSCore from crashing.*

Specify a reason string for an authorization failure

Adobe Media Server 4.0

In the Authorization plug-in, specify a reason string for the authorization failure. In the Authorization plug-in, the Interface IFmsAuthServerContext has been extended to IFmsAuthServerContext2. This interface overrides the IFmsAuthServerContext::OnAuthorize method to include the AuthFailureDesc structure. Use the AuthFailureDesc structure to specify a reason string for the authorization failure, error code, and status code. To use this feature use FmsCreateAuthAdaptor3() and pass it IFmsAuthServerContext2 as the parameter.

The server uses the IFmsAuthServerContext2::StatusCode enum to send the client a NetStatus message about the failure.

For more information, see the IFmsAuthServerContext2 class in the Flash Media Server Plug-in API Reference.

Accessing client statistics

Accessing client statistics through the Authorization plug-in in the C++ layer can provide better server performance than accessing client statistics in a server-side script. The Authorization plug-in aggregates requests before sending them to the server.

The following client statistics are defined in the FmsClientStats struct in the FmsAuthAdaptor.h file:

Statistic	Description
bytes_in	Total number of bytes received.
bytes_out	Total number of bytes sent.
msg_in	Total number of messages received.
msg_out	Total number of messages sent.
msg_dropped	Total number of dropped messages.

To access client statistics, implement the FmsCreateAuthAdaptor2() function. For an example, see the AuthModule.cpp file installed to *RootInstall*\samples\plugins\auth.

When a client connects to the server, the server sends an E_CONNECT event to the Authorization plug-in. After the plug-in receives the E_CONNECT event, you can call IFmsAuthServerContext::getClientStats() to get statistics. Every event, such as E_PLAY and E_STOP, contains a field called F_CLIENT_STATS_HANDLE. Pass this client stats handle back to the server in the getClientStats() call. If you use F_CLIENT_STATS_HANDLE during any notification event (except E_CONNECT and E_DISCONNECT), you do not need to save the handle.

To verify that a client is connected, check the return status of the call to getClientStats(). This check is especially necessary if you are using multiple Authorization plug-ins. For example, if plug-in A subscribes to the E_CONNECT event and plug-in B subscribes to the E_DISCONNECT event. When the client disconnects from the server, the adaptor A does not receive the E_DISCONNECT message.

When the server receives a disconnect message from the client, it notifies the Authorization plug-in with an E_DISCONNECT event. The F_CLIENT_STATS_HANDLE field is invalid after the E_DISCONNECT event.

Smart seeking

Flash Media Server 3.5.3

When the client NetStream.inBufferSeek property is true, "smart seek" is enabled. Smart seek lets Flash Player seek within a back buffer and a foward buffer. The player sends commands to Adobe Media Server to manage the buffer. The commands are "seekRaw", "startTransmit", and "stopTransmit".

When smart seek is enabled and a client calls NetStream.seek(), the Authorization plug-in receives an E_CLIENT_SEEK event. Use this event to write seek events to a log. The F_STREAM_SEEK_POSITION property is available for this event. This read-only property is the position to which the client wants to seek.

***Note:** The E_CLIENT_SEEK event is a notification event. You cannot use it to prevent the client from seeking within the client-side buffer.*

The E_START_TRANSMIT event is a notification and authorization event. The E_STOP_TRANSMIT event is a notification event. Use the E_START_TRANSMIT event to block the client from asking the server to start sending data.

The Authorization plug-in receives an E_START_TRANSMIT event when the Flash Player buffer falls below a threshold. This command asks the server to transmit more data because the buffer is running low. The F_STREAM_TRANSMIT_POSITION property is available for this event. This read-only property is the position (in milliseconds) from which the client wants the server to start transmission.The Authorization plug-in receives an E_STOP_TRANSMIT event when the Flash Player buffer is above a threshold. This command asks the server to suspend transmission because there is enough data in the buffer. The F_STREAM_TRANSMIT_POSITION property is available during this event. This read-only property is the position (in milliseconds) of the data at the end of the client buffer when the client sends a stopTransmit command.

Stream reconnecting

Flash Media Server 3.5.3

Flash Media Server 3.5.3 and Flash Player 10.1 allow you to build applications that reconnect streams seamlessly when a connection is dropped or when a client switches from a wired to a wireless network connection.

Use the E_PLAY event to capture information from the F_STREAM_OFFSET property and the F_STREAM_TRANSITION property. The F_STREAM_OFFSET property (read-only) indicates where to resume streaming after a reconnection, in seconds.The F_STREAM_TRANSITION property (read/write) indicates the transition mode sent by the client in the NetStream.play2() call. The values for Stream Reconnect are "resume" and "appendAndWait".

More Help topics

"Reconnecting streams when a connection drops" on page 219

Accessing events and fields

The server provides an interface to an event object, `IFmsAuthEvent`, that gives the plug-in access to events. The plug-in has access to certain fields during each event. Not all fields are accessible during all events and each field is either read-only or read/write.

Fields can pertain to an application, a client, or a stream. For example, the value of fields with a name in the form `F_CLIENT_*`, such as `F_CLIENT_URI` and `F_CLIENT_REDIRECT_URI`, come from a client. An event object associated with a client event contains values for these fields.

Events associated with server-side scripts aren't necessarily associated with a client, and therefore do not always contain values for `F_CLIENT_*` fields. If it is necessary to have values for the `F_CLIENT_*` fields from server-side script calls, the server-side script call must be invoked by a method attached to a Client object. For example, the `play()` method invokes the `E_FILENAME_TRANSFORM` event.

Authorization plug-in events

The following table lists the Authorization plug-in events and whether the event is available in the `notify()` function, the `authorize()` function, or both:

Event name	Server availability	Notify or Authorize	Description
`E_APPSTART`	2.5	notify	Start an application.
`E_APPSTOP`	2.5	notify	Stop an application.
`E_CONNECT`	2.5	notify, authorize	A client established a TCP control connection to the server.
`E_DISCONNECT`	2.5	notify	The TCP connection between the client and the server is broken.
`E_FILENAME_TRANSFORM`	2.5	notify, authorize	The server has requested permission to map the logical filename requested by a client.
`E_PLAY`	2.5	notify, authorize	Play a stream.
`E_STOP`	2.5	notify	Stop a stream.
`E_SEEK`	2.5	notify, authorize	Seek a stream.
`E_PAUSE`	2.5	notify, authorize	Pause delivery of a stream.
`E_PUBLISH`	2.5	notify, authorize	Publish a stream.
`E_UNPUBLISH`	2.5	notify	Unpublish a stream.
`E_LOADSEGMENT`	2.5	notify, authorize	Load a segment.
`E_ACTION`	2.5	N/A	An action attached to an event was executed.
`E_CODEC_CHANGE`	3.5	notify	A new codec is discovered in the stream.
`E_RECORD`	3.5	notify, authorize	Record a stream.
`E_RECORD_STOP`	3.5	notify	Stop recording a stream.
`E_CLIENT_PAUSE`	3.5	notify	Client smart-pauses a stream.

Event name	Server availability	Notify or Authorize	Description
`E_SWF_VERIFY`	3.5.3	notify, authorize	Called before the SWF Verification process begins. Use this event to disallow the verification which terminates the connection. You can also use this event to provide a digest for the calling SWF. Use this mechanism to manage an inventory of SWFs that sit outside the server and aren't cached and managed by the server. This event is called multiple times if the SWF is large enough to require partial matching.
`E_SWF_VERIFY_COMPLETE`	3.5.3	notify	SWF Verification has completed successfully for this client.
`E_CLIENT_SEEK`	3.5.3	notify	Client smart-seeks a stream.
`E_START_TRANSMIT`	3.5.3	notify, authorize	Start stream transmission.
`E_STOP_TRANSMIT`	3.5.3	notify	Stop stream transmission.
`E_MAXEVENT`	2.5	N/A	The total number of events in the enum field.

Authorization plug-in fields

The following table lists the Authorization plug-in fields:

Field	Server version	Data type	Description
`F_APP_INST`	2.5	String	The application instance name.
`F_APP_NAME`	2.5	String	The application name.
`F_APP_URI`	2.5	String	The URI of the application to which the client connected. The value does not include the server name or port information.
`F_CLIENT_AMF_ENCODING`	2.5	I8	The AMF (Action Message Format) encoding of the client.
`F_CLIENT_AUDIO_CODECS`	2.5	I32	A list of audio codecs supported on the client.
`F_CLIENT_AUDIO_SAMPLE_ACCESS`	2.5	String	Gives the client access to raw, uncompressed audio data from streams in the specified folders.
`F_CLIENT_AUDIO_SAMPLE_ACCESS_LOCK`	2.5	I8	A Boolean value preventing a server-side script from setting the `Client.audioSampleAccess` property (`true`), or not (`false`).
`F_CLIENT_CONNECT_TIME`	3.5.3	I64	The time the client connected to the server, in seconds since Jan 1, 1970. You can retrieve this value from any event that has an associated client. The `getField()` function returns `S_SUCCESS` if this field is successfully retrieved.
`F_CLIENT_DIFFSERV_BITS`	3.5.2	U8	DSCP bits to be used in `setsockopt` for a particular NetConnection.
`F_CLIENT_DIFFSERV_MASK`	3.5.2	U8	DSCP mask to be used in `setsockopt`.
`F_CLIENT_FAR_ID`	3.5	String	The far ID of the client

Field	Server version	Data type	Description
`F_CLIENT_FAR_NONCE`	3.5	String	The far nonce of the client.
`F_CLIENT_FORWARDED_FOR`	4.0	String	Any x-forwarded-for header included by an HTTP Proxy on an RTMPT session.
`F_CLIENT_ID`	2.5	I64	A 64-bit integer value that uniquely identifies the client (this value is not unique across processes).
`F_CLIENT_IP`	2.5	String	The client IP address.
`F_CLIENT_NEAR_ID`	3.5	String	The near ID of the client
`F_CLIENT_NEAR_NONCE`	3.5	String	The near nonce of the client.
`F_CLIENT_PAGE_URL`	2.5	String	The URL of the client SWF file. (This field is set only when clients use Flash Player 8 or above).
`F_CLIENT_PROTO`	2.5	String	The protocol the client used to connect to the server.
`F_CLIENT_PROTO_VER`	3.5	String	The version of the protocol that the client used to connect to the server.
`F_CLIENT_READ_ACCESS`	2.5	String	A string of directories containing application resources (shared objects and streams) to which the client has read access.
`F_CLIENT_READ_ACCESS_LOCK`	2.5	I8	A Boolean value preventing a server-side script from setting the `Client.readAccess` property (`true`), or not (`false`).
`F_CLIENT_REDIRECT_URI`	3.0	String	A URI to redirect the connection to.
`F_CLIENT_REFERRER`	2.5	String	The URL of the SWF file or server where this connection originated.
`F_CLIENT_SECURE`	2.5	I8	A Boolean value indicating whether a connection is secure (`true`) or not (`false`).
`F_CLIENT_STATS_HANDLE`	3.5	I64	A string that uniquely identifies the client stats handle.
`F_CLIENT_SWFV_DEPTH`	3.5.3	I64	During SWF Verification, the 64-bit size of the original SWF that corresponds with the digest.
`F_CLIENT_SWFV_DIGEST`	3.5.3	U8*	During SWF Verification, the 32-byte hash digest, as a byte buffer.
`F_CLIENT_SWFV_EXCEPTION`	3.5.3	I8	A Boolean value indicating whether this client is an exception to SWF Verification (`true`) or not (`false`). Set this value to `true` if you want a client who would normally be subject to SWF Verification to be excepted from that requirement. This field is similar to the UserAgentExceptions function in the Application.xml configuration file but in this case you except a single client.
`F_CLIENT_SWFV_RESULT`	4.0	None	The completed result of a SWF Verification attempt. Possible values are in the eSWFMatch enum.
`F_CLIENT_SWFV_TTL`	3.5.3	I32	During SWF Verification, the time to live (in seconds) of each SWF hash in the cache.
`F_CLIENT_SWFV_VERSION`	3.5.3	I8	During SWF Verification, the version of the protocol.

Field	Server version	Data type	Description
F_CLIENT_TYPE	2.5	I32	The client type. Possible values are Normal, Group, GroupElement (a client connected through an edge server), Service, All.
F_CLIENT_URI	2.5	String	The URI the client specified to connect to the server. The value does not include the application name or instance name.
F_CLIENT_URI_STEM	2.5	String	The stem of the URI the client specified to connect to the server, without the query string.
F_CLIENT_USER_AGENT	2.5	String	The client user-agent.
F_CLIENT_USERDATA	3.5.3	U8*	Use this field to associate some user data with a client. Set this field from any authoization event that has an associated client. Retrieve this field in any subsequent event associated with the client. The getField() and setField() functions return S_SUCCESS if user data is successfully get or set. You can set anything that can be stored in an FmsVariant. When setting data, the server makes a copy before saving. When getting data, the plug-in should make a copy if the data needs to remain valid past the lifetime of the event.
F_CLIENT_VHOST	2.5	String	The virtual host of the application the client is connected to.
F_CLIENT_VIDEO_CODECS	2.5	I32	A list of video codecs supported on the client.
F_CLIENT_VIDEO_SAMPLE_ACCESS	2.5	String	Gives the client access to raw, uncompressed video data from streams in the specified folders.
F_CLIENT_VIDEO_SAMPLE_ACCESS_LOCK	2.5	I8	A Boolean value preventing a server-side script from setting the Client.videoSampleAccess property (true), or not (false).
F_CLIENT_WRITE_ACCESS	2.5	String	A string of directories containing application resources (shared objects and streams) to which the client has write access.
F_CLIENT_WRITE_ACCESS_LOCK	2.5	I8	A Boolean value preventing a server-side script from setting the Client.writeAccess property (true), or not (false)."
F_MAXFIELD	3.5	None	The total number of fields in the enum field.
F_OLD_STREAM_NAME	3.5	String	The old stream for a switch or a swap.
F_OLD_STREAM_QUERY	3.5	String	The old stream query string for a switch or a swap.
F_OLD_STREAM_TYPE	3.5	String	The old stream type for a switch or a swap.
F_SEGMENT_END	3.0	I64	The segment end boundary in bytes; available in E_LOADSEGMENT only.
F_SEGMENT_START	3.0	I64	The segment start boundary in bytes; available in E_LOADSEGMENT only.
F_STREAM_CODEC	3.5	U16	The codec value discovered in the stream; available in E_CODEC only.

Field	Server version	Data type	Description
F_STREAM_CODEC_TYPE	3.5	U16	The codec type discovered in the stream; available in E_CODEC only.
F_STREAM_ID	3.5	I32	The stream ID.
F_STREAM_IGNORE	2.5	I8	A Boolean value indicating whether to ignore timestamps (true), or not (false).
F_STREAM_LENGTH	2.5	Float	The length of the stream. For stream events other than E_PLAY, use the F_STREAM_SEEK_POSITION, F_SEGMENT_START and F_SEGMENT_END fields.
F_STREAM_LIVE_EVENT	3.0	None	Use this field to assign a stream to a live event. The server uses the live event to place the stream into the HTTP Dynamic Streaming manifest file. This field is the same as the Server-Side ActionScript Stream.liveEvent property.
F_STREAM_LIVE_PUBLISH_PENDING	3.5.2	I32	A Boolean value. If false, when a client plays a stream and issues a SWITCH transition and the stream is not yet published the client receives a NetStream.Play.StreamNotFound message. If true, the client does not receive the message because the server waits for the stream to publish.
F_STREAM_NAME	2.5	String	The stream name.
F_STREAM_OFFSET	3.8	Float	The offset value when F_STREAM_TRANSITION is offset.
F_STREAM_PATH	2.5	String	The physical path of the stream on the server. When the plug-in is deployed on an edge server, the physical path of the cache.
F_STREAM_PAUSE	3.0	I8	A Boolean value indicating whether to pause or unpause; available in E_PAUSE only.
F_STREAM_PAUSE_TIME	3.0	Float	The time at which a stream is paused; available in E_PAUSE only.
F_STREAM_PAUSE_TOGGLE	3.0	I8	The stream pause toggle; available in E_PAUSE only.
F_STREAM_POSITION	2.5	Float	The position of the stream; for stream events other than E_PLAY, use the F_STREAM_SEEK_POSITION, F_STREAM_PAUSE_TIME, F_SEGMENT_START, and F_SEGMENT_END fields.
F_STREAM_PUBLISH_BROADCAST	3.0	I32	Broadcast or multicast; currently not supported.
F_STREAM_PUBLISH_TYPE	3.0	I32	The stream publishing type (available for E_PUBLISH only): 0 for record, 1 for record and append to existing stream, 2 for record and append to existing stream while preserving gaps in the recording, -1 for live.
F_STREAM_QUERY	2.5	String	A query appended to a stream, for example, streamName?streamQuery.
F_STREAM_RECORD_MAXDURATION	3.5	Float	The maximum duration (in seconds) of a stream recording; available in E_RECORD only.
F_STREAM_RECORD_MAXSIZE	3.5	Float	The maximum size (in kb) of a stream recording; available in E_RECORD only.

Field	Server version	Data type	Description
F_STREAM_RESET	3.0	I8	A Boolean value indicating whether to allow adding a stream to a playlist (true), or not (false); for stream events other than E_PLAY, use F_STREAM_PUBLISH_TYPE.
F_STREAM_SEEK_POSITION	3.0	I8	The position to which the stream should seek; available in E_SEEK only.
F_STREAM_TRANSITION	3.5	String	The transition mode string for a switch or a swap.
F_STREAM_TRANSMIT_POSITION	3.0	Float	The position to which the stream should start/stop; available in E_START_TRANSMIT and E_STOP_TRANSMIT only.
F_STREAM_TYPE	2.5	String	The file type of the stream.

Authorization plug-in events and fields support matrix

The following tables shows the fields you can access from each event. A single table is too wide to display because there are too many events. Therefore, the matrix is divided into mulitple tables. Each table lists all the fields and some events.

	E_APPSTART	E_APPSTOP	E_CONNECT	E_DISCONNECT
F_APP_INST	Read-only	Read-only	Read-only	Read-only
F_APP_NAME	Read-only	Read-only	Read-only	Read-only
F_APP_URI	Read-only	Read-only	Read-only	Read-only
F_CLIENT_AMF_ENCODING	N/A	N/A	Read-only	Read-only
F_CLIENT_AUDIO_CODECS	N/A	N/A	Read-only	Read-only
F_CLIENT_AUDIO_SAMPLE_ACCESS	N/A	N/A	Read/Write	Read-only
F_CLIENT_AUDIO_SAMPLE_ACCESS_LOCK	N/A	N/A	Read/Write	Read-only
F_CLIENT_CONNECT_TIME	N/A	N/A	Read-only	Read-only
F_CLIENT_DIFFSERV_BITS	N/A	N/A	Read/Write	N/A
F_CLIENT_DIFFSERV_MASK	N/A	N/A	Read/Write	N/A
F_CLIENT_FAR_ID	N/A	N/A	Read-only	Read-only
F_CLIENT_FAR_NONCE	N/A	N/A	Read-only	Read-only
F_CLIENT_FORWARDED_FOR	N/A	N/A	Read-only	Read-only
F_CLIENT_ID	N/A	N/A	Read-only	Read-only
F_CLIENT_IP	N/A	N/A	Read-only	Read-only
F_CLIENT_NEAR_ID	N/A	N/A	Read-only	Read-only

	E_APPSTART	**E_APPSTOP**	**E_CONNECT**	**E_DISCONNECT**
`F_CLIENT_NEAR_NONCE`	N/A	N/A	Read-only	Read-only
`F_CLIENT_PAGE_URL`	N/A	N/A	Read-only	Read-only
`F_CLIENT_PROTO`	N/A	N/A	Read-only	Read-only
`F_CLIENT_PROTO_VER`	N/A	N/A	Read-only	Read-only
`F_CLIENT_READ_ACCESS`	N/A	N/A	Read/Write	Read-only
`F_CLIENT_READ_ACCESS_LOCK`	N/A	N/A	Read/Write	Read-only
`F_CLIENT_REDIRECT_URI`	N/A	N/A	Read/Write	Read-only
`F_CLIENT_REFERRER`	N/A	N/A	Read-only	Read-only
`F_CLIENT_SECURE`	N/A	N/A	Read-only	Read-only
`F_CLIENT_STATS_HANDLE`	N/A	N/A	Read-only	Read-only
`F_CLIENT_SWFV_DEPTH`	N/A	N/A	N/A	N/A
`F_CLIENT_SWFV_DIGEST`	N/A	N/A	N/A	N/A
`F_CLIENT_SWFV_EXCEPTION`	Read-only	Read-only	Read/Write	Read-only
`F_CLIENT_SWFV_RESULT`	N/A	N/A	N/A	N/A
`F_CLIENT_SWFV_TTL`	N/A	N/A	N/A	N/A
`F_CLIENT_SWFV_VERSION`	N/A	N/A	N/A	N/A
`F_CLIENT_TYPE`	N/A	N/A	Read-only	Read-only
`F_CLIENT_URI`	N/A	N/A	Read-only	Read-only
`F_CLIENT_URI_STEM`	N/A	N/A	Read-only	Read-only
`F_CLIENT_USER_AGENT`	N/A	N/A	Read-write	Read-only
`F_CLIENT_USERDATA`	N/A	N/A	Read-write	Read-only
`F_CLIENT_VHOST`	N/A	N/A	Read-only	Read-only
`F_CLIENT_VIDEO_CODECS`	N/A	N/A	Read-only	Read-only
`F_CLIENT_VIDEO_SAMPLE_ACCESS`	N/A	N/A	Read/Write	Read-only
`F_CLIENT_VIDEO_SAMPLE_ACCESS_LOCK`	N/A	N/A	Read/Write	Read-only

	E_APPSTART	E_APPSTOP	E_CONNECT	E_DISCONNECT
F_CLIENT_WRITE_ACCESS	N/A	N/A	Read/Write	Read-only
F_CLIENT_WRITE_ACCESS_LOCK	N/A	N/A	Read/Write	Read-only
F_MAXFIELD	N/A	N/A	N/A	N/A
F_OLD_STREAM_NAME	N/A	N/A	N/A	N/A
F_OLD_STREAM_QUERY	N/A	N/A	N/A	N/A
F_OLD_STREAM_TYPE	N/A	N/A	N/A	N/A
F_SEGMENT_END	N/A	N/A	N/A	N/A
F_SEGMENT_START	N/A	N/A	N/A	N/A
F_STREAM_CODEC	N/A	N/A	N/A	N/A
F_STREAM_CODEC_TYPE	N/A	N/A	N/A	N/A
F_STREAM_ID	N/A	N/A	N/A	N/A
F_STREAM_IGNORE	N/A	N/A	N/A	N/A
F_STREAM_LENGTH	N/A	N/A	N/A	N/A
F_STREAM_LIVE_EVENT	N/A	N/A	N/A	N/A
F_STREAM_LIVE_PUBLISH_PENDING	N/A	N/A	N/A	N/A
F_STREAM_NAME	N/A	N/A	N/A	N/A
F_STREAM_OFFSET	N/A	N/A	N/A	N/A
F_STREAM_PATH	N/A	N/A	N/A	N/A
F_STREAM_PAUSE	N/A	N/A	N/A	N/A
F_STREAM_PAUSE_TIME	N/A	N/A	N/A	N/A
F_STREAM_PAUSE_TOGGLE	N/A	N/A	N/A	N/A
F_STREAM_POSITION	N/A	N/A	N/A	N/A
F_STREAM_PUBLISH_BROADCAST	N/A	N/A	N/A	N/A
F_STREAM_PUBLISH_TYPE	N/A	N/A	N/A	N/A
F_STREAM_QUERY	N/A	N/A	N/A	N/A
F_STREAM_RECORD_MAXDURATION	N/A	N/A	N/A	N/A
F_STREAM_RECORD_MAXSIZE	N/A	N/A	N/A	N/A
F_STREAM_RESET	N/A	N/A	N/A	N/A

	E_APPSTART	E_APPSTOP	E_CONNECT	E_DISCONNECT
`F_STREAM_SEEK_POSITION`	N/A	N/A	N/A	N/A
`F_STREAM_TRANSITION`	N/A	N/A	N/A	N/A
`F_STREAM_TRANSMIT_POSITION`	N/A	N/A	N/A	N/A
`F_STREAM_TYPE`	N/A	N/A	N/A	N/A

The following table displays all the fields and more events:

	E_FILENAME_TRANSFORM	E_PLAY	E_STOP
`F_APP_INST`	Read-only	Read-only	Read-only
`F_APP_NAME`	Read-only	Read-only	Read-only
`F_APP_URI`	Read-only	Read-only	Read-only
`F_CLIENT_AMF_ENCODING`	Read-only	Read-only	Read-only
`F_CLIENT_AUDIO_CODECS`	Read-only	Read-only	Read-only
`F_CLIENT_AUDIO_SAMPLE_ACCESS`	Read-only	Read-only	Read-only
`F_CLIENT_AUDIO_SAMPLE_ACCESS_LOCK`	Read-only	Read-only	Read-only
`F_CLIENT_CONNECT_TIME`	Read-only	Read-only	Read-only
`F_CLIENT_DIFFSERV_BITS`	N/A	Read/Write	Read-only
`F_CLIENT_DIFFSERV_MASK`	N/A	Read/Write	Read-only
`F_CLIENT_FAR_ID`	Read-only	Read-only	Read-only
`F_CLIENT_FAR_NONCE`	Read-only	Read-only	Read-only
`F_CLIENT_FORWARDED_FOR`	Read-only	Read-only	Read-only
`F_CLIENT_ID`	Read-only	Read-only	Read-only
`F_CLIENT_IP`	Read-only	Read-only	Read-only
`F_CLIENT_NEAR_ID`	Read-only	Read-only	Read-only
`F_CLIENT_NEAR_NONCE`	Read-only	Read-only	Read-only
`F_CLIENT_PAGE_URL`	Read-only	Read-only	Read-only
`F_CLIENT_PROTO`	Read-only	Read-only	Read-only
`F_CLIENT_PROTO_VER`	Read-only	Read-only	Read-only

	E_FILENAME_TRANSFORM	E_PLAY	E_STOP
F_CLIENT_READ_ACCESS	Read-only	Read-only	Read-only
F_CLIENT_READ_ACCESS_LOCK	Read-only	Read-only	Read-only
F_CLIENT_REDIRECT_URI	Read-only	Read-only	Read-only
F_CLIENT_REFERRER	Read-only	Read-only	Read-only
F_CLIENT_SECURE	Read-only	Read-only	Read-only
F_CLIENT_STATS_HANDLE	Read-only	Read-only	Read-only
F_CLIENT_SWFV_DEPTH	N/A	N/A	N/A
F_CLIENT_SWFV_DIGEST	N/A	N/A	N/A
F_CLIENT_SWFV_EXCEPTION	Read-only	Read-only	Read-only
F_CLIENT_SWFV_RESULT	N/A	N/A	N/A
F_CLIENT_SWFV_TTL	N/A	N/A	N/A
F_CLIENT_SWFV_VERSION	N/A	N/A	N/A
F_CLIENT_TYPE	Read-only	Read-only	Read-only
F_CLIENT_URI	Read-only	Read-only	Read-only
F_CLIENT_URI_STEM	Read-only	Read-only	Read-only
F_CLIENT_USER_AGENT	Read-write	Read-write	Read-only
F_CLIENT_USERDATA	Read-write	Read-write	Read-only
F_CLIENT_VHOST	Read-only	Read-only	Read-only
F_CLIENT_VIDEO_CODECS	Read-only	Read-only	Read-only
F_CLIENT_VIDEO_SAMPLE_ACCESS	Read-only	Read-only	Read-only
F_CLIENT_VIDEO_SAMPLE_ACCESS_LOCK	Read-only	Read-only	Read-only
F_CLIENT_WRITE_ACCESS	Read-only	Read-only	Read-only
F_CLIENT_WRITE_ACCESS_LOCK	Read-only	Read-only	Read-only
F_MAXFIELD	N/A	N/A	N/A
F_OLD_STREAM_NAME	N/A	Read-write	N/A
F_OLD_STREAM_QUERY	N/A	Read-write	N/A
F_OLD_STREAM_TYPE	N/A	Read-write	N/A

	E_FILENAME_TRANSFORM	**E_PLAY**	**E_STOP**
`F_SEGMENT_END`	N/A	N/A	N/A
`F_SEGMENT_START`	N/A	N/A	N/A
`F_STREAM_CODEC`	N/A	N/A	N/A
`F_STREAM_CODEC_TYPE`	N/A	N/A	N/A
`F_STREAM_ID`	Read-only	Read-only	Read-only
`F_STREAM_IGNORE`	Read-only	Read-write	Read-only
`F_STREAM_LENGTH`	Read-only	Read-write	Read-only
`F_STREAM_LIVE_EVENT`	N/A	N/A	N/A
`F_STREAM_LIVE_PUBLISH_PENDING`	N/A	Read-write	N/A
`F_STREAM_NAME`	N/A	Read-write	N/A
`F_STREAM_OFFSET`	N/A	Read-only	N/A
`F_STREAM_PATH`	Read-write	Read-only	Read-only
`F_STREAM_PAUSE`	N/A	N/A	N/A
`F_STREAM_PAUSE_TIME`	N/A	N/A	N/A
`F_STREAM_PAUSE_TOGGLE`	N/A	N/A	N/A
`F_STREAM_POSITION`	Read-only	Read-write	Read-only
`F_STREAM_PUBLISH_BROADCAST`	N/A	N/A	N/A
`F_STREAM_PUBLISH_TYPE`	N/A	N/A	N/A
`F_STREAM_QUERY`	Read-only	Read-write	Read-only
`F_STREAM_RECORD_MAXDURATION`	N/A	N/A	N/A
`F_STREAM_RECORD_MAXSIZE`	N/A	N/A	N/A
`F_STREAM_RESET`	Read-only	Read-write	Read-only
`F_STREAM_SEEK_POSITION`	N/A	N/A	N/A
`F_STREAM_TRANSITION`	N/A	Read-write	N/A
`F_STREAM_TRANSMIT_POSITION`	N/A	N/A	N/A
`F_STREAM_TYPE`	Read-write	Read-write	Read-only

The following table displays all the fields and more events:

	E_CLIENT_SEEK	E_SEEK	E_CLIENT_PAUSE	E_PAUSE
F_APP_INST	Read-only	Read-only	N/A	Read-only
F_APP_NAME	Read-only	Read-only	N/A	Read-only
F_APP_URI	Read-only	Read-only	N/A	Read-only
F_CLIENT_AMF_ENCODING	Read-only	Read-only	N/A	Read-only
F_CLIENT_AUDIO_CODECS	Read-only	Read-only	N/A	Read-only
F_CLIENT_AUDIO_SAMPLE_ACCESS	Read-only	Read-only	N/A	Read-only
F_CLIENT_AUDIO_SAMPLE_ACCESS_LOCK	Read-only	Read-only	N/A	Read-only
F_CLIENT_CONNECT_TIME	Read-only	Read-only	N/A	Read-only
F_CLIENT_DIFFSERV_BITS	N/A	N/A	N/A	N/A
F_CLIENT_DIFFSERV_MASK	N/A	N/A	N/A	N/A
F_CLIENT_FAR_ID	Read-only	Read-only	N/A	Read-only
F_CLIENT_FAR_NONCE	Read-only	Read-only	N/A	Read-only
F_CLIENT_FORWARDED_FOR	Read-only	Read-only	N/A	Read-only
F_CLIENT_ID	Read-only	Read-only	N/A	Read-only
F_CLIENT_IP	Read-only	Read-only	N/A	Read-only
F_CLIENT_NEAR_ID	Read-only	Read-only	N/A	Read-only
F_CLIENT_NEAR_NONCE	Read-only	Read-only	N/A	Read-only
F_CLIENT_PAGE_URL	Read-only	Read-only	N/A	Read-only
F_CLIENT_PROTO	Read-only	Read-only	N/A	Read-only
F_CLIENT_PROTO_VER	Read-only	Read-only	N/A	Read-only
F_CLIENT_READ_ACCESS	Read-only	Read-only	N/A	Read-only
F_CLIENT_READ_ACCESS_LOCK	Read-only	Read-only	N/A	Read-only
F_CLIENT_REDIRECT_URI	Read-only	Read-only	N/A	Read-only
F_CLIENT_REFERRER	Read-only	Read-only	N/A	Read-only
F_CLIENT_SECURE	Read-only	Read-only	N/A	Read-only
F_CLIENT_STATS_HANDLE	Read-only	Read-only	N/A	Read-only
F_CLIENT_SWFV_DEPTH	N/A	N/A	N/A	N/A

	E_CLIENT_SEEK	E_SEEK	E_CLIENT_PAUSE	E_PAUSE
F_CLIENT_SWFV_DIGEST	N/A	N/A	N/A	N/A
F_CLIENT_SWFV_EXCEPTION	Read-only	Read-only	Read-only	Read-only
F_CLIENT_SWFV_RESULT	N/A	N/A	N/A	N/A
F_CLIENT_SWFV_TTL	N/A	N/A	N/A	N/A
F_CLIENT_SWFV_VERSION	N/A	N/A	N/A	N/A
F_CLIENT_TYPE	N/A	Read-only	N/A	Read-only
F_CLIENT_URI	N/A	Read-only	N/A	Read-only
F_CLIENT_URI_STEM	N/A	Read-only	N/A	Read-only
F_CLIENT_USER_AGENT	N/A	Read-only	N/A	Read-only
F_CLIENT_USERDATA	N/A	Read-only	N/A	Read-write
F_CLIENT_VHOST	N/A	Read-only	N/A	Read-only
F_CLIENT_VIDEO_CODECS	N/A	Read-only	N/A	Read-only
F_CLIENT_VIDEO_SAMPLE_ACCESS	N/A	Read-only	N/A	Read-only
F_CLIENT_VIDEO_SAMPLE_ACCESS_LOCK	N/A	Read-only	N/A	Read-only
F_CLIENT_WRITE_ACCESS	N/A	Read-only	N/A	Read-only
F_CLIENT_WRITE_ACCESS_LOCK	N/A	Read-only	N/A	Read-only
F_MAXFIELD	N/A	N/A	N/A	N/A
F_OLD_STREAM_NAME	N/A	N/A	N/A	N/A
F_OLD_STREAM_QUERY	N/A	N/A	N/A	N/A
F_OLD_STREAM_TYPE	N/A	N/A	N/A	N/A
F_SEGMENT_END	N/A	N/A	N/A	N/A
F_SEGMENT_START	N/A	N/A	N/A	N/A
F_STREAM_CODEC	N/A	N/A	N/A	N/A
F_STREAM_CODEC_TYPE	N/A	N/A	N/A	N/A
F_STREAM_ID	N/A	Read-only	N/A	Read-only
F_STREAM_IGNORE	N/A	Read-only	Read-only	Read-only
F_STREAM_LENGTH	N/A	Read-write	Read-only	Read-only
F_STREAM_LIVE_EVENT	N/A	N/A	N/A	N/A
F_STREAM_LIVE_PUBLISH_PENDING	N/A	N/A	N/A	N/A

	E_CLIENT_SEEK	E_SEEK	E_CLIENT_PAUSE	E_PAUSE
F_STREAM_NAME	N/A	N/A	Read-only	N/A
F_STREAM_OFFSET	N/A	N/A	N/A	N/A
F_STREAM_PATH	N/A	Read-only	N/A	Read-only
F_STREAM_PAUSE	N/A	N/A	Read-only	Read-only
F_STREAM_PAUSE_TIME	N/A	N/A	Read-only	Read-only
F_STREAM_PAUSE_TOGGLE	N/A	N/A	Read-only	Read-only
F_STREAM_POSITION	N/A	Read-only	Read-only	Read-only
F_STREAM_PUBLISH_BROADCAST	N/A	N/A	N/A	N/A
F_STREAM_PUBLISH_TYPE	N/A	N/A	N/A	N/A
F_STREAM_QUERY	N/A	Read-only	Read-only	Read-only
F_STREAM_RECORD_MAXDURATION	N/A	N/A	N/A	N/A
F_STREAM_RECORD_MAXSIZE	N/A	N/A	N/A	N/A
F_STREAM_RESET	N/A	Read-only	Read-only	Read-only
F_STREAM_SEEK_POSITION	Read-only	N/A	N/A	N/A
F_STREAM_TRANSITION	N/A	N/A	N/A	N/A
F_STREAM_TRANSMIT_POSITION	N/A	N/A	N/A	N/A
F_STREAM_TYPE	N/A	Read-only	Read-only	Read-only

The following table displays all the fields and more events:

	E_PUBLISH	E_UNPUBLISH	E_LOADSEGMENT	E_CODEC_CHANGE
F_APP_INST	Read-only	Read-only	Read-only	Read-only
F_APP_NAME	Read-only	Read-only	Read-only	Read-only
F_APP_URI	Read-only	Read-only	Read-only	Read-only
F_CLIENT_AMF_ENCODING	Read-only	Read-only	Read-only	Read-only
F_CLIENT_AUDIO_CODECS	Read-only	Read-only	Read-only	Read-only
F_CLIENT_AUDIO_SAMPLE_ACCESS	Read-only	Read-only	Read-only	Read-only
F_CLIENT_AUDIO_SAMPLE_ACCESS_LOCK	Read-only	Read-only	Read-only	Read-only
F_CLIENT_CONNECT_TIME	Read-only	Read-only	Read-only	Read-only

	E_PUBLISH	E_UNPUBLISH	E_LOADSEGMENT	E_CODEC_CHANGE
F_CLIENT_DIFFSERV_BITS	N/A	N/A	N/A	N/A
F_CLIENT_DIFFSERV_MASK	N/A	N/A	N/A	N/A
F_CLIENT_FAR_ID	Read-only	Read-only	Read-only	Read-only
F_CLIENT_FAR_NONCE	Read-only	Read-only	Read-only	Read-only
F_CLIENT_FORWARDED_FOR	Read-only	Read-only	Read-only	Read-only
F_CLIENT_ID	Read-only	Read-only	Read-only	Read-only
F_CLIENT_IP	Read-only	Read-only	Read-only	Read-only
F_CLIENT_NEAR_ID	Read-only	Read-only	Read-only	Read-only
F_CLIENT_NEAR_NONCE	Read-only	Read-only	Read-only	Read-only
F_CLIENT_PAGE_URL	Read-only	Read-only	Read-only	Read-only
F_CLIENT_PROTO	Read-only	Read-only	Read-only	Read-only
F_CLIENT_PROTO_VER	Read-only	Read-only	Read-only	Read-only
F_CLIENT_READ_ACCESS	Read-only	Read-only	Read-only	Read-only
F_CLIENT_READ_ACCESS_LOCK	Read-only	Read-only	Read-only	Read-only
F_CLIENT_REDIRECT_URI	Read-only	Read-only	Read-only	Read-only
F_CLIENT_REFERRER	Read-only	Read-only	Read-only	Read-only
F_CLIENT_SECURE	Read-only	Read-only	Read-only	Read-only
F_CLIENT_STATS_HANDLE	Read-only	Read-only	Read-only	Read-only
F_CLIENT_SWFV_DEPTH	N/A	N/A	N/A	N/A
F_CLIENT_SWFV_DIGEST	N/A	N/A	N/A	N/A
F_CLIENT_SWFV_EXCEPTION	Read-only	Read-only	Read-only	Read-only
F_CLIENT_SWFV_RESULT	N/A	N/A	N/A	N/A
F_CLIENT_SWFV_TTL	N/A	N/A	N/A	N/A
F_CLIENT_SWFV_VERSION	N/A	N/A	N/A	N/A
F_CLIENT_TYPE	Read-only	Read-only	Read-only	Read-only
F_CLIENT_URI	Read-only	Read-only	Read-only	Read-only
F_CLIENT_URI_STEM	Read-only	Read-only	Read-only	Read-only
F_CLIENT_USER_AGENT	Read-only	Read-only	Read-only	Read-only

	E_PUBLISH	E_UNPUBLISH	E_LOADSEGMENT	E_CODEC_CHANGE
F_CLIENT_USERDATA	Read-write	Read-only	Read-write	Read-only
F_CLIENT_VHOST	Read-only	Read-only	Read-only	Read-only
F_CLIENT_VIDEO_CODECS	Read-only	Read-only	Read-only	Read-only
F_CLIENT_VIDEO_SAMPLE_ACCESS	Read-only	Read-only	Read-only	Read-only
F_CLIENT_VIDEO_SAMPLE_ACCESS_LOCK	Read-only	Read-only	Read-only	Read-only
F_CLIENT_WRITE_ACCESS	Read-only	Read-only	Read-only	Read-only
F_CLIENT_WRITE_ACCESS_LOCK	Read-only	Read-only	Read-only	Read-only
F_MAXFIELD	N/A	N/A	N/A	N/A
F_OLD_STREAM_NAME	N/A	N/A	N/A	N/A
F_OLD_STREAM_QUERY	N/A	N/A	N/A	N/A
F_OLD_STREAM_TYPE	N/A	N/A	N/A	N/A
F_SEGMENT_END	N/A	N/A	Read-only	N/A
F_SEGMENT_START	N/A	N/A	Read-only	N/A
F_STREAM_CODEC	N/A	N/A	N/A	Read-only
F_STREAM_CODEC_TYPE	N/A	N/A	N/A	Read-only
F_STREAM_ID	Read-only	Read-only	Read-only	Read-only
F_STREAM_IGNORE	Read-only	Read-only	Read-only	Read-only
F_STREAM_LENGTH	Read-write	Read-only	Read-only	Read-only
F_STREAM_LIVE_EVENT	Read-write	N/A	N/A	N/A
F_STREAM_LIVE_PUBLISH_PENDING	N/A	N/A	N/A	N/A
F_STREAM_NAME	N/A	N/A	N/A	N/A
F_STREAM_OFFSET	N/A	N/A	N/A	N/A
F_STREAM_PATH	Read-only	Read-only	Read-only	Read-only
F_STREAM_PAUSE	N/A	N/A	N/A	N/A
F_STREAM_PAUSE_TIME	N/A	N/A	N/A	N/A
F_STREAM_PAUSE_TOGGLE	N/A	N/A	N/A	N/A
F_STREAM_POSITION	Read-only	Read-only	Read-only	Read-only
F_STREAM_PUBLISH_BROADCAST	Read-write	Read-only	N/A	Read-only
F_STREAM_PUBLISH_TYPE	Read-write	Read-only	N/A	Read-only

	E_PUBLISH	E_UNPUBLISH	E_LOADSEGMENT	E_CODEC_CHANGE
F_STREAM_QUERY	Read-write	Read-only	Read-only	Read-only
F_STREAM_RECORD_MAXDURATION	N/A	N/A	N/A	N/A
F_STREAM_RECORD_MAXSIZE	N/A	N/A	N/A	N/A
F_STREAM_RESET	Read-only	Read-only	Read-only	Read-only
F_STREAM_SEEK_POSITION	N/A	N/A	N/A	N/A
F_STREAM_TRANSITION	N/A	N/A	N/A	N/A
F_STREAM_TRANSMIT_POSITION	N/A	N/A	N/A	N/A
F_STREAM_TYPE	Read-write	Read-only	Read-only	Read-only

The following table displays all the fields and more events:

	E_RECORD	E_RECORD_STOP
F_APP_INST	Read-only	Read-only
F_APP_NAME	Read-only	Read-only
F_APP_URI	Read-only	Read-only
F_CLIENT_AMF_ENCODING	Read-only	Read-only
F_CLIENT_AUDIO_CODECS	Read-only	Read-only
F_CLIENT_AUDIO_SAMPLE_ACCESS	Read-only	Read-only
F_CLIENT_AUDIO_SAMPLE_ACCESS_LOCK	Read-only	Read-only
F_CLIENT_CONNECT_TIME	Read-only	Read-only
F_CLIENT_DIFFSERV_BITS	N/A	N/A
F_CLIENT_DIFFSERV_MASK	N/A	N/A
F_CLIENT_FAR_ID	Read-only	Read-only
F_CLIENT_FAR_NONCE	Read-only	Read-only
F_CLIENT_FORWARDED_FOR	Read-only	Read-only
F_CLIENT_ID	Read-only	Read-only
F_CLIENT_IP	Read-only	Read-only
F_CLIENT_NEAR_ID	Read-only	Read-only
F_CLIENT_NEAR_NONCE	Read-only	Read-only
F_CLIENT_PAGE_URL	Read-only	Read-only
F_CLIENT_PROTO	Read-only	Read-only
F_CLIENT_PROTO_VER	Read-only	Read-only

	E_RECORD	**E_RECORD_STOP**
`F_CLIENT_READ_ACCESS`	Read-only	Read-only
`F_CLIENT_READ_ACCESS_LOCK`	Read-only	Read-only
`F_CLIENT_REDIRECT_URI`	Read-only	Read-only
`F_CLIENT_REFERRER`	Read-only	Read-only
`F_CLIENT_SECURE`	Read-only	Read-only
`F_CLIENT_STATS_HANDLE`	N/A	N/A
`F_CLIENT_SWFV_DEPTH`	N/A	N/A
`F_CLIENT_SWFV_DIGEST`	N/A	N/A
`F_CLIENT_SWFV_EXCEPTION`	Read-only	Read-only
`F_CLIENT_SWFV_RESULT`	N/A	N/A
`F_CLIENT_SWFV_TTL`	N/A	N/A
`F_CLIENT_SWFV_VERSION`	N/A	N/A
`F_CLIENT_TYPE`	Read-only	Read-only
`F_CLIENT_URI`	Read-only	Read-only
`F_CLIENT_URI_STEM`	Read-only	Read-only
`F_CLIENT_USER_AGENT`	Read-only	Read-only
`F_CLIENT_USERDATA`	Read-write	Read-only
`F_CLIENT_VHOST`	Read-only	Read-only
`F_CLIENT_VIDEO_CODECS`	Read-only	Read-only
`F_CLIENT_VIDEO_SAMPLE_ACCESS`	Read-only	Read-only
`F_CLIENT_VIDEO_SAMPLE_ACCESS_LOCK`	Read-only	Read-only
`F_CLIENT_WRITE_ACCESS`	Read-only	Read-only
`F_CLIENT_WRITE_ACCESS_LOCK`	Read-only	Read-only
`F_MAXFIELD`	N/A	N/A
`F_OLD_STREAM_NAME`	N/A	N/A
`F_OLD_STREAM_QUERY`	N/A	N/A
`F_OLD_STREAM_TYPE`	N/A	N/A
`F_SEGMENT_END`	Read-only	N/A
`F_SEGMENT_START`	N/A	N/A
`F_STREAM_CODEC`	N/A	N/A
`F_STREAM_CODEC_TYPE`	N/A	N/A
`F_STREAM_ID`	Read-only	Read-only
`F_STREAM_IGNORE`	Read-only	Read-only

	E_RECORD	**E_RECORD_STOP**
`F_STREAM_LENGTH`	Read-only	Read-only
`F_STREAM_LIVE_EVENT`	N/A	N/A
`F_STREAM_LIVE_PUBLISH_PENDING`	N/A	N/A
`F_STREAM_NAME`	N/A	N/A
`F_STREAM_OFFSET`	N/A	N/A
`F_STREAM_PATH`	Read-only	Read-only
`F_STREAM_PAUSE`	N/A	N/A
`F_STREAM_PAUSE_TIME`	N/A	N/A
`F_STREAM_PAUSE_TOGGLE`	N/A	N/A
`F_STREAM_POSITION`	Read-only	Read-only
`F_STREAM_PUBLISH_BROADCAST`	N/A	N/A
`F_STREAM_PUBLISH_TYPE`	N/A	N/A
`F_STREAM_QUERY`	Read-only	Read-only
`F_STREAM_RECORD_MAXDURATION`	Read-write	Read-only
`F_STREAM_RECORD_MAXSIZE`	Read-write	Read-only
`F_STREAM_RESET`	Read-only	Read-only
`F_STREAM_SEEK_POSITION`	N/A	N/A
`F_STREAM_TRANSITION`	N/A	N/A
`F_STREAM_TRANSMIT_POSITION`	N/A	N/A
`F_STREAM_TYPE`	Read-only	Read-only

The following table displays all the fields and more events:

	E_SWF_VERIFY	**E_SWF_VERIFY_COMPLETE**	**E_START_TRANSMIT**	**E_STOP_TRANSMIT**
`F_APP_INST`	Read-only	Read-only	N/A	N/A
`F_APP_NAME`	Read-only	Read-only	N/A	N/A
`F_APP_URI`	Read-only	Read-only	N/A	N/A
`F_CLIENT_AMF_ENCODING`	Read-only	Read-only	N/A	N/A
`F_CLIENT_AUDIO_CODECS`	Read-only	Read-only	N/A	N/A
`F_CLIENT_AUDIO_SAMPLE_ACCESS`	Read-only	Read-only	N/A	N/A
`F_CLIENT_AUDIO_SAMPLE_ACCESS_LOCK`	Read-only	Read-only	N/A	N/A

	E_SWF_VERIFY	E_SWF_VERIFY_COMPLETE	E_START_TRANSMIT	E_STOP_TRANSMIT
`F_CLIENT_CONNECT_TIME`	Read-only	Read-only	N/A	N/A
`F_CLIENT_DIFFSERV_BITS`	N/A	N/A	N/A	N/A
`F_CLIENT_DIFFSERV_MASK`	N/A	N/A	N/A	N/A
`F_CLIENT_FAR_ID`	Read-only	Read-only	N/A	N/A
`F_CLIENT_FAR_NONCE`	Read-only	Read-only	N/A	N/A
`F_CLIENT_FORWARDED_FOR`	Read-only	Read-only	N/A	N/A
`F_CLIENT_ID`	Read-only	Read-only	N/A	N/A
`F_CLIENT_IP`	Read-only	Read-only	N/A	N/A
`F_CLIENT_NEAR_ID`	Read-only	Read-only	N/A	N/A
`F_CLIENT_NEAR_NONCE`	Read-only	Read-only	N/A	N/A
`F_CLIENT_PAGE_URL`	Read-only	Read-only	N/A	N/A
`F_CLIENT_PROTO`	Read-only	Read-only	N/A	N/A
`F_CLIENT_PROTO_VER`	Read-only	Read-only	N/A	N/A
`F_CLIENT_READ_ACCESS`	Read-only	Read-only	N/A	N/A
`F_CLIENT_READ_ACCESS_LOCK`	Read-only	Read-only	N/A	N/A
`F_CLIENT_REDIRECT_URI`	Read-only	Read-only	N/A	N/A
`F_CLIENT_REFERRER`	Read-only	Read-only	N/A	N/A
`F_CLIENT_SECURE`	Read-only	Read-only	N/A	N/A
`F_CLIENT_STATS_HANDLE`	N/A	N/A	N/A	N/A
`F_CLIENT_SWFV_DEPTH`	Read-only	Read-only	N/A	N/A
`F_CLIENT_SWFV_DIGEST`	Read/Write	Read-only	N/A	N/A
`F_CLIENT_SWFV_EXCEPTION`	Read-only	Read-only	Read-only	Read-only
`F_CLIENT_SWFV_RESULT`	N/A	N/A	N/A	N/A
`F_CLIENT_SWFV_TTL`	Read/Write	Read-only	N/A	N/A
`F_CLIENT_SWFV_VERSION`	Read-only	Read-only	N/A	N/A
`F_CLIENT_TYPE`	Read-only	Read-only	N/A	N/A
`F_CLIENT_URI`	Read-only	Read-only	N/A	N/A

	E_SWF_VERIFY	**E_SWF_VERIFY_COMPLETE**	**E_START_TRANSMIT**	**E_STOP_TRANSMIT**
`F_CLIENT_URI_STEM`	Read-only	Read-only	N/A	N/A
`F_CLIENT_USER_AGENT`	Read-only	Read-only	N/A	N/A
`F_CLIENT_USERDATA`	Read-only	Read-only	N/A	N/A
`F_CLIENT_VHOST`	Read-only	Read-only	N/A	N/A
`F_CLIENT_VIDEO_CODECS`	Read-only	Read-only	N/A	N/A
`F_CLIENT_VIDEO_SAMPLE_ACCESS`	Read-only	Read-only	N/A	N/A
`F_CLIENT_VIDEO_SAMPLE_ACCESS_LOCK`	Read-only	Read-only	N/A	N/A
`F_CLIENT_WRITE_ACCESS`	Read-only	Read-only	N/A	N/A
`F_CLIENT_WRITE_ACCESS_LOCK`	Read-only	Read-only	N/A	N/A
`F_MAXFIELD`	N/A	N/A	N/A	N/A
`F_OLD_STREAM_NAME`	N/A	N/A	N/A	N/A
`F_OLD_STREAM_QUERY`	N/A	N/A	N/A	N/A
`F_OLD_STREAM_TYPE`	N/A	N/A	N/A	N/A
`F_SEGMENT_END`	N/A	N/A	N/A	N/A
`F_SEGMENT_START`	N/A	N/A	N/A	N/A
`F_STREAM_CODEC`	N/A	N/A	N/A	N/A
`F_STREAM_CODEC_TYPE`	N/A	N/A	N/A	N/A
`F_STREAM_ID`	N/A	N/A	N/A	N/A
`F_STREAM_IGNORE`	N/A	N/A	N/A	N/A
`F_STREAM_LENGTH`	N/A	N/A	N/A	N/A
`F_STREAM_LIVE_EVENT`	N/A	N/A	N/A	N/A
`F_STREAM_LIVE_PUBLISH_PENDING`	N/A	N/A	N/A	N/A
`F_STREAM_NAME`	N/A	N/A	N/A	N/A
`F_STREAM_OFFSET`	N/A	N/A	N/A	N/A
`F_STREAM_PATH`	Read-only	Read-only	N/A	N/A
`F_STREAM_PAUSE`	N/A	N/A	N/A	N/A
`F_STREAM_PAUSE_TIME`	N/A	N/A	N/A	N/A
`F_STREAM_PAUSE_TOGGLE`	N/A	N/A	N/A	N/A
`F_STREAM_POSITION`	N/A	N/A	N/A	N/A

	E_SWF_VERIFY	E_SWF_VERIFY_COMPLETE	E_START_TRANSMIT	E_STOP_TRANSMIT
F_STREAM_PUBLISH_BROADCAST	N/A	N/A	N/A	N/A
F_STREAM_PUBLISH_TYPE	N/A	N/A	N/A	N/A
F_STREAM_QUERY	N/A	N/A	N/A	N/A
F_STREAM_RECORD_MAXDURATION	N/A	N/A	N/A	N/A
F_STREAM_RECORD_MAXSIZE	N/A	N/A	N/A	N/A
F_STREAM_RESET	N/A	N/A	N/A	N/A
F_STREAM_SEEK_POSITION	N/A	N/A	N/A	N/A
F_STREAM_TRANSITION	N/A	N/A	N/A	N/A
F_STREAM_TRANSMIT_POSITION	N/A	N/A	Read-only	Read-only
F_STREAM_TYPE	N/A	N/A	N/A	N/A

Developing a File plug-in

File plug-in overview

The File plug-in gives you control over where and how the server reads content from the file system. The plug-in provides an interface between the operating system's file I/O mechanism and the server. You can configure or modify the sample file to create an alternative to the default operating-system-based file system I/O.

Previous versions of the server supported synchronous access to the file system. Each request for a read operation on a file had to wait for the previous requests in the queue to be completed. The File plug-in supports asynchronous access, making it easier to implement network-based file I/O.

You can code the File plug-in to do the following:

- Grab files from a remote location over HTTP and serve them to clients through the core server process to off-load content management duties.
- Remap files to a different physical location.
- Retrieve external SWF files for verification.

***Note:** The File plug-in works with stream files and SWF files.*

If a custom File plug-in is not present or is inactive, the server uses the standard operating system file system for backwards compatibility. You can only use one File plug-in.

More Help topics

Flash Media Server Plug-In API Reference

Responding to server calls

The File plug-in is asynchronous—when the server calls the plug-in, the plug-in doesn't respond immediately. The server calls the plug-in, and the plug-in calls the server back on the response interface.

When the server calls a function on the plug-in, the plug-in interface can return an error code (-1) indicating that the operation failed. If the function returns an error, the plug-in should not call the server back.

If a call to the plug-in returns successfully (returns 0), the plug-in should call the server back and pass the context received from the server.

The `close()` and `remove()` calls are an exception to this rule. When the server calls the `close()` and `remove()` functions on the plug-in, the server is not interested in the response. To free resources associated with the request before the callback occurs, the server sometimes passes `NULL` as the `pCtx` pointer. If the context is `NULL`, then it is not necessary to call the server back. If the plug-in calls the server back with the `NULL` context, the server ignores it.

Retrieving external SWF files for verification

Verifying SWF files ensures that only legitimate applications can access this instance of Adobe Media Server. This feature prevents third parties from creating their own applications that attempt to stream your resources.

In Content Distribution Networks, the SWF files that you want to verify reside in an external content repository or on another server in a cluster. You can use the File plug-in to retrieve SWF files that are stored in external locations so that the server can verify them. By using the File plug-in, SWF files do not have to reside locally on Adobe Media Server. Developers can update SWF files frequently without impacting the server. Using the File plug-in facilitates and simplifies content management by developers.

When using the File plug-in, the server verifies SWF files at the file level, regardless of the application. You cannot enable SWF verification through the File plug-in on a per-application basis.

Note: *The ability to retrieve SWF files using the File plug-in is only available in version 1.0 of the plug-in (Adobe Media Server 3.5 or later).*

To verify SWF files through the File plug-in, follow this workflow:

- Enable SWF verification in Application.xml.
- Enable SWF verification in Server.xml.
- Implement the File plug-in.

Enable SWF verification in Application.xml

1 In an XML editor, open Application.xml.

2 Set the `enabled` attribute in the `SWFVerification` tag to true.

```
<SWFVerification enabled="true">
```

This tag enables verification for all applications, regardless of whether this file is the vhost-level or application-level XML file.

3 Leave the `SWFFolder` tag empty. By doing so, the server passes the default `SWFFolder` value to the plug-in. The default value is the application's folder appended with SWFs. For example, applications/*application_name*/SWFs.

Your File plug-in must redirect this value to your external content repository.

4 (optional) Configure additional verification tags.

Enable SWF verification in Server.xml

1 In an XML editor, open Server.xml.

2 In the `Plugins` tag, set the `enabled` attribute in the `FilePlugin` tag to true:

```
<Plugins>
    <FilePlugin enabled="true">
        <Content type="Streams">true</Content>
        <Content type="SWF">false</Content>
    </FilePlugin>
</Plugins>
```

3 To enable verification of SWF files, set the `<Content type="SWF">` attribute to `true`.

4 (optional) To configure global verification, leave the `SWFFolder` tag empty. By doing so, the server passes the default `SWFFolder` value to the plug-in. Your File plug-in must redirect this value to your external content repository.

With global verification, you can configure verification for a group of SWF files that are common to all applications.

5 (optional) Configure additional verification settings: `DirLevelSWFScan`, `MaxNumberofRequests`, and any desired user-defined keys in the `UserDefined` element. See the XML reference in the *Configuration and Administration Guide*.

Implement the File plug-in

The File plug-in can open directories or files in external locations and pass a list of directories and files to the server. The File plug-in does not contain a specific attribute to distinguish stream files from SWF files. The server performs verification like it does when SWF files are stored on the server.

See the File plug-in sample included with the server. Some highlights of the code are presented here.

The File plug-in sample file includes a function that calls `IFmsFileAdaptor::open`. The `open()` function includes a parameter, `sFileName`. The value of `sFileName` is passed to the File plug-in by the server; it is the default value of the `SWFFolder` tag in either Application.xml or Server.xml.

The plug-in redirects the `SWFFolder` value to the external content repository. The sample creates a handle to that repository and passes it back to the server. The `getAttributes()` function also takes `sFileName` as a parameter.

After calling the `open()` function, the sample includes code that determines if the value of `sFileName` is a file or a directory. Files are opened. Directories are opened and written to a temporary file. The sample uses the new attribute `FMSFileAttribute::kMode` to determine if `sFileName` is a file or directory (1 for a stream, 0 for a directory.)

The File plug-in returns a handle to this temporary file to the server. If the directory contains any subdirectories or SWF files, the server opens the directory or file, and continues the process until all the subdirectories and files are read. Through the plug-in's `read()` method, the server reads the file and performs verification. Through the plug-in's `close()` method, the server closes the file when fully read.

Use the File plug-in to manage content for live HTTP streaming

Adobe Media Server 4.5

Use the File plug-in to manage content for live HTTP streaming, including asynchronous file IO operations. The live packager application (livepkgr) ingests a live stream and packages it into fragments (F4F files) and additional helper files. The server operations that record these files are routed through the File plug-in. Use the File plug-in to manage the location of the following files:

- .f4f

- .f4x
- .bootstrap
- .control
- .drmmeta
- .meta

Adobe Media Server 4.5 adds the following APIs:

- `rename()`

  Moves a file. Pass the absolute path of the source file and the destination file. This call overwrites a file of the same name in the destination path.

  The server calls this method on the File plug-in synchronously or asynchronously. If the server calls this method asynchronously, call `onRename()` to resume operations to the server.

- `truncate()`

  Truncates a file to a specified file size. The server calls this method on the File plug-in synchronously or asynchronously. If the server calls this method asynchronously, call `onTruncate()` to resume operations on the server.

- `onRename()`

  Calls the resume operation on the server after the server has called `rename()` on the File plug-in asynchronously.

- `onTruncate()`

  Calls the resume operation on the server after the server has called `truncate()` on the File plug-in asynchronously.

For detailed information about each API, see Flash Media Server Plug-in API Reference. In addition, these APIs are used in the sample File plug-in installed to *rootinstall*/samples/plugins/file/SimpleFileAdaptor.h.