



HELIX MOBILE PRODUCER USER'S GUIDE

Helix™ Mobile Producer 11 powered by Envivio™

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INTRODUCTION

Welcome to the *Helix Mobile Producer User's Guide* from RealNetworks®. This manual shows you everything you need to know about this product, which enable you to convert audio and video into streaming media for wireless devices such as mobile telephones and personal digital assistants (PDAs).

Using Helix™ Mobile Producer powered by Envivio™, anyone can create streaming media easily from a variety of sources. You can convert from existing audio or video files, record directly from audio/video (A/V) devices, or use these applications together with Helix Server to broadcast and stream live content.

What is Helix?

Helix from RealNetworks is a universal digital media delivery platform. With industry-leading performance, integrated content distribution, advertising, user authentication, Web services support, and native delivery of many types of media files (RealMedia™, Windows Media, QuickTime, MPEG-4, and 3GPP), Helix from RealNetworks is a robust digital media foundation that meets the needs of businesses and networking service providers.

Helix Mobile Producer

With Helix Mobile Producer, you can format your video and audio content to reach the world of wireless devices. This innovative encoding software enables mobile operators and content providers to convert streaming media to a full spectrum of standards for mobile products. Helix Mobile Producer is a key component of RealNetworks' end-to-end solution (encoding, serving, and playback) for mobile multimedia, supporting both on-demand and live

encoding and making it possible to generate content with the bit rates, frame sizes, and other settings most suitable for mobile devices.

Standard and Professional versions

Helix Mobile Producer gives you the ability to encode streaming audio or video presentations as files. Helix Mobile Producer Live is delivered in two versions, Standard and Professional. The Professional edition includes the following additional features:

- Command-line interface
- SNMP monitoring
- Batch encoding through the graphical interface

Why Use Helix Mobile Producer?

When it comes to 3GPP mobile encoding, Helix Mobile Producer does it all. This product is ideal for either live or on-demand content and will convert most common types of video and audio to standards-compliant 3GPP formats. This section discusses the features and capabilities of this software in terms of two key areas: quality and productivity.

Quality

Helix Mobile Producer provides support for a range of encoding technologies that enable high-quality media delivery, using the features described in the following paragraphs.

MPEG-4 SVP (Simple Video profile) video

Stream or download high-quality video to users' cell phones or PDAs, at bit rates from 10 to 384 kilobits per second (Kbps). Helix Mobile Producer supports the following MPEG-4 SVP levels:

level 0	up to 64 Kbps
level 0b	up to 128 Kbps
level 1	up to 64 Kbps
level 2:	up to 128 Kbps
level 3:	up to 384 Kbps

H.263 Profile 0 video

This is another standards-based codec that you can use to deliver good-quality video at bit rates from 10 to 384 Kbps, but without the licensing requirement of MPEG-4. Helix Mobile Producer supports the following H.263 Profile 0 levels:

level 10:	up to 64 Kbps
level 20:	up to 128 Kbps
level 30:	up to 384 Kbps
level 45:	up to 128 Kbps

H.264

Helix Mobile Producer supports the following H.264 baseline profile levels:

level 1:	up to 64 Kbps
level 1b:	up to 128 Kbps
level 1.2:	up to 384 Kbps
level 1.3	up to 768 Kbps. Note that the encoder limit for this codec is 384 kbps. 768 kbps reflects the limit for the level

Variable-bit-rate encoding

This feature enables the video codec to vary the bit rate throughout a clip, depending on the type of content being encoded. With variable-bit-rate encoding, the more action there is in a scene, the more bits are used for that scene. This makes for a more consistent quality of video encoding for downloadable content, while keeping download times as short as possible.

2-pass video compression

Double-pass encoding increases output quality by analyzing video data before encoding the input video.

AMR audio

Using this feature, you can deliver high-caliber voice clips at bit rates ranging from 4,750 to 12,200 bps.

AAC-LC audio
AAC Plus audio
Enhanced AAC Plus audio

This feature enables you to deliver music and mixed-audio content at bit rates from 8 to 128 Kbps. AAC provides better-quality audio than does MP3 at the same bit rate.

Video scaling

You can use imported video files to scale standard screen resolutions ranging from QQCIF (88 x 72) to CIF (352 x 288).

Prefiltering

By cropping or making adjustments in frame-rate conversion, inverse-telecine, deinterlacing, noise filtering, brightness, contrast, and gamma settings, you can improve video quality.

Productivity

Helix Mobile Producer increases your productivity by providing support for a number of media formats and tools that make automated media encoding possible.

Input file support

With DirectShow and QuickTime 6 installed, Helix Mobile Producer can read a variety of file types, including AVI, MOV, DV, MPEG-1, MPEG-2 (with an additional DirectShow or QuickTime plug-in), MP3, MPEG-4, and WAV files.

XML job files

By using XML job files, you can define all of the settings for a given file-to-file encoding job, making it easy both to edit the files and to reuse the same settings for batch processing.

Command-line interface

A simple command-line interface enables you to create batch-processing scripts or to wrap your 3GPP encoding into your own customized, automated system.

Additional Documentation Resources

In addition to this manual, you might want to consult the following RealNetworks books, the first three of which are available for downloading at

the following Web address:

<http://service.real.com/help/library/encoders.html>.

- *Introduction to Streaming Media*

Start with this guide if you are new to streaming media or RealNetworks products. Written for the beginning user, this book explains how to put together a basic presentation using different production techniques.

- *RealNetworks Production Guide*

This guide is the main reference manual for streaming media production. Refer to the production guide for instructions and tips on producing audio and video clips, as well as for complete information about using the Synchronized Multimedia Integration Language (SMIL).

- *RealPlayer Scripting Guide*

If you are a Web programmer, refer to this guide for instructions about using JavaScript or VBScript with RealPlayer™ from RealNetworks. Using these scripting languages, you can customize RealPlayer to turn it into your own Internet jukebox, for example.

- *Helix Server Administration Guide*

The basic reference for the Helix Server, this guide explains how to set up, configure, and run Helix Server to stream multimedia. You need this guide only if you are running Helix Server yourself. You can download this manual from the following Web address:

<http://service.real.com/help/library/servers.html>.

Technical Support

For general information about RealNetworks Technical Support, visit the following Web page:

<http://service.real.com>

STREAMING MEDIA BASICS

This chapter introduces you to streaming media and how you can use Helix Mobile Producer to create streaming media. It gives you a brief look at how streaming works, the different types of media that you can create with Helix Mobile Producer, and the various RealNetworks products that you use when streaming.

What is Streaming Media?

Before the advent of streaming media, users had to wait for media files, or clips, to be downloaded from the Internet or from a network server before they could play the clips. With streaming media, users can see and hear these clips almost instantaneously.

A streaming clip consists of small packets of information that are sent over a network connection. The user receives these information packets in a “stream” and, using a player (such as RealPlayer from RealNetworks), experiences the media piece by piece.

The mechanics of the streaming process are virtually invisible to the user. In fact, the process is similar to viewing a film, with each data packet being analogous to a single frame in a filmstrip. When a film is run through a projector and displayed on a screen, the audience is not aware of each individual frame they are seeing; rather, they experience the film as one continuous flow. Similarly, when users receive and play streaming media clips on their computers, what they experience is a continuous stream rather than a succession of discrete data packets.

How is Streaming Media Created?

Helix Mobile Producer creates streaming media data packets by a process called “encoding.” During encoding, the source media is transformed into

streaming media through the use of “codecs” (compression/decompression algorithms). The entire process is summed up in the following four steps:

1. Helix Mobile Producer receives the source media as a file, or as live audio or video.
2. Helix Mobile Producer uses a codec to compress the media source's data into packets.
3. The data packets are streamed to the user over the Internet or an intranet.
4. At the user's end, the same codecs are used to piece the media back together so that the user can play the clip.

Working with Audio and Video Clips

Although for you the steps involved in encoding streaming audio are similar to those involved in encoding video, there are some basic differences in the way Helix Mobile Producer processes audio and video streams.

Encoding audio is simpler than encoding video. A basic streaming audio clip is created by using an audio file or live audio source as the input. Helix Mobile Producer uses various audio codecs to convert your standard audio input into a format that can be streamed.

A more complex task for Helix Mobile Producer is converting standard video input into streaming media. A video clip is created either by converting an existing video file or by capturing a live video source, such as from a video camera or a VCR, and sending it to your computer by way of a video capture card. Helix Mobile Producer converts different attributes of the video—such as frame rate, type of motion, and size of the image—into a video clip by using a video codec. If the video also includes audio data, that must also be converted by the audio codecs.

Targeting Audiences

Before Helix Mobile Producer can compress the source data, it needs to know something about the intended audience for the resulting media clip. An audience is defined by the bit rate at which their computers can connect to the Internet or a given network. For example, a person using a 56-Kbps dial-up modem to connect to your media stream is a member of the 56K Modem audience.

Single-Bit-Rate Streaming



Because some data is lost during the compression process, picking the correct audience is key in preserving as much of your source data as possible.

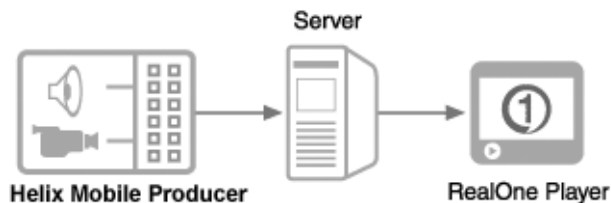
Other RealNetworks Products

Helix Mobile Producer is a member of the RealNetworks family of software products. These components work in tandem with one another to create, stream, and play your media clips:

- Helix Mobile Producer creates streaming clips for mobile devices.
- Helix Server streams the media clips to users.
- RealPlayer enables users to play the streamed media clips.

The following diagram illustrates how these RealNetworks products work together.

RealNetworks Software Products



Helix Server

Just as a Web server delivers pages to Web browsers over the Internet, Helix Server serves streaming media clips to your audience. It enables users to stream the media clips rather than download them. By streaming the content, a user can begin to watch the clip almost immediately and doesn't have to wait for the entire file to be downloaded.

There are two different ways to deliver your content: as an on-demand file or as a live broadcast. On-demand delivery entails encoding your media content before your audience needs it. With live broadcasting, on the other hand, the media is being encoded at the same time that your audience is playing it. For both types of delivery, you can use Helix Server in conjunction with Helix Mobile Producer.

Client Software

A client, such as RealPlayer installed on a mobile device, plays the streamed media. Note that with Helix Server, you can also deliver streaming content to computers running Windows Media Player or QuickTime as their client (player) software.

SYSTEM REQUIREMENTS AND INSTALLATION

This chapter discusses the hardware and software requirements for Helix Mobile Producer, and it explains how to install the products on a computer running Microsoft Windows.

Note: Windows 2000 and Windows XP are the only operating systems supported for use with Helix Mobile Producer.

System Requirements

The following table lists the hardware and software requirements and specifications for installing and running Helix Mobile Producer or Helix Mobile Producer Live on your computer.

**Helix Mobile Producer and Helix Mobile Producer Live
System Requirements and Specifications**

Category	Item	Requirements or specifications
Minimum system requirements and support	Microsoft Windows operating system	Windows 2000 or Windows XP Pentium III equivalent or greater 128 MB of RAM (256 MB recommended) NTFS is required for large files that cannot be handled by a FAT32 file system DirectX 8.1 or later Optional Requirements: DirectShow MPEG-2 Decoder to import MPEG-2 and QuickTime 6.0 or later to import MOV and MP4 files (QuickTime 6.5 recommended)
	Graphics card	Color graphics card capable of 1024 x 768 resolution
	Networking	Ethernet card
Supported import formats	Video	AVI, MOV, MPEG-1, MPEG-2, MPEG-4, DV
	Audio	AIF, MP3, WAV

Helix Mobile Producer and Helix Mobile Producer Live System Requirements and Specifications (continued)

Category	Item	Requirements or specifications
Supported capture devices	Osprey 100	Video: S-Video and Composite Audio: None
	Osprey 200/210/220	Video: S-Video and Composite Audio: Analog RCA and XLR
	Osprey 500 DV/500 DV Pro	Video: S-Video and Composite, IEEE 1394/DV, and SDI Audio: AES/EBU
	Osprey 540	Video: S-Video and Composite, IEEE 1394/DV, and SDI Audio: Analog RCA and XLR, AES/EBU, and Embedded SDI Audio
	Techsmith Camtasia	Video: Screen capture only
	Pinnacle PCTV Rave	Video: S-Video and Composite Audio: RCA
	Philips ToUcam Pro	Video: Webcam Audio: mono, 16-bit, 8 / 11.025 / 22.05 / 44.1 kHz
Export	File format	3GPP, 3GPP2, 3GPP v6 MPEG-4 files, optionally prepared for RTP streaming AMR MP3 QCP (QCELP file format) RealMedia

Warning! We do not support running of our encoders with a third party encoder(s) on the same machine. The primary reason is that other encoders do not have the concept of resource sharing. Unlike Real encoders, which will adapt to a constrained system environment by encoding at lower complexity, or encoding fewer frames, the third party encoders will dominate the system at the expense of our encoders, therefore producing a bad result.

Installing Helix Mobile Producer

This section explains how to install Helix Mobile Producer on your computer and how to register your copy of the product with RealNetworks.

To install Helix Mobile Producer, you must have at least a “Power User” access level.

Note: You should have received your license files through email. If you are installing a trial or demo version, these trial license files will expire 30-60 days after you receive them. If you are installing a purchased copy, your license files will never expire.

► **To install and register Helix Mobile Producer:**

1. Double-click the icon for the installer file, **helix_mobile_producer_11_windows_setup.exe**, then read and follow the online instructions displayed during the installation process.
2. Specify the path to the licensing file where indicated, and then click **Next**, as shown in the following illustration.



3. Follow the rest of the installation instructions.

Installing the SNMP Component

If you do not already have the Windows SNMP service installed, you will need to install it from the Windows installation CD.

Note: The SNMP monitoring component of Helix Mobile Producer is only available with Professional level licenses.

If you are installing a Professional license and wish to install SNMP, we recommend that you manually halt the SNMP service before installing, then restart the service after installation is complete.

► **To manually halt the SNMP service:**

1. From the Windows **Start** menu, go to **Programs>Administrative Tools** and open the **Services** application.
2. If SNMP is installed on your system, there will be two SNMP services named "SNMP Service" and "SNMP Trap Service".
From the **Services** list, select these two services (one at a time) and click the **Stop Service** button in the toolbar, or select **Stop** from the **Action** menu.
3. Run the Helix Mobile Producer installer (You can leave the **Services** application running). Be sure to check the **Install SNMP** check box on the third panel of the install.

After the installation is complete, you can restart the SNMP services.

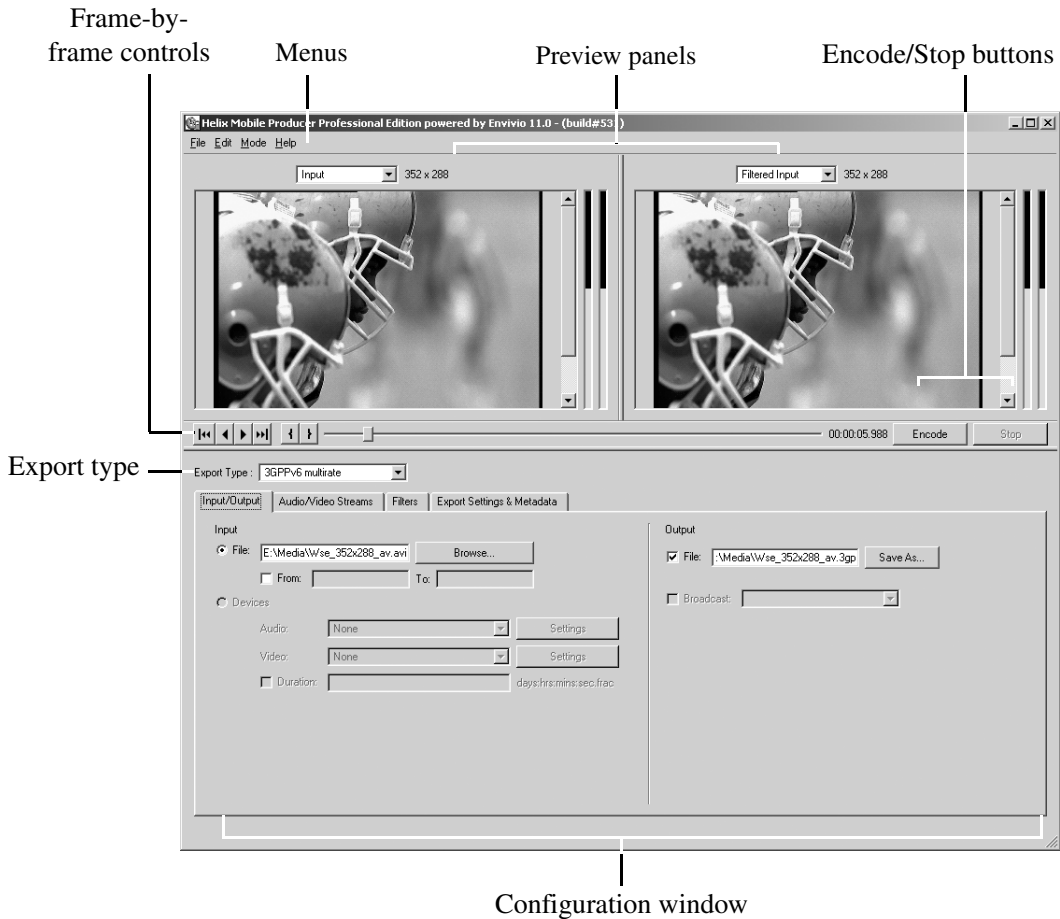
QUICK START

This chapter explains how to start Helix Mobile Producer. It introduces key concepts that will help you encode audio and video from prerecorded files or live media sources. It also explains the basic workflow for encoding a media.

Starting Helix Mobile Producer

To start Helix Mobile Producer, double-click the **Helix Mobile Producer** icon or use the Windows **Start** menu. Programs are listed directly under **Start>Programs>Helix>Helix Mobile Producer**.

Helix Mobile Producer Workspace



The Helix Mobile Producer workspace is composed of several panels, which are described in the following paragraphs.

- The upper window is divided into two preview panels. Each panel may display one of the following:
 - a preview of the original video input.
 - a preview of the filtered input
 - a preview of the encoded output after application of the selected audience(s).
 - or, no preview
 - ...and the left/right audio channels
- The horizontal bar in the middle of the workspace displays the frame-by-frame controls and the encoding progression bar. It contains the **Encode**, and **Stop** buttons.
- The lower window allows selecting the export type and has tabs for configuring input/output, streams, and filters parameters, and for defining export settings and Metadata.

Quick Concepts

This section describes key concepts that will help you understand how to use Helix Mobile Producer effectively.

Media Sources

A Helix Mobile Producer presentation can include one digital video source, one digital audio source, or both. Note that input media sources can be either existing files or media that's "captured" live.

Jobs

The job defines how to encode your clip or broadcast. It selects the media inputs, sets the encoding properties, and defines the destinations, such as whether to save the encoded stream to a clip, forward it to a server for broadcast, or both.

For More Information: Chapter 6 explains how to define jobs through the graphical application. See Chapter 9 for information about specifying jobs on the command line. Appendix B explains the job file syntax.

Audiences

When you set up a job, you choose one or more audiences. An audience defines several aspects of the encoding job. Most importantly, it defines the bit rate at which a clip or broadcast streams. For a downloadable clip, the audience selection affects the clip's quality and file size. Selecting an audience is therefore one of the most important decisions you make when encoding a clip. Helix Mobile Producer predefines a number of audiences designed to fit a variety of streaming and downloading requirements.

For More Information: Chapter 5 explains how to select audiences through the graphical application. Chapter 9 does the same for the command-line application. You can also create your own audiences, as explained in Appendix A.

Encoding

Encoding compresses digital media so that it takes up less space and can be transmitted faster. The compression process identifies the essential components of the input media and discards the superfluous parts.

Be aware that the addition of data such as hint tracks and headers to audio or video streams during the encoding process results in the encoded files being somewhat larger than the sum of the encoded media bit rates multiplied by the duration of the file ((audio bit rate + video bit rate) x duration). This does not affect the bit rate or quality of the media, because this additional data remains on the server and is not streamed to the client.

Note: The hint track is required by the streaming server to optimize the streaming experience. A 3GPP media file must have a hint track to be properly streamed from most streaming servers. The hint track is not required if the media file is to be downloaded and played locally, however. To avoid having Helix Mobile Producer add a hint track to such a file, clear the **Prepare for streaming** check box on the **Export settings and Metadata** tab before encoding the audio or video clip.

Getting Started

This section outlines the main steps involved in preparing to encode audio or video input: setting up input for offline or live encoding, choosing an audience, and setting output options.

Basic Workflow

Perform the sequence of actions outlined in this section to encode media for file output or broadcast output by using Helix Mobile Producer.

► **To encode media with Helix Mobile Producer:**

1. Set the export type
 - The export type defines the selectable inputs and outputs, audiences and encoder types.
2. Set input parameters:
 - Select an input media source: file (offline encoding) or capture (live encoding).
 - For file input, select the file's path. (If your input is an AVI file, you would then choose how you want to open AVI files.)
 - For captured input (for example, camera or VTR), select audio and video input sources, and then set the duration.
3. Set output options:
 - Specify the destination for file output.
 - Specify the parameters for broadcast output.
4. Select audience(s):
 - Select one or more audience from the Audience template list, then adjust the values if necessary.
5. Set the filters
6. Set the export parameters and metadata.
7. Encode the input media (file or live).

Setting Up the Export Type

You need to select a target export type to define the selectable inputs and outputs, audiences and encoder types. Selecting an export type restricts choices regarding encoder type and number of audiences, unavailable parameters are grayed out.

You can export encoded files in any format listed in the following table.

File Formats Supported for Exporting

Format	Description
3GPP v5 (single rate)	European third generation partnership project file format, release 5. Includes definition of the IP-Multimedia System (IMS) and High Speed Downlink Packet Access (HSDPA), assuring users full multimedia application capabilities.
3GPP v6	European third generation partnership project file format, version 6. Includes networking with Wireless Local Area Networks (WLAN), Presence, speech enabled services, interpretability with other network technologies, and use of new radio frequencies.
3GPP2 (single rate)	North American and Asian third-generation partnership project file format
ISMA Profile 0	MPEG-4 ISMA 1 profile 0 format
AMR	Adaptive multirate speech codec
QCP	Speech codec
MP3	MPEG-1 audio layer 3 file format
RealMedia	RealNetworks media file format

Setting Up Input and Output

Setting Up Source Files for Offline Encoding with Helix Mobile Producer

You need to specify a source file before you can start encoding offline input media with Helix Mobile Producer.

Selecting a Source File

Use the following procedure to specify an input source file that you want to encode.

► To select a source file:

1. Click **File>Open Input File**.
2. Select an input source file.

Note: When you select a video source, the preview of the input video is displayed in the **Input** preview panel. If you select an audio source, a headphones icon is displayed.

Setting Up Capture Sources for Live Encoding

You need to specify live input sources before you can encode live media.

Note: Live encoding is only available with Helix Mobile Producer Live.

Selecting a Media Source

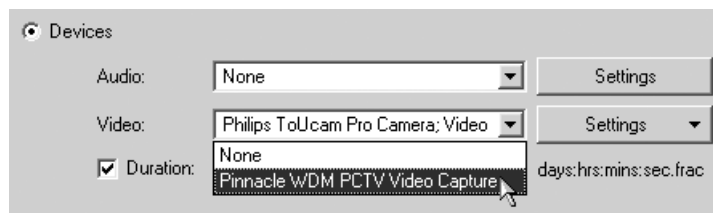
The first thing you must do if you want to encode a live media stream is to decide what source you will use for your input media, and then select that source in the Input panel in the Helix Mobile Producer workspace.

► To set up capture sources:

1. Select a video source from the list of available video capture devices.

Note: Only the first ten detected inputs are displayed.

Selecting a video capture device

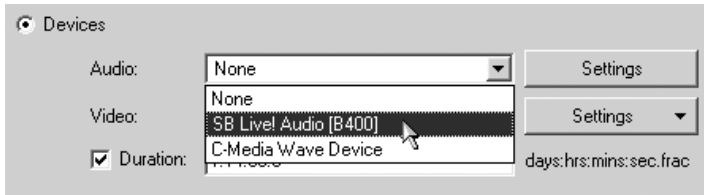


Note: When you select a video source, a preview of the input video is displayed in the Preview panel. Note that no output preview is displayed for RealMedia encoding.

2. Select an audio source from the list of available audio capture devices.

Note: Only the first ten detected inputs are displayed.

Selecting an audio capture device



Note: For any given instance of live media that you want to encode and broadcast, you might need to select only a video capture source or only an audio capture source, and not both.

Setting Output Options

Using Helix Mobile Producer, you can encode input audio and video as output files or live broadcasts.

File Output

You can specify the path where you want to generate the output file by clicking the **Browse** button in the Output panel and then selecting the path to the output directory.

Live Broadcast Output

You can use a streaming media server as a destination for encoded output. When you use this method, the encoding and transmission of live media directly to the streaming server and then straight to your audience occur simultaneously. This is called live broadcasting.

You can use either of two modes for live broadcasts:

- RTP broadcast mode (for 3GPP, 3GPP2, and MP4 only)
- Helix broadcast mode (for RealMedia only)

For More Information: Refer to “Chapter 7: Broadcast” beginning on page 33.

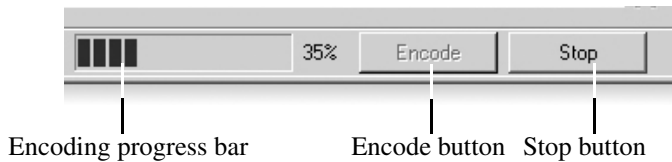
Encoding Input Media

As soon as you have specified the source media and set up encoding preferences by selecting a template and adjusting encoding parameters (as explained earlier in this chapter), you are ready to start encoding your media.

Starting to Encode

To begin encoding, click the **Encode** button below the Output panel on the Helix Mobile Producer workspace. When encoding begins, the progress bar indicates the encoding status and the elapsed time, as shown in the following illustration. Any time you want to stop the encoding process, simply click the **Stop** button to the right of the **Encode** button.

Control Buttons and Progress Indicators for Encoding



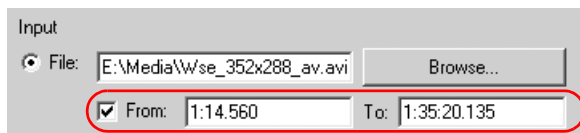
Partial Encoding

By default, Helix Mobile Producer encodes the entirety of whatever input media it receives. There are, however, options you can use to encode only part of an input file or to encode live input for only a specified amount of time.

Partially Encoding an Offline Media File

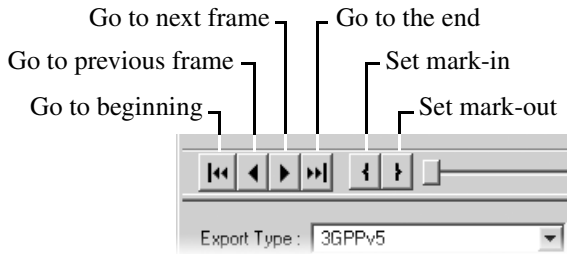
To encode only a part of an input file, select the **From** check box and then type the start and end points you want for the encoding, as shown in the following illustration. Note that these points are measured from the beginning of the input file, in tenths of a second.

Partial encoding of offline media



Another option is to use the **Set mark-in** and **Set mark-out** buttons to specify graphically the start point and end point for the encoding, respectively. See these and other buttons in the following illustration.

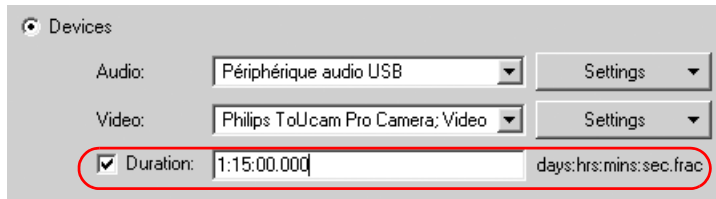
Frame-by-Frame Controls



Time Limited Live Broadcast

After you have set the encoding process in motion for live media, Helix Mobile Producer will, by default, continue encoding the live stream until either the **Stop** button is clicked or the specified duration period comes to an end. To set this latter option, select the **Duration** check box and then type whatever duration period you want, as shown in the following illustration.

Setting the Duration of a Live Broadcast



AUDIENCES

Before you encode a clip or broadcast, you choose the audience or audiences to use. For each audience, Helix Mobile Producer encodes a separate stream based on the speed of the network connection or a certain level of quality that you want to preserve. This chapter explains how to use *audiences*.

Working with Audiences


An audience defines a range of parameters used to encode a clip or broadcast. It defines the audio codecs, the video codec, the maximum frame rate, and the maximum streaming bandwidth, for example. You can use the same audience to encode a video clip or an audio-only clip.


Adding an Audience to a Job

Helix Mobile Producer provides a number of predefined audiences designed to fit a variety of streaming and downloading requirements. You can add an audience template to the encoding job, or remove it.

Note: You can only add multiple audiences to 3GPP v6 Multirate and RealMedia export types.

The lower panel of the Audiences palette lists the available audience templates. Each template has a name that describes the audience, such as GPRS News, and may list the template's average bandwidth usage. The upper panel of the palette lists the audiences chosen for the job.

- ▶ To add an audience to the job:
 1. Click the audience name in the lower panel.
 2. Click the **Up arrow** button, .

Note: To add multiple audiences to the same job, you can choose only the audiences not listed as variable bit rate (VBR) audiences. To delete an audience from the job, highlight the template name in the upper panel, and click the **Delete** button, .


Tip: To view the properties for an audience added to the job, click the audience name. The audience properties are displayed in the right panel.

Creating a New Audience for the Active Job

Creating a new audience

Note: If it is not a 3GPP v6 Multirate or a RealMedia export type, you must first remove the existing audience before adding a new audience.

► To create a new audience:

1. In the upper panel, click the **New** button, .
2. Modify this audience.

Choosing a Template Name

When you edit a template or create a new one, you can specify a name by double-clicking on the audience name. The graphical application uses this name to identify the template.

You can also use this name to specify the audience with the command-line application. Helix Mobile Producer uses the template name as the file name for the audience file, appending the .xml extension automatically.

Changing Audience Settings


You can edit any audience template to change it permanently. You can also create a new template from an existing one to define your own audiences.

Note: These changes are recorded in the template and the active job, but not in any previous jobs that also used the audience template. To update an older job, edit the job file manually. Or, import the job in the graphical application,

delete the existing audience or audiences, then add the new audiences.

Changing Audience Values for the Active Job


To change the audience properties for an active job:

1. Add the audience to your job as described in “Adding an Audience to a Job” on page 25.
2. Click the audience name in the upper audiences palette to display the audience properties.
3. You can then change the audience settings.
4. Optionally, you can click the **Up arrow** button,  to save the modified audience as a template.

Editing, creating or deleting an Audience Template


1. Select an audience template from the list. This displays the template's settings in the right-hand side of the window.

Tip: Within the list, templates are ordered according to the audience's average streaming speed or quality setting.

2. You can then perform any of the following actions:
 - a. Edit the settings for the template as described in Chapter 8.
 - b. Click the **'Delete'** button,  to delete the selected template.

Deleting an Audience

► To delete an audience:

1. Select the audience you want to delete.
2. Click the **Delete** button, .

Audience Files

Helix Mobile Producer stores audience information in an XML-formatted audience file that you can edit by hand. You can also copy and modify an existing file to create your own audience definition.

Appendix A explains the audience file syntax. Refer to Appendix B for information about how to copy audience file information into a job file.

Bit Rates in Audiences

Bit rates are stated in kilobits per second (Kbps). There are several parameters that you must consider when creating an audience:

- You need to know the channel capacity supported by the connection.
- The maximum streaming bit rate is the highest bit rate that the *bit-rate smoother* allows for audio or video transmission by a streaming server. The maximum streaming bit rate must be less than the channel capacity.
- The combined media bit rate is the sum of the audio bit rate and the video bit rate in a presentation.
- The audio bit rate is the specified bit rate for an audio stream.

Audience Considerations

You can adjust the job parameters and create specific audiences that take your content and broadcast requirements into account.

Consider the following criteria when developing a job:

- The type of content
- The size and length of the presentation
- The level of video quality you want
- The level of audio quality you want
- The available bit rate
- The amount of time required for encoding

SureStream Audiences for RealMedia Clips

When you choose to encode a clip for a particular audience, you are telling Helix Mobile Producer to create a stream for that audience.

Note that although you can choose more than one audience for an encoding job, the amount of encoding time required and the size of the encoded output will increase with every audience stream that is added. For that reason, generally it is recommended that you select no more than two or three audiences that you know you need for a given clip.

JOBS

A job is a central aspect of encoding with Helix Mobile Producer. The job defines how to encode your clip or broadcast. It selects the media inputs, sets the encoding properties, and defines the destinations, such as whether to export the encoded stream to a clip, forward it to a server for broadcast, or both.

Working with Jobs

Each time you encode a clip or broadcast, Helix Mobile Producer creates a job that records the encoding settings. You can export each job to a separate job file, which is an XML formatted text file that you can manually edit as described in Appendix B. Once you have exported a job file, you can import it to apply the same settings to another clip or broadcast. Although using a job file is optional, you'll find job files highly effective if you encode a lot of clips or broadcasts.

The Structure of a Job

Whether you use the graphical application or the command-line application, you can export your job settings in a job file for later use. You can also modify a job file's settings, either through Helix Mobile Producer or by hand, to create a new job quickly. The job file uses an XML-based format to define all of the job's encoding settings.

For More Information: For information about specifying jobs on the command line. Appendix B explains the job file syntax.

Creating a New Job File

Whenever you encode a clip or broadcast, Helix Mobile Producer creates a job automatically, letting you choose whether or not to export the job settings to a file. You can also create a job file without actually encoding any streams. This

allows you to define job profiles to use later. The following steps describe the basic procedure for creating, editing, and saving a job file.

► **To create a new job:**

1. Start a new job file by choosing **File>New Job** or pressing **Ctrl+N**. Specify the export type and the encoding properties (input type, output type, audio/video tracks). A new, untitled job is created.
2. Define your encoding settings. You can set up your audiences, define clip information, turn on video filters, create input and output file names, and so forth.
3. You can export your job by choosing **File>Export Job**. The job file is automatically exported with the file extension `.xml`.

Warning! Give the job file a descriptive title that will help you to remember the purpose of the job. For example, you might name a job file `GeneralVideoForDialUpModems.xml`.

Note: You can share an exported job by giving the job file to another Helix Mobile Producer user. If you defined inputs and outputs, the user needs access to those same sources and destinations to run the job.

Using and Modifying Existing Jobs

Once you have exported a job file, you can use it as the basis for other encoding jobs. Suppose that a certain job file has all the basic settings you want, but for a certain clip you also want to turn on the video noise reduction filter. You just import the job file, turn on the noise reduction filter, and encode the clip. When you're done, you can discard the changes, or export the job under a new job file.

Tip: Helix Mobile Producer supplies a number of predefined job files that are stored in the **defaults** subdirectory under the Helix Mobile Producer main directory. You can use these files as templates for your own jobs, or create your own jobs from scratch using the graphical user interface.

► **To open and modify an existing job file:**

1. To open a job, choose **File>Import Job (Ctrl+I)** command and select the job file.

2. Make any necessary changes, which may include the following:
 - If you defined inputs and outputs in the job file, Helix Mobile Producer uses these same inputs and outputs. You can easily change this by deleting the input and output names and defining new ones, as described in “Setting Up Input and Output” on page 20.
 - Double-check the audience settings for the job to ensure that you are encoding the output properly. See “Setting Job Parameters” on page 47 for more information.
 - A job file can record specific clip information. You may need to add or change this information each time you encode with an existing job file.
3. When you finish making necessary changes, choose **File>Export Job (Ctrl+E)** to export the changes to the existing job file. If you want to create a new job file, enter a new file name.

Restoring the Active Job before Saving

To restore the active job, that is reset the settings stored on the disk, click **File>Revert** or press **Ctrl+R**.

BROADCAST

This chapter explains the different broadcast methods and describes the different parameters for each broadcast method.

RTP Broadcast Mode

RTP (Real-time Transport Protocol) is a standards-based packet format designed as the companion to the RTSP protocol. MPEG Players, for example, use RTP as its packet format. Helix Server fully supports RTP, and shifts to RTP automatically when streaming to an RTP-based client.

The following illustration shows the broadcast options that are available in RTP mode. These options are described in the table following the illustration.

RTP options



The screenshot shows a configuration window for RTP broadcast options. It includes a checked checkbox for 'Broadcast', a dropdown menu set to 'RTP', a text field for 'Server Address' containing '127.0.0.1', a 'Port Range' field with '5050' and '5054' separated by a hyphen, a 'TTL' field with '3', and an 'SDP file' field with 'live.sdp'. A 'Save As...' button is located to the right of the SDP file field.

<input checked="" type="checkbox"/> Broadcast:	RTP
Server Address:	127.0.0.1
Port Range:	5050 - 5054
TTL:	3
SDP file:	live.sdp

The following table lists and describes the broadcast options you can select when using RTP broadcast mode.

RTP Broadcast Options	
Option	Description
Server Address	A randomized IP address that is automatically created. You can modify this address by entering the IP address for the streaming server.
Port Range	The SDP file will pass this port to your server. The destination will include the port you specified and three ports above. For example, if you specify port 8558, the ports 8559, 8560, and 8561 will also be included. You must select an available port range that is not in use. For an audio-only broadcast, the actual port used is two ports higher than the selected port. For example, if you select port 8558, port 8560 is used. Note: Do not set your server to use these ports in the server administration interface.
TTL	TTL stands for Time to Live. This is a value from 0 through 255 that defines the scope within which multicast data packets should be sent over a network using IP as its protocol. Each router decrements the TTL by 1. When the value for a given data packet reaches a predefined lower limit, the router throws the packet away. By default, TTL is set to 1.
SDP file	SDP stands for Session Description Protocol. The SDP file, also known as the announce file, enables viewers to connect to a live encoding session. This file contains information about the audio and video streams in the live broadcast (for example, the type and format of the media) and information that players need to receive the media (for example, address and port data). By default, an announce file is generated automatically each time a live encoding session starts.

For more information about RTP, please read the following documentation:

- **RTP**— "RTP: A Transport Protocol for Real-Time Applications," RFC 1889, available at <http://www.ietf.org/rfc/rfc1889.txt>.
- **RTP**— "RTP Profile for Audio and Video Conferences with Minimal Control," RFC 1890, available at <http://www.ietf.org/rfc/rfc1890.txt>.

Tip on Firewall Configuration

Topology



Usually it is very troublesome to get the Helix Mobile Producer UDP signal to go through the firewall to the streaming server for live broadcast.

► A simple solution is to follow these steps:

1. Map a public IP address to the streaming server via the firewall, so that the streaming server has two IP addresses, one public IP address and one private IP address.
2. Configure the firewall to open up 4 UDP ports for every single live stream (for example, if you use the default port for the streaming server is 8558, open UDP ports 8558-8561)
3. Make sure the firewall does not block the ICMP signal. Helix Mobile Producer needs this to communicate with the streaming server.
4. Change the public IP address within the SDP file generated by Helix Mobile Producer to the private IP address. On one hand, RTSP/RTP player can only connect to the the streaming server via the private address, but on the other hand, Helix Mobile Producer needs to know the public IP address of the streaming server in order to send the live stream to it. So modifying the SDP file by hand is the easiest way to do this.

After all these steps, you should be able to connect to the streaming server using a handset within the protected private network.

Helix Broadcast Mode

You can use a computer running Helix Server as the destination server for your encoded output. On receiving your output media, Helix Server will immediately broadcast the stream to your audience.

For the RealAudio/RealVideo export type, four methods (3 push methods and one pull method) are available:

- **Push** broadcasting

In push broadcasting, Helix Mobile Producer initiates the broadcast stream and delivers the stream to Helix Server when the broadcast begins. There are several methods of push broadcasting:

- **account-based, push broadcast**—This is the simplest broadcasting method to set up, requiring no configuration on Helix Server. Helix Mobile Producer also receives statistical feedback from Helix Server. See “Account-based Push Broadcast” on page 38.
- **password-only, push broadcast**—This is similar to account-based broadcasting, but it is more efficient with bandwidth. It requires server set-up and does not provide server feedback, however. See “Password-Only Push Broadcast” on page 41.
- **multicast push**—This method is available if you have a multicast-enabled network and you want to deliver the broadcast stream to several servers. See “Push Multicast Broadcast” on page 42.

- **Pull**

In pull broadcasting, Helix Mobile Producer does not deliver the broadcast stream as soon as you start encoding. Rather, it waits for Helix Server to request the stream, which occurs when the first RealPlayer user requests the broadcast.

The following illustration shows the broadcast options that are available in Helix broadcast mode.

The Broadcast Methods

Helix Mobile Producer provides several different broadcast methods that you can use. Broadcasting methods break down into the two general areas of push and pull:

Broadcast Transport Protocols

When you use a non-multicast, push broadcast method, you specify whether to use TCP or UDP when delivering the broadcast stream to Helix Server. UDP is the preferred protocol because of the lower network overhead. But you may want to use TCP when delivering the broadcast over a lossy environment.

Note: The monitoring connection of an account-based broadcast always uses TCP whether the data transport stream is UDP or TCP.

• UDP

When Helix Mobile Producer uses the connectionless UDP protocol, it does not receive notice that broadcast packets have arrived or are missing. This generally reduces the network communications overhead and improves the quality of the broadcast. Helix Server notifies Helix Mobile Producer if it requires data to be resent. Thus, using UDP enhances the performance of your broadcast.

A firewall between Helix Mobile Producer and Helix Server may block UDP packets. The best solution is to configure the firewall to allow UDP packets through the ports that Helix Mobile Producer and Helix Server use the broadcast transmission. These ports, which vary depending on the broadcast method, are described in the sections about setting up each type

of broadcast. See “Advanced Settings - Account-Based Push” on page 39 and “Advanced Settings - Password-Only Push” on page 42.

- **TCP**

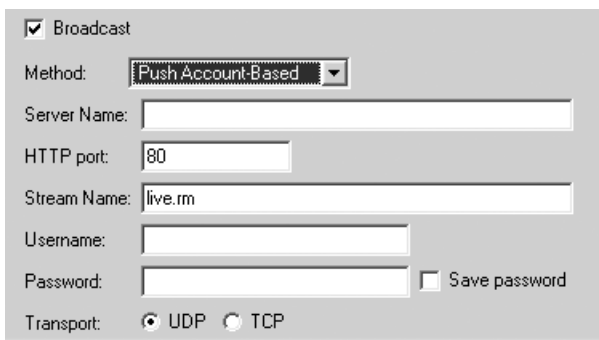
If you select the TCP protocol when broadcasting, Helix Mobile Producer and Helix Server establish a two-way connection. If a broadcast packet is lost in transmission, the network itself requests the packet to be resent. This makes TCP a highly reliable protocol for broadcast on a network prone to high packet loss. It is also more likely to pass through firewalls that you cannot configure to accept UDP communications.

Using TCP incurs a higher network and machine overhead, however, and can lead to inefficiencies. Helix Server does not need all stream packets to keep a broadcast flowing to RealPlayer. If a packet is lost, the network requests the packet again from Helix Mobile Producer. That packet may no longer be available, however. Or, if it is available, it may arrive at Helix Server too late to be useful. Hence, both Helix Mobile Producer and Helix Server must handle some network requests that do not benefit the broadcast.

Account-based Push Broadcast

The following illustration shows the account-based push options that are available in Helix broadcast mode. These options are described in the table following the illustration.

The Account-based Push Parameters



The screenshot shows a settings dialog box for Helix broadcast mode. At the top, there is a checked checkbox labeled "Broadcast". Below it, the "Method:" dropdown menu is set to "Push Account-Based". The "Server Name:" field is empty. The "HTTP port:" field contains the value "80". The "Stream Name:" field contains the value "live.rm". The "Username:" and "Password:" fields are empty. To the right of the "Password:" field is a checkbox labeled "Save password" which is unchecked. At the bottom, the "Transport:" section has two radio buttons: "UDP" (which is selected) and "TCP".

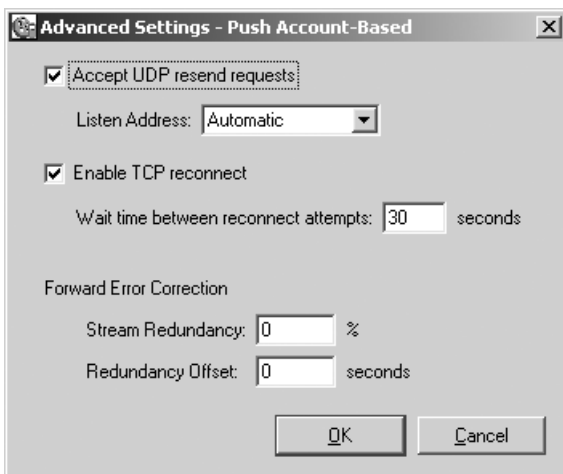
Account-based Push Parameters

Parameter	Default value	Description
Server Name	none	The Server name is the DNS or IP of the server to broadcast to.
HTTP Port	80	The Helix Server port that Helix Mobile Producer Live contacts when the broadcast begins. Following this initial contact, Helix Server and Helix Mobile Producer Live negotiate which server port to use for the actual encoded output. This needs to be the same as the HTTP port configured on the Helix Server.
Stream Name	live.rm	The broadcast stream name that will be exposed through any servers that connect.
Username	-	The username. This parameter is optional.
Password	-	The password. This parameter is optional.
Save Password	unchecked	If checked, Helix Mobile Producer stores the password in clear text in the job file, if the job is subsequently saved.
Transport	UDP	The transport method for the broadcast. UDP is the default, but TCP may be desirable for more complex networks (such as broadcasting over the internet to the server), or networks that do not allow UDP traffic.
Server Name	none	The Server name is the DNS or IP of the server to broadcast to.

Advanced Settings - Account-Based Push

The following illustration shows the account-based push advanced parameters. The following table describes the advanced settings dialog for Account based push.

Account-Based Push Advanced Parameters



Account-Based Push - Advanced Parameters

Parameter	Default value	Description
Accept UDP resend requests	checked	When UDP is selected as the transport method, this setting determines whether or not resend requests will be accepted from the server for lost packets.
Listen Address	Automatic	The listen address the encoder will listen for connection on. For more information, see “Listen Address Values” on page 41.
Enable TCP reconnects	checked	When enabled, and using the TCP transport method, if the connection is ever lost to the server, the encoder will attempt to reestablish it after the specified wait time (below).
Wait time between reconnect attempts	30 seconds	The wait time between attempts to reconnect to the server when using TCP transport. In most cases, 30 seconds is sufficient time for the server to do an automatic restart.
FEC Stream Redundancy	0	The percentage of redundant packets the encoder sends to the server. This effectively sends out duplicate packets, increasing the overall bandwidth used for the broadcast.
Redundancy Offset	0	If stream redundancy is set to 100%, this value determines the delay before sending the redundant stream packets.

Listen Address Values

The listen address sets the IP address that Helix Mobile Producer Live uses to listen for packet resend requests from Helix Server. For the listen address, you can use one of the following possible values:

- Automatic. This is the safest setting, and will work with most firewall configurations.
- System IP (the IP address of the machine).
- System IP 2 (the second IP of the machine is multi-homed).
- An IP address typed in by the user.

If your Helix Mobile Producer Live machine has multiple IP addresses, enter the IP address that Helix Mobile Producer Live should use for communications from Helix Server.

If you are broadcasting through a firewall performing network address translation (NAT), set the listen address to the IP address of the firewall or the value 0.0.0.0. The 0.0.0.0 value tells Helix Server to allow a Helix Mobile Producer Live connection from any IP address. The connection still requires the valid password, however.

Password-Only Push Broadcast

The following illustration shows the Push Password-Only options that are available in Helix broadcast mode. These options are described in the table following the illustration.

The Password-Only Push Parameters

The screenshot shows a configuration window for Helix broadcast mode. The 'Broadcast' checkbox is checked. The 'Method' dropdown menu is set to 'Push Password-Only'. The 'Server Name' field is empty. The 'Port Range' is set to '3020' and '3050'. The 'Stream Name' field contains 'live.rm'. The 'Password' field is empty, and the 'Save password' checkbox is unchecked. The 'Transport' section has 'UDP' selected with a radio button, and 'TCP' is unselected.

Password-Only Push Parameters

Parameter	Default value	Description
Server Name	none	The Server name is the DNS or IP of the server to broadcast to (for example, server.real.com). See the Broadcast Spec for detailed description.
Port Range	3020 - 3050	The port range for streaming.
Stream Name	live.rm	The broadcast stream name that will be exposed through any servers that connect.
Password	-	The server admin password.
Save Password	unchecked	If checked, stores the password in clear text in the job file, if the job is subsequently saved.
Transport	UDP	The transport method for the broadcast. UDP is the default, but TCP may be desirable for more complex networks (such as broadcasting over the internet to the server), or networks that do not allow UDP traffic.

Advanced Settings - Password-Only Push

Advanced parameters for Push Account-Based are the same as Push Account-Based. See “Account-based Push Broadcast” on page 38.

Push Multicast Broadcast

The following illustration shows the multicast push options that are available in Helix broadcast mode. These options are described in the table following the illustration.

The Push Multicast Parameters

The screenshot shows a configuration window for 'Push Multicast Parameters'. At the top, there is a checked checkbox for 'Broadcast'. Below it is a 'Method' dropdown menu currently showing 'Push Multicast'. The 'Multicast Address' field is empty. The 'Port Range' is set to '3020' and '3050'. The 'Stream Name' field contains 'live.rm'. The 'Password' field is empty, and there is an unchecked checkbox for 'Save password'. The 'TTL' field is set to '3'. At the bottom, there are radio buttons for 'Transport', with 'UDP' selected and 'TCP' unselected.

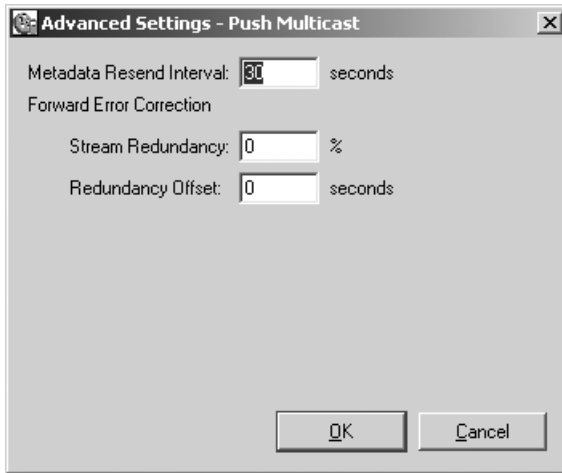
Push Multicast Broadcast Parameters

Parameter	Default value	Description
Multicast Address	none	The multicast IP address.
Port Range	3020-3050	The port range for streaming.
Stream Name	live.rm	The broadcast stream name that will be exposed through any servers that connect.
Password	-	The server admin password
Save Password	unchecked	If checked, stores the password in clear text in the job file, if the job is subsequently saved.
TTL	3	TTL stands for Time To Live. A value in the range 0 through 255 defines the scope within which multicast packets should be sent over a network using Internet Protocol (IP). Each router decrements the TTL by one. When the value reaches a predefined lower limit, the router throws the packet away.
Transport	UDP	The transport method for the broadcast. UDP is the default, but TCP may be desirable for more complex networks (such as broadcasting over the internet to the server), or networks that do not allow UDP traffic.

Advanced Settings - Push Multicast

The following illustration shows the Push Account-Based advanced parameters. The following table describes the advanced settings dialog for Account based push.

Push Multicast Advanced Parameters



Push Multicast Advanced Parameters

Parameter	Default value	Description
Metadata resend interval	30 seconds	The frequency at which to send stream metadata to servers connected to the Multicast. When a new server connects to the multicast, it must receive the header metadata before it can subsequently provide the stream to clients.
FEC Stream Redundancy	0	The percentage of redundant packets the encoder sends to the server. This effectively sends out duplicate packets, increasing the overall bandwidth used for the broadcast.
Redundancy Offset	0	If stream redundancy is set to 100%, this value determines the delay before sending the redundant stream packets.

Pull Broadcast

The Pull Broadcast Parameters

Pull broadcast Parameters

Parameter	Default value	Description
Listen Address	Automatic	The listen address the encoder will listen for connection on. For more information, refer to “Listen Address Values” on page 41.
Listen Port	3031	The port the encoder will listen on for connections.
Stream Name	live.rm	The broadcast stream name that will be exposed through any servers that connect.
Server Password	none	
Save Password	unchecked	If checked, stores the password in clear text in the job file, if the job is subsequently saved.
Server Timeout	30	The number of seconds Helix Mobile Producer will wait for a ping from a connected Helix Server before assuming no clients are connected to the stream and closing the connection.

Note: There are no Advanced Settings for Pull broadcast.

SETTING ENCODING PARAMETERS

This chapter provides detailed descriptions of audio and video encoding parameters and options.

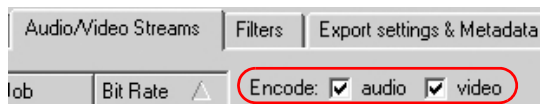
Setting Job Parameters

When encoding a media, you can define job-specific parameters: you can specify the track you want to encode (audio and/or), the bit rate control (variable or constant bit rate) and the number of encoding passes.

Audio/video encoding

If the input source contains both audio and video tracks, you can indicate the track you want to encode by selecting the corresponding option button, as shown in the following illustration.

Controls for Selecting Tracks



Bit rate control

Each audience is either constant bit rate (CBR) or variable bit rate (VBR). The CBR audiences are designed for streaming. When you use them, you can encode multiple audiences into a clip or broadcast. For example, a single clip can stream to 56 Kbps modems and faster broadband connections when you use CBR encoding.

The VBR audiences are designed for downloadable clips, but can also be used for streaming in some instances. When you encode a downloadable clip, you select a VBR template according to how highly you want to compress the input, or how well you want to preserve the input's quality. Unlike CBR clips that can have multiple audiences, VBR clips can include just one audience.

Constant bit rate (CBR)

Use this option if your content is going to be streamed over a limited-bit-rate channel such as a network. If you select this parameter, the output stream is delivered at the single, constant bit rate that you've specified.

Note: Selecting this setting might lower the quality of the video output. For MPEG-4 encoding, some video frames might be omitted to ensure that the output video stream conforms to the CBR that you selected. Higher bit rate values reduce coding artifacts, but they use more of the available bit rate.

When you create a clip, you can choose constant bit rate (CBR) or variable bit rate (VBR) encoding. In Helix Mobile Producer, each audience template is either CBR or VBR. CBR encoding is the more traditional method of encoding streaming video. It maintains a consistent bit rate for the stream, such as a constant 34 Kbps when streaming to 56 Kbps modems. You should generally use CBR video when streaming at bandwidths below 350 Kbps, and anytime you want to use a SureStream clip to encode multiple bandwidths into the same clip or broadcast. At high bandwidths, however, you can use CBR or VBR video.

Variable bit rate (VBR)

Use this option to set the variable bit rate, or average bit rate, for encoding the video. This bit rate is expressed in kilobits per second (Kbps). By default, the maximum bit rate is twice the average bit rate.

Variable bit rate encoding generally provides superior video quality to constant bit rate encoding. It gives more bandwidth to scenes that are hard to compress, making the most visible difference in videos that have fast moving, high-action scenes. VBR is not compatible with SureStream technology, however, so you can encode a VBR clip only for a single bandwidth. VBR encoding is suited for bandwidths of 350 Kbps or higher.

Unlike a CBR clip, a VBR clip does not maintain a constant streaming rate. Instead, it has a target average bit rate (or quality) and a maximum bit rate. For example, Helix Mobile Producer's 450 Kbps VBR download audience averages 450 Kilobits of data per second. However, the audience has a maximum of 900 Kbps, meaning that the stream can, if necessary, consume up to 900 Kilobits of data a second. These data spikes typically occur during high-action sequences, resulting in greater video clarity and higher frame rates than CBR encoding.

Tip: In Helix Mobile Producer, audience templates that use VBR encoding include ‘VBR’ in the audience name. If the template does not say ‘VBR,’ it is a CBR template.

For More Information: Two-pass encoding is an important component for creating a high-quality VBR file. For more information, refer to “Single pass/double pass encoding” on page 52.

VBR Clips for Download

VBR clips are better suited for downloading than are CBR clips. Because a viewer downloads the entire clip before playing it, the bandwidth spikes inherent in a VBR clip will not cause playback problems. As you encode clips for download, you can consider the VBR encoding speeds as a guide to quality and file size. For example, a 450 Kbps VBR download will generally have lower quality and a smaller file size than a 750 Kbps VBR download.

Tip: Quality depends on the source content. For a video that has relatively little action and small dimensions, for example, 750 Kbps VBR encoding may offer little improvement over 450 Kbps VBR encoding.

VBR Clips for Streaming and Broadcasting

Although VBR audience templates are geared for video downloads, you can also use VBR encoding for streaming clips and broadcasting live events. The primary reason to do this is that VBR offers better quality than CBR. For example, depending on the content, a 350 Kbps VBR stream may have roughly the same visual quality as a CBR stream encoded at 450 Kbps.

To benefit from VBR, the streaming network must be able to accommodate bandwidth spikes. Local area networks (LANs) and cable modem users are good candidates for VBR streaming. Because all viewers in these networks share a large pool of bandwidth, a VBR clip’s intermittent bandwidth spikes tend not to overload a single viewer’s connection bandwidth. Connections in which bandwidth is not shared, such as DSL, can prove problematic. A 450 Kbps VBR clip has a maximum bandwidth of 900 Kbps, for instance. If a DSL connection has a maximum throughput of 500 Kbps, the VBR clip may stall.

When you stream a prerecorded VBR clip on demand, each viewer generally starts playback at a different time. This means that the bandwidth spikes inherent in the clip are spread out over time for all viewers. When you

broadcast a VBR stream, however, bandwidth spikes occur for each viewer at the same time. When broadcasting, therefore, your network needs to handle higher cumulative spikes than when you stream prerecorded clips.

Note: Streaming a VBR clip requires that you use Helix Server version 9 or higher.

Tip: Multicasting, which is available on some intranets, helps to overcome the bandwidth spikes of standard unicasting by delivering one stream to all viewers, rather than a separate stream to each viewer. For more information, refer to the multicasting chapter of Helix Server Administration Guide.

VBR Encoding Settings

For each VBR audience template, three settings affect how the clip or broadcast is encoded: maximum bit rate, average bit rate, and quality. Each VBR clip uses these two settings to determine how it encodes its data. To create a VBR clip appropriate for your needs, it's important to understand how these settings interact.

Average Bit Rate

A VBR clip's average bit rate value reflects the average bandwidth of the clip measured as the total number of Kilobits consumed divided by the timeline in seconds. The actual bandwidth at any point during clip playback may be lower or higher than this value. Through most of the video, however, the second-by-second bandwidth use will be close to this value.

Maximum Bit Rate

A VBR clip's maximum bit rate caps the bandwidth that the clip can consume. It is typically set by default to twice the average bit rate. A 450 Kbps VBR clip has a maximum bandwidth of 900 Kbps, for example. So during a high-action scene, for example, a video encoded with this audience may have a bandwidth spike of up to 900 Kbps. These spikes are only occasional, however, and the overall clip playback rate stays close to the average bit rate.

Although you can set the maximum bit rate value higher or lower, a higher setting is unlikely to result in significantly greater quality. You can lower the setting if you want to reduce the bandwidth spikes on your network. Keep in mind, however, that lowering this maximum decreases the benefit of VBR encoding. The closer the maximum bit rate value approaches the average bit rate value, the more the clip behaves like a CBR clip.

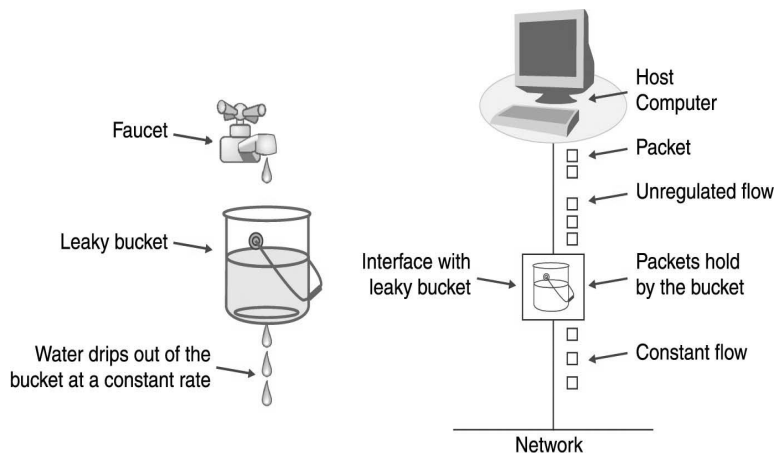
Tip: Keep the maximum bit rate value between 50 percent to 100 percent greater than the average bit rate setting. For a 450 Kbps template, for example, the maximum rate should be 675 to 900 Kbps.

The leaky bucket algorithm

Both CBR and VBR are consistent with the “leaky bucket” concept. Leaky bucket is a flow-control mechanism designed to reduce the effect of the inevitable variability in input data streams as they are “injected” into communication networks. The leaky bucket algorithm performs the following functions:

- It controls what the encoder can inject into the network.
- It prevents the data “bucket” from overflowing by automatically regulating the bit rates in the event that the encoder generates too much data or very complex data.
- It prevents “burstiness” in the input stream, ensuring a smooth, even data flow.

The Leaky Bucket Algorithm



The video buffer is the normative MPEG-4 video buffering verifier (VBV). The VBV is the instantaneous bit-rate smoother, or transmission buffer, that ensures that the instantaneous bit rate never exceeds the target bit rate behind the video buffer. You can also use it as a reception buffer.

Single pass/double pass encoding

Although it is specified at the Job level of the encoder, 2-pass encoding is specific to the video codecs.

For offline encoding only, you can select either the single pass or double pass encoding, by selecting the Enable double pass radio button. Double-pass encoding is for offline encoding with Helix Mobile Producer. It uses information from the first pass to reallocate the bits during the second pass. This enables you to encode video input at a very high level of quality. Keep in mind though, that double-pass encoding takes twice as long as single-pass encoding.

Note: This setting is ignored for live encoding.

Two-Pass Encoding

With two-pass encoding, which is used only when encoding from a digitized source file, Helix Mobile Producer runs through the entire source video once to gather information about how best to encode the streaming clip. It then makes a second pass to encode the streams. Two-pass encoding can substantially increase clip quality, but it requires more encoding time. The first pass takes about as long as it would to encode the source file for one target audience.

Although two-pass encoding helps when you use constant bit rate encoding, it provides greater benefit for variable bit rate (VBR) encoding, described in “Variable bit rate (VBR)” on page 48. With two-pass encoding, Helix Mobile Producer can analyze the entire video file to determine how best to vary the playback bit rate through the length of the clip. Without two-pass encoding, Helix Mobile Producer sequentially analyzes small sections of the source file during encoding, creating a string of VBR sections within the clip.

Tip: Use two-pass encoding whenever you encode from a digitized file. Turn it off only if you must decrease the encoding time.

Setting the Bit Rate

Use this option to set the standard bit rate for encoding the video stream, in kilobits per second (Kbps). Note that to set a constant bit rate or a variable (average) bit rate for the clip, you use the **Bit rate control** option, as described in “Bit rate control” on page 47.

Setting Audio Parameters

This section explains how to use the various encoding options to set the basic parameters for encoding audio input.

The default audio encoding parameters in Helix Mobile Producer have been set carefully to provide the best overall quality of audio output for the most common streaming audio scenarios. We recommend changing these parameters *only* if you have advanced knowledge of audio encoding.

The Audio Parameters (AAC encoder)

The screenshot shows a dialog box for the AAC encoder with the following settings:

- Audio Codec: AAC
- Bit rate: 8000 bps
- Use joint stereo
- Custom Sampling Rate
- Channel mode: mono
- Sampling rate: 8.000 kHz

Encoder type

The following table lists and describes the six audio encoding formats supported by Helix Mobile Producer.

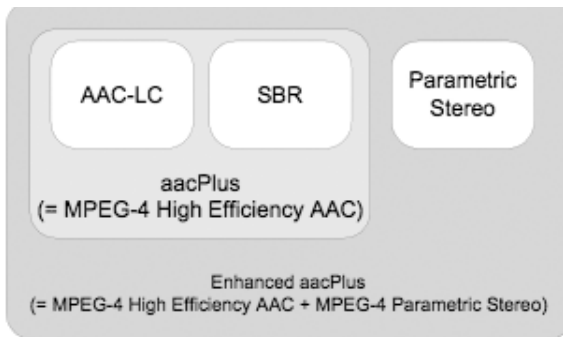
Audio Encoding Formats

Format	Description
AAC	Advanced Audio Coding Low complexity. AAC-LC allows the following settings: <ul style="list-style-type: none"> bit rate - a value from 8 000 to 320 000 bps, default value is 96 000 bps joint stereo option the audio sampling rate in kHz - allowed range is dependant of the defined audio bit rate the number of channels (1 or 2) - allowed range is dependant of the defined audio bit rate
AAC Plus	AAC Plus allows the same settings as AAC-LC, except joint stereo option. Note: AAC Plus is only available for use with the 3GPP v6 export types.

Audio Encoding Formats (continued)

Format	Description
Enhanced AAC Plus	Enhanced AAC Plus allows the same settings as AAC-LC, except joint stereo option. Note: AAC Plus is only available for use with the 3GPP v6 export types.
AMR narrowband	Adaptive multirate speech codec. The options for AMR-NB are limited to the bit rate selection, and the Use Silence Detection option.
AMR wideband	Adaptive multirate speech codec. The options for AMR-WB are limited to the bit rate selection, and the Use Silence Detection option.
QCELP	QualComm Code Excited Linear Predictive Coding speech codec. The only option is one of these two bit rates: <ul style="list-style-type: none"> • 14 000 bps (full-rate) • 6 800 bps (half-rate)
RealAudio	RealNetworks audio file format. The only option is the codec flavor.
MP3	MPEG-3 audio file format. The only option is the bit rate (in bps). Note: Stereo or Mono is specified automatically: bit rates below 64 000 are mono, 80 000 and above are stereo. If the input is mono, then the output is also mono.

Make-up of the Enhanced AAC Plus codec



Bit rate

Use this option to specify the bit-rate value for an audio stream in bits per second (bps). Different bit-rate ranges are available for different audio encoders.

Note: This option is not available for the RealAudio encoder.

Use joint stereo (AAC audio encoder only)

Joint stereo coding removes redundancies between the left and right audio channels of stereo sources. Select this option to encode channel redundancies and differences separately. Using this setting can reduce the bit rate by as much as half, but it also might introduce some artifacts. Clear the **Use joint stereo** check box if you want the redundant portions to be encoded twice.

Custom Sampling Rate (AAC, AAC Plus and Enhanced AAC Plus encoders)

When you check this option, you can specify the channel mode (mono or stereo), and the sampling rate.

Silence Detection - DTX compression (AMR encoder only)

This option enables the encoder to detect silence periods in speech. Such periods are then coded with a "silence targeted algorithm" that improves coding efficiency and decreases the bitrate.

Flavor (RealAudio encoder only)

Select an encoding preset depending on the input audio stream (voice, music,...).

Setting Video Parameters

This section explains how to use the various encoding options to set the parameters for encoding video input. The following illustration shows the video parameters in the Audio/Video Streams tab.

The Video parameters (H.264 video encoder)

Video Codec: H.264

Bit rate: 37.8 kbps (Target - Audio)

Video codec level: Automatic

Max keyframe distance: 5000 milliseconds

Change frame rate: 10 fps

Encoding complexity: High

Video mode: Normal Motion Video

Encoder type

The following table lists and describes the three video encoding formats that Helix Mobile Producer supports.

Video Encoding Formats

Format	Description
MPEG-4	ISO/IEC video codec
H.264	MPEG-4 AVC. International Telecommunications Union video codec
H.263	International Telecommunications Union video codec
RealVideo	RealNetworks video codec

RealVideo Codecs

RealVideo 10 is the default RealVideo codec used with Helix Mobile Producer, but you can also encode with older RealVideo codecs. RealNetworks recommends using RealVideo 10 unless you need faster encoding performance during broadcasts, or you need to stream video to RealPlayer 8.

RealVideo 10 Codec

The RealVideo 10 codec creates the highest quality compressed video possible. It offers improved visual quality over RealVideo 9 and RealVideo 8, especially with fast-action scenes and on-screen text. Because RealVideo 10 performs more complex analysis of video data than earlier codecs, encoding may take more than twice the time required with RealVideo 9. To help shorten the encoding time and improve the quality of live broadcasts, Helix Mobile Producer provides the following features:

- RealVideo 10 is optimized for multiprocessor machines. Helix Mobile Producer uses a second processor when the video height is greater than 180 pixels. When you encode a SureStream CBR clip for multiple audiences, Helix Mobile Producer uses up to two processors for each audience. For example, encoding a RealVideo clip for a 56 Kbps modem audience and a 256 Kbps broadband audience can utilize up to four processors.
- When encoding pre-recorded clips, you can lower the codec complexity level. This reduces the clip's visual quality, but decreases the encoding time.
- For live broadcasts, Helix Mobile Producer uses automatic load management features to compensate for the increased encoding demands of RealVideo 10. This helps to ensure that Helix Mobile Producer does not fall behind the video input rate. See "Broadcast Load Management".

RealVideo 10 is compatible with RealOne Player and later. Users of older RealPlayers are prompted to update to RealPlayer 10 when they attempt to play RealVideo 10 content. Playback of RealVideo 10 content consumes the same amount of system resources on the viewer's computer as playback of RealVideo 9. Viewers, therefore, will not notice any performance slowdown when playing a RealVideo 10 clip compared to a RealVideo 9 clip.

RealVideo 9 Codec

RealVideo 9 improves on RealVideo 8 with higher compression and improved visual quality. RealOne Player and later can play RealVideo 9 clips. Users who have older versions of RealPlayer are prompted to autoupdate to RealPlayer 10 when the viewer attempts to play a RealVideo 9 clip.

RealVideo 8 Codec

The RealVideo 8 codec is backwards-compatible to RealPlayer 8. The video quality is not as high as with RealVideo 9 and 10, but encoding is faster. Additionally, RealVideo 8 requires fewer resources on the RealPlayer machine to decompress. This makes it suitable for the slower processors of mobile, handheld devices.

Bit Rate

Bit rate is a value in bps that must be at least 8 kbps. The value is calculated by taking the Total bit rate of the audience, and subtracting the audio bit rate. The remainder is the video bit rate for the audience.

Max key frame distance

This value defines the keyframe interval in seconds with accuracy to 1 decimal place and a range of 0 to 30 seconds.

It is called Maximum time between key frames because it only defines an upper-limit. Key frames may be inserted more frequently due to scene changes or for other reasons. This value guarantees that there will be at least 1 key frame every x seconds, where x is the user-specified value.

If 0 is specified, every frame encoded will be a keyframe, which will drastically reduce the frame rate

The settings in this section are common to all video codecs.

Video Mode and Encoding Complexity are 2 parameters that come from RealVideo encoders that will be applied to all the video codecs. Note that for RealMedia encoding, the definition is more precise.

Change Frame Rate

Frame rate is specified on a per-audience basis for the video stream (this allows different video streams in a multi-rate file to have different frame rates). The value is specified in frames per second, with an accuracy of 1 decimal place. (example: 30 fps, or 7.5 fps)

The acceptable range is 0.5 up to 30 frames per second (0.5 would represent one frame ever two seconds).

Maximum Frame Rate

In multi-rate files, each video stream can have it's own frame rate setting. There is no guarantee that the target frame rate will be achieved. Limiting factors will include the frame rate of the source, the bit rate of the video stream, the dimensions of the video stream, the codec used, and the load on the machine.

RealVideo codecs should be feed all video frames, along with the target frame rate to allow the codec to decide how to achieve the best results. For example, if the target frame rate is 15 fps, but the bit rate and frame size make achieving this consistently impossible, the RealVideo codecs will drop down to a lower constant frame rate for the best end user experience.

Select this option to change the frame rate (the number of frames per second, or fps) in the original video source. Note that reducing the frame rate

decreases the bandwidth used by the encoder and might affect the minimum bit rate.

Be aware that reducing the frame rate diminishes the quality of the motion in the video stream and that it's not necessary to do this for most video sources.

Encoding Complexity

Encoding complexity determines the amount of processing the codec will do; more processing results in higher quality, but is also slower. When doing a high bit rate live encode, it may be desirable to set a lower complexity level that the system has the resources to maintain throughout the encode.

Encoding complexity

Encoding complexity level	Description
High	Default level and best quality. Also slowest. (2x or more the processing resources of Medium)
Medium	Balance between quality and complexity. (2x or more the processing resources of Low)
Low	Fastest encoding, but lowest quality.

Video Mode

Video mode helps the codec determine how to spend bits. The choices are defined in the table below:

Video mode

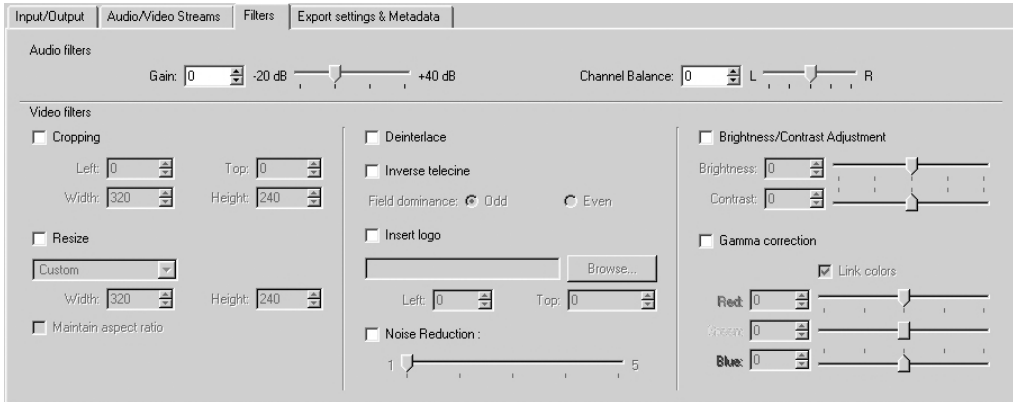
Video Mode Setting	Description
Normal Motion Video	Default: Suitable setting for almost all video content. Provides a balance between framerate and image sharpness.
Sharpest Image	Provides the sharpest quality image at the expense of smoothness of motion and framerate
Smoothest Motion	Provides the smoothest motion/highest framerates at the expense of image clarity
Slide Show	The absolute sharpest quality, displays video in a slide show mode with very low framerates.

This encoder will always perform CBR encoding even if VBR is chosen at the job level. Advanced settings for the H.264 codec are documented in the following subsections.

Setting Audio and Video Filters

This section describes the various audio and video filters included in Helix Mobile Producer. You can use these filters (shown in the following illustration) to adjust and customize your streaming video content in a number of different ways before you begin encoding it.

The Filters Tab



Audio Filters

Channel Balance

Use the slider to adjust the left/right balance for stereo audio inputs (from -100 to +100). Negative settings skew the balance towards the left channel, positive values skew it towards the right channel. A setting of 0 has no effect.

Note: This filter is only effective when both the input source audio and the output codec are stereo.

Gain

Use the slider to set the volume level of the source audio before encoding. The allowed range for the gain parameter is in decibels between -20 dB to 40 dB. When set to 0, this filter does nothing.

Video Filters

Video prefilters are applied in a specific order in the encoding chain from first to last as follows:

- Inverse Telecine
- Cropping
- Deinterlace
- Brightness/Contrast
- Gamma Correction
- Resize
- Logo Insertion
- Change Frame Rate

Inverse Telecine

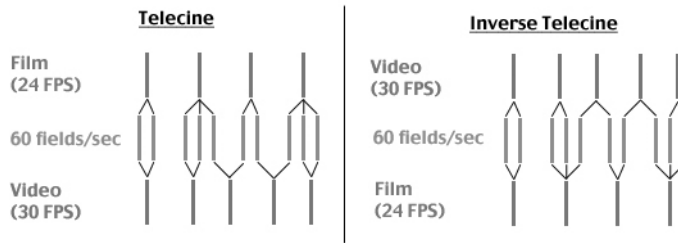
Telecine is the conversion of film content (24 fps) into NTSC broadcast format (29.97 fps). This video prefilter converts NTSC (29.97 fps) telecined content back to the original film format (24 fps). Artifacts created by the "telecine" process are removed. A mandatory parameter is needed to decide the field priority of the input content (odd/even).

The Inverse Telecine portion is designed to take 29.97 fps input that originated from 24 fps film and convert it back to 24 fps.

Note: This filter will only work with 29.97 fps source material. If the original content was not telecined, the quality of the output will be poor.

The **Field Dominance** parameter specifies the field order (either odd or even). If the wrong field order is specified, the resulting video will be noticeably distorted.

Inverse Telecine Diagram



Cropping

This video prefilter crops the video to a specified region.

The video cropping prefilter exposes values for Left, Top, Width, and Height for the cropping region. The basic restrictions on possible values are that the cropped image must be no less than 32x32 pixels (or in the case of H.263, 128x96) and no larger than the original image.

Note: RealVideo codecs require their input video size to be divisible by 4.

Deinterlace

This filter will remove interlace artifacts (jaggies) that are visible when converting 29.97 fps - 60 fields/second interlaced video (this is what is broadcast to your television) to 29.97 fps progressive video (what you see on a computer screen). The jaggies that result from this conversion are a result of the even/odd fields in the source video, and is most noticeable on moving images.

Use this filter for video input that is interlaced. Keep in mind, however, that if the original video source was *not* interlaced and you use the deinterlace filter anyway, the quality of the output will be very poor.

Note: A video is interlaced if you can see a "combing" effect in the preview panel. Essentially, an interlaced frame consists of two images captured at different times and combined in one frame in such a way that the two instants merge into a single image.

Deinterlace filter applied on an interlaced video (detail)Jaggedness in an
Interlaced VideoJaggedness Removed
with the De-interlace Filter**Color Adjustment**

This video prefilter adjusts the brightness and/or contrast of the input video.

Both brightness and contrast have a value range of between -100 to +100 in whole digit increments. A value of 0 has no effect. For the brightness filter, a value of -100 equals full black, and +100 equals full white.

Brightness, contrast. You can adjust the brightness and contrast of a video stream by dragging the sliders back and forth or by entering numeric values in the corresponding text boxes.

Gamma correction. You can adjust the gamma values of the input video by moving the sliders for the three colors components (R, G, B) individually (the default method). Alternatively, you can select the **Link colors** option to link the three sliders so that they move together, to increase or decrease full color saturation.

Each component has a value range of between -100 to +100. A value of 0 has no effect.

Resize

This video prefilter resizes the video to a specified width and height value. The possible range of values for these parameters varies by the video codec, as detailed in the table below.

Resizing Values

Codec	Min. Size	Max Size	Min. increment
H.263 (Only CIF, QCIF, and SQCIF are supported.)	128x96 (SQCIF)	384x288 (CIF)	-
H.264	32x32	640x288	4

Resizing Values

Codec	Min. Size	Max Size	Min. increment
MPEG4	32x32	640x288	4
RealVideo	32x32	2048x2048	4

You can resize your video input either by selecting a predefined size (for example, 50% or CIF) or by specifying the height and width of the video output (in pixels).

Insert Logo

Use this option to add a still image logo overlay onto the video input frames before encoding. A common use case is to insert a station or content provider identifier.

The logo insertion filter has three parameters.

- The first is a filename and path to an image file. JPEG, BMP, PNG, and GIF are the supported image formats; animated GIFs are not supported.
- The distance of the left edge of the logo from the left edge of the video.
- The distance of the top edge of the logo from the top edge of the video.

Change Frame Rate

Sets the maximum output frame rate of a video stream. In multi-rate files, each video stream can have its own frame rate setting. There is no guarantee that the target frame rate will be achieved. Limiting factors will include the frame rate of the source, the bit rate of the video stream, the dimensions of the video stream, the codec used, and the load on the machine.

RealVideo codecs should be fed all video frames, along with the target frame rate to allow the codec to decide how to achieve the best results. For example, if the target frame rate is 15 fps, but the bit rate and frame size make achieving this consistently impossible, the RealVideo codecs will drop down to a lower constant frame rate for the best end user experience.

Select this option to change the frame rate (the number of frames per second, or fps) in the original video source and specify the ratio for dividing frames. Note that reducing the frame rate decreases the bandwidth used by the encoder and might affect the minimum bit rate.

When changing the frame rate, you can specify a frame-rate divide ratio (either 2:1, 3:1, 4:1, or 5:1) or a target frame rate (from 1 through 30 fps). If you do

change the rate, we recommend using 2:1, which is the default divide ratio. For example, if your original video source has a frame rate of 30 fps and you use the 2:1 divide ratio, the frame rate in the output will be 15 fps.

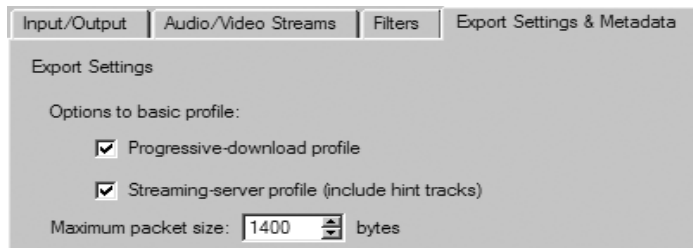
Be aware that reducing the frame rate diminishes the quality of the motion in the video stream and that it's not necessary to do this for most video sources.

Setting Export Parameters

The following subsections describe the available export options and provide detailed instructions on how to use them, as well as providing ancillary information that will help you prepare your video and audio clips for exporting.

These export options are shown in the following illustration.

The Export Parameters (3GPP v5 export type)



Note: If you modify the export type, by default both options will be selected.

If no option is selected, the media file will be downloaded and played locally but not streamed from a server. Note that you will have to wait for the complete file download before playing it.

Streaming-server profile (MPEG-4 and H.263 encoders only)

If you plan to stream video or audio content from a streaming server, you can use the **Streaming-server profile** option to include "hint tracks" that tell the server how to send the streams and what to set as the maximum bit rate for the presentation.

Be aware that the addition of data such as hint tracks and headers to audio or video streams during the encoding process results in the encoded files being somewhat larger than the sum of the encoded media bit rates multiplied by

the duration of the file ((audio bit rate + video bit rate) x duration). This does not affect the bit rate or quality of the media, because this additional data remains on the server and does not get streamed to the client.

Note: The hint track is required by the streaming server to optimize the streaming experience. A 3GPP media file must have a hint track in order to be properly streamed from most streaming servers. The hint track is not required if the media file is to be downloaded and played locally and not streamed from a server. To avoid having Helix Mobile Producer add a hint track to such a file, select the **Basic-profile** check box on the **Export** tab before encoding the audio or video clip.

Progressive-download profile (MPEG-4 and H.263 encoders only)

By default, MPEG-4 files are generated with the metadata following the raw media data, which results in efficient writing of the MPEG-4 files. However, this means that a player must completely download an MPEG-4 file before playing it, as the player cannot begin playback until it has received the metadata.

If you have selected the **Progressive-download profile** option, the MPEG-4 file is reordered at the end of the encoding process so that the metadata appears at the *beginning* of the file. This enables a player to download and play the content at the same time, rather than having to wait for the entire file to be downloaded before playing it.

Maximum Packet Size (3GPP, MPEG-4 and RealMedia export types only)

The maximum packet size should be less than the maximum transmission unit (MTU) of the network over which the content will be streamed. Packet header "overhead" should also be taken into account in this calculation. By default, the maximum packet size is configured to be appropriate for general Internet usage, for which the MTU of 1,500 bytes takes into account UDP and IP packet headers. For other types of target networks, such as ATM or wireless networks, the maximum packet size should be adjusted to reflect the corresponding larger or smaller MTU.

A maximum packet size that is too small will add unnecessarily high overhead, as the packet headers will take a larger proportion of the bit rate relative to the packet payloads.

A maximum packet size that is too large will reduce the error resiliency of the stream because the packets will be fragmented on the network, meaning that the loss of one packet fragment will cause the entire packet to be discarded.

Calculating Maximum Packet Size

Use the following procedure to determine the maximum packet size that you can use for a presentation that will be streamed over a network.

► **To define the output packet size:**

1. Identify the MTU for your network.
2. Identify the packet structure used on your network.
3. Subtract the header size from the MTU to get the number of bytes available for the RTP packet, as shown in the following text and tables.

RTP streaming over UDP over Ethernet

The MTU is 1,500, and the packet structure is as follows:

Packet Structure

20 byte IP header	8 byte UDP header	12 byte RTP header	N byte RTP payload
----------------------	----------------------	-----------------------	-----------------------

Use the following equation to determine the value to set for maximum packet size over UDP packet size:

$$1,500 - 20 - 8 = 1,472$$

RTP streaming over RTSP interleave over Ethernet

The MTU is 1,500, and the packet structure is as follows:

Packet Structure

20 byte IP header	28 byte TCP header	4 byte - RTSP interleave header	12 byte RTP header	N byte RTP payload
----------------------	-----------------------	---------------------------------------	-----------------------	-----------------------

Use the following equation to determine the value to set for maximum packet size over RTSP interleave:

$$1,500 - 20 - 28 - 4 = 1,448$$

The following table lists the types of packet headers and the typical size for each one.

Typical Packet Header Sizes

Header	Size (in bytes)
IP	20
PPPoE	8
RTP	12
RTSP interleave	4
TCP	28
UDP	8

The following table lists the applicable RFCs and gives a description and the typical MTU for each one.

Typical MTUs

RFC	Description	MTU
894	Minimally required	68
1051	ARCNet	508
1356	X.25, ISDN	576
1055	Serial line IP (SLIP)	1,066
1042, 2516	IEEE 802.3 / 802.2, PPPoE	1,492
894, 895	Ethernet	1,500
1390	FDDI	4,352
1042	4-Mb token ring	4,464
1042	802.4 token bus	8,166
None	16-Mb token ring	17,914
1374	HIPPI	65,535

Loss Protection (RealVideo encoder only)

Adds error correction information to the video stream to assist the player in reconstructing lost packets.

Latency mode (RealMedia export type only)

Each RealVideo clip has a maximum startup latency that determines how long video requires to display after RealPlayer begins to receive the stream.

There are three possible values:

- Normal (longest delay from real-time, but best viewing experience),
- Moderate (shorter initial delay that may increase and produce some re-buffering,
- Low (shortest delay, but intermittent sections of media may be lost).

If you select Normal, you can specify the maximum latency. This specifies the video buffer size, from 1000 ms to 25000 ms. The default is 4000 ms.

Increasing the latency may be particularly useful for videos that stream at low bit rates and start out with high action sequences. The longer latency creates a larger data buffer for the starting sequence, and generally improves the video's appearance. Bear in mind, however, that a long latency time may cause restless viewers to stop the presentation before it begins playback.

Note: The startup latency does not affect how quickly a downloaded clip begins to play. Increasing the latency value, however, can improve the visual quality in downloaded clips that begin with fast-action sequences.

Setting Metadata

Helix Mobile Producer can encode clip information directly into a clip or broadcast. This is highly recommended for all jobs because the clip information tells the viewer about the stream and can help search engines categorize clip.

You can specify the following information.

Title

In this field, enter the title of the clip or broadcast. Because this title appears in the RealPlayer interface, it is best to use a short title.

Author

This field holds the name of the person or organization that created the clip.

Copyright

Here, enter the copyright string, such as (c) 2004 ABC Corporation.

Keywords

The keywords field holds words that certain audio and video search engines can read to categorize the clip. Add a few words that will help your audience search for your clip. Separate each term with spaces. Unless you are adding a proper name, use lowercase for each term. Avoid overly generic terms such as video or music.

Description

This field holds a description of the clip that appears when the viewer displays extended clip information. This allows you to describe the clip in detail without creating a long title.

Rating (3GPPv6 and RealMedia export types only)

Selecting a rating in the pull-down list is highly recommended for any content not intended for all age groups. The rating depends on the selected rating system.

Note: 3GPP v6 accepts any two 4-char codes, plus a description. Some predefined values are included, but other values can also be entered.

MPAA (Motion Picture Association of America).

The current MPAA movie ratings consist of:

- G – General Audiences: All ages admitted.
- PG – Parental Guidance Suggested: Some material may not be suitable for children.
- PG-13 – Parents Strongly cautioned: Some material may be inappropriate for children under 13.
- R – Restricted: Under 17 requires accompanying parent or adult guardian.
- NC-17 – No one 17 and under admitted.

BBFC (British Board of Film Classification)

The current MPAA movie ratings consist of:

- U – Universal: Suitable for all

- PG – Parental Guidance: All ages admitted, but parents are advised that certain scenes may be unsuitable for small children
- 12 – Suitable for those aged 12 and over (video only)
- 15 – Suitable for those aged 15 and over
- 18 – Suitable for those aged 18 and over
- R18 – Restricted 18: Suitable for those aged 18 and over and only available at licensed cinemas and shops

OFLC (Australian Office of Film and Literature Classification)

The OFLC uses the following rating system:

- G – Suitable for all viewers.
- PG – Parental Guidance recommended for children under 15 years of age.
- M – Mature, recommended for audiences 15 years and over.
- MA – Mature Accompanied. This category is legally restricted in that children under 15 cannot see "MA" films or rent them on video unless accompanied by a parent or adult guardian.
- R – Restricted. This category is legally restricted to adults. No one under 18 may view these movies in a cinema or rent them on videocassette.
- X – Restricted. This rating applies to sexually explicit material which is restricted to viewers 18 years of age and over.

RealMedia

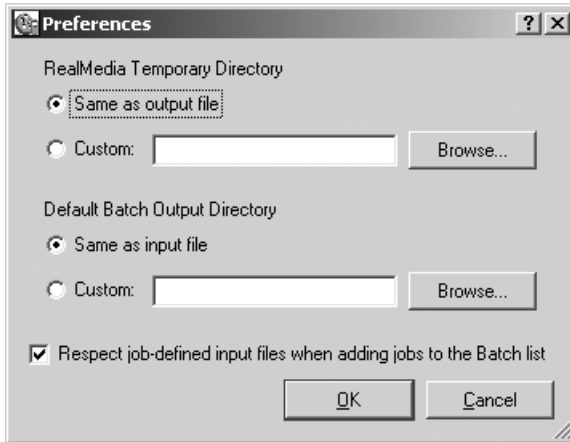
For RealMedia, the rating can be one the following:

- Not rated
- All Ages
- Older Children
- Younger Teens
- Older Teens (15 years and over)
- Adult Supervision Recommended
- Adults Only

Setting Preferences

This section explains how to use the options in the **Preferences** dialog box (shown in the following illustration). To access this dialog box, click **Edit>Preferences** in the menu bar at the top of the Helix Mobile Producer workspace.

The Preferences Dialog Box



Preference Options

The following paragraphs discuss the options that enable you to set the default Realmedia temporary directory, and the default batch output directory.

Temporary Directory

By the default, the temporary directory is the same as the output file directory, but you can specify a custom directory.

Default Batch Output Directory

By the default, the batch output directory is the same as the input file directory, but you can specify a custom directory.

Respect Job-Defined Input Files...

If you uncheck this option, the input file defined in the job will be ignored.

USING THE COMMAND-LINE ENCODER

This chapter shows you how to run Helix Mobile Producer from the command line in Windows. Helix Mobile Producer gives you the ability to encode streaming audio or video presentations as files or encodes live capture input for live broadcasts and file output.

Basic Encoding

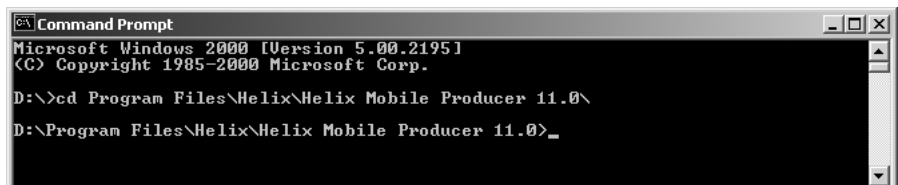
Helix Mobile Producer provide a simple command-line interface you can use to encode your audio or video input.

Getting Started

You access the command line from the Windows command prompt.

► To use the command-line interface:

1. Open the Command Prompt window.
2. Set the current directory to match the location of the **hmprod.exe** program by typing `cd` followed by the path to the program.



```
Command Prompt
Microsoft Windows [Version 5.00.2195]
(C) Copyright 1985-2000 Microsoft Corp.

D:\>cd Program Files\Helix\Helix Mobile Producer 11.0\
D:\Program Files\Helix\Helix Mobile Producer 11.0>_
```

3. Type **hmprod** followed by command-line arguments to start the command-line encoder.

Note: Starting the command-line encoder without including any arguments will display the syntax and command-line flags for the application you have specified (**hmprod.exe**, in this case).

► **To set the path to Helix Mobile Producer:**

To avoid navigating to the directory that contains Helix Mobile Producer, include the path to the executable (.exe) file in your Path variable. For example, enter the following command at the command prompt:

```
set PATH=%PATH%;D:\Program Files\Helix\Helix Mobile Producer;
```

Basic Encoding Commands

You can use the command-line interface to encode streaming media in one of two ways: offline (file input to file output) or live (live capture input to file output or broadcast output).

To run a job file from the command line, use the following syntax:

```
hmprod -j <job_file> [input] [destination] [options]
```

A job file is an XML file that specifies all settings related to an encoding session. Job files can be created via the GUI application (see Chapter 6), or edited with any text or XML editor.

You can specify additional arguments and flags on the command line. These are described in the next section, “Command-Line Reference” on page 74.

Batch encoding

To perform batch encoding of many input files, use a directory name as the input and a directory name as the output. For example:

```
hmprod -j myjob.xml -i c:\inputs\ -o c:\outputs\ ...
```

This creates one output file for each input file using the same settings for every file. Each valid source file in the input directory will be encoded sequentially.

Command-Line Reference

The command-line interface has several flags and arguments that you should be familiar with before doing any encoding from the command line. To simplify command-line use and to make it easier to reuse encoding settings, you can use job files to define all of the encoding settings.

Helix Mobile Producer Command-Line Flags

The command-line flags are listed and described in the following paragraphs.

Job file parameters

The job file options are listed in the following table:

Job File Option	
Syntax	<code>-job <jobfile></code>
Description:	Denotes the XML job file that specifies all of the encoding parameters. Note that the job filename can be a relative or absolute path on a network or local drive. If it is relative, the location must be relative to the current working directory (the directory you executed producer from).
Limitation:	This flag cannot be used with any option other than Logging Category (-lc)
Example:	<code>-j MyJob.hmpj</code>

Inputs

The possible options for audio/video inputs are the following:

- `-i` Input file or directory
- `-d` Duration
- `-st` Start time
- `-et` End time

The file or directory input option is described in the following table:

Input File Option	
Syntax	<code>-i</code>
Description:	A filename pointing to a file or directory of files to be encoded. The input file may be one of the following: - A single filename pointing to a supported input media type - An input directory name (batch) Note that the job filename can be a relative or absolute path on a network or local drive. If it is relative, the location must be relative to the current working directory (the directory you executed producer from).

Input File Option

Batch Encoding	<p>If a directory is specified, a batch encode is performed where each input file is encoded sequentially one after another.</p> <p>All files are used as input and are output to the directory as defined in the Output File option (-o).</p> <p>If the output directory is not specified, output files are placed in the same directory as the input file.</p> <p>If the input filename does not exist or, for directory, does not resolve to at least one valid input file, an error is generated and encoding stopped.</p>
Supported input file formats	<p>The following list describes the input file types supported and the platforms they are supported on:</p> <ul style="list-style-type: none"> - Windows - Any compressed and uncompressed file types supported by DirectShow, - Any compressed and uncompressed file types supported by QuickTime. <p>The later two items under Windows include compressed AVI, compressed QT, MPEG1, MPEG2, MP3 and many other compressed and uncompressed file formats.</p> <p>Note that for MPEG2 you need to have an MPEG2 DirectShow filter which is installed by many software DVD players.</p>
Limitation:	<p>This tag cannot be used in conjunction with a job that specifies server destinations.</p>
Examples:	<pre>-i myfile.avi -i c:\media\movie1.avi -i c:\media\</pre>

The duration option is described in the following table:

Duration Option

Syntax	-d
Description:	<p>Specifies the amount of elapsed time before which encoding will stop when encoding from an input audio or video device. May also be used with file inputs.</p> <p>The format is d:h:m:s.ms.</p> <p>Note that if the duration is not provided, or if the value is zero, the duration is infinite.</p>
Examples:	<pre>-d 90 (90 seconds) -d 1:00:00 (1 hour)</pre>

The start time option is described in the following table:

Start Time Option

Syntax	-st
Description:	Specifies the start time within a file to begin the encode. The default value is 0 (beginning of the file) The format is d:h:m:s.ms. May be used with either duration (-d) or End Time (-et), or by itself (to indicate the end time is the end of the file).
Limitation:	-
Example:	-st 90 (begins encoding from 90 seconds into the file) -st 1:00:00 (begins encoding from 1 hour into the file.)

The end time option is described in the following table:

End Time Option

Syntax	-et
Description:	Specifies the end time within a file to end encoding. The format is d:h:m:s.ms. Note that if the end time is not provided, the end time will be the end of the file. In the same way, If the end time is beyond the duration of the file, then encoding will stop at the end of the file and a warning will be logged.
Limitation:	This flag cannot be used with -d
Examples:	-d 90 (stop encoding at 90 seconds from the beginning of the file.) -d 1:00:00 (stop encoding at 1 hour into the file.)

Outputs

The option described in the following table defines the file and server destinations and other outputs generated by the command line.

Output Option

Syntax	-o
Description:	The output filename or directory name. Note that this flag is optional for file inputs (-i).
Limitation:	See below.
Examples:	-o C:\Windows\Desktop\movie.rm -o \home\johndoe\commercialbreak.rm -o "C:\My Movies"

The value is interpreted as follows:

Input/Output interpretation		
Input (-i)	Output (-o)	Action
existing file	valid filename	The output will be written to the requested output filename
existing file or directory	existing directory	The output filename for each input file is derived from the input filename by replacing the input file extension with the appropriate output extension and written to the specified output directory.
directory	valid filename	This is interpreted as an error since it is unclear if the specified output was intended to be a directory or a filename. Helix Mobile Producer logs an error and stops encoding.
existing file or directory	Omitted	The output filename for each input file is derived from the input filename by replacing the input file extension with the appropriate output extension and written to the same directory as the input file.

Note: The file name or the directory name can be a relative or absolute path on a network or local drive. If a relative path is provided, the location is relative to the current working directory (the directory you executed producer from).

Warning! File size limits - If the output file size reaches 4 GB (limit of RealAudio/RealVideo file format) the output file is automatically rolled and a new file is started. The new file is appended with a number (e.g. movie1.rm).

There is no archiving or file rolling support for non-rm output formats.

Clip information

The clip information includes the descriptive metadata that is inserted into RealAudio/RealVideo files such as Title, Author and Copyright.

The title option is described in the following table:

Title Option	
Syntax	-t
Description:	Title to be inserted into all output media clips and live streams. The title value can be any user supplied string up to 255 bytes (i.e. 255 single-byte characters). For RealMedia, the string is ASCII. For 3GPP v6, the string is UTF-8. Example: This string gets inserted into the header of all output RealAudio/RealVideo and 3GPP v6 files and live streams. This information is exposed to the user through the player's Clip Info region.
Limitation:	-
Example:	-t "My Clip Title"

The author option is described in the following table:

Author Option	
Syntax	-a
Description:	Author to be inserted into all output media clips and live streams. Default: No default Values: Any user supplied string up to 255 bytes (i.e. 255 single-byte characters). For RealMedia, the string is ASCII. For 3GPP v6, the string is UTF-8. Example: -a "William Shakespeare" Details: This string gets inserted into the header of all output RealAudio/RealVideo and 3GPP v6 files and live streams. This information is exposed to the user through the player's Clip Info region
Limitation:	-
Examples:	-

The copyright option is described in the following table:

Copyright Option	
Syntax	-c
Description:	<p>Copyright to be inserted into all output media clips and live streams.</p> <p>Default: No default</p> <p>Values: Any user supplied string up to 255 bytes (i.e. 255 single-byte characters). For RealMedia, the string is ASCII. For 3GPP v6, the string is UTF-8.</p> <p>Example: -c "RealNetworks(c) 2004"</p> <p>Details: This string gets inserted into the header of all output RealAudio/RealVideo and 3GPP v6 files and live streams. This information is exposed to the user through the player's Clip Info region.</p>
Limitation:	-
Examples:	-

The keywords option is described in the following table:

Keywords Option	
Syntax	-k
Description:	<p>Keywords to be inserted into all output media clips and live streams.</p> <p>The keywords value can be any user supplied string up to 255 bytes (i.e. 255 single-byte characters).</p> <p>For RealMedia, the string is ASCII.</p> <p>For 3GPP v6, the string is UTF-8.</p> <p>Note that this string gets inserted into the header of all output RealAudio/RealVideo and 3GPP v6 files and live streams.</p>
Limitation:	-
Example:	-k "Keyword1 keyword2 keyword3"

Description Option

Syntax	-de
Description:	Description to be inserted into all output media clips and live streams. The description value can be any user supplied string up to 1023 bytes in size (i.e. 1023 single-byte characters). For RealMedia, the string is ASCII. For 3GPP v6, the string is UTF-8. Example: Note that this string gets inserted into the header of all output RealAudio/RealVideo and 3GPP v6 files and live streams.
Limitation:	Any text beyond the 64k character limit is truncated.
Example:	-de "A story about a galaxy far far away..."

The rating option is described in the following table:

Rating Option

Syntax	-r
Description:	Defines the age range for which the content is applicable to.
Limitation:	See tables below

Rating Option for 3GPP v6

Syntax	-r [RatingBody:]RatingCriteria[,RatingInfo] RatingBody is the rating body that defines the rating criteria being used. The MPAA for example. RatingCriteria is the rating for the content, such as <i>PG13</i> or <i>G</i> . RatingInfo is a text description of the rating.
Description:	Defines the rating body, the rating criteria, and a description of the rating. Possible values: see table below.
Limitation:	The only required parameter is the RatingCriteria. If the RatingBody is not entered, it is left blank.
Examples:	-r R -r MPAA:PG13 -r BBFC:R18,"Suitable only for adults." -r PG,"Parental guidance suggested."

3GPP v6 Possible Rating Values

Rating Body:	MPAA	BBFC	OFLC
Ratings:	G	U	G
	PG	PG	PG
	PG13	12	M
	R	15	MA
	NC17	18	R
		R18	X

In the case of 3GPP, the Rating box specifies both the rating body and the 4 character code for the rating. The table below defines some example values, as listed on <http://www.movie-ratings.net/>. See the 3GPP v6 Spec for more information. It is possible to specify any two 4-character codes for this option, separated by a comma, and followed by a text description

Rating Option for RealMedia

Syntax	-r
Description:	Possible Values: any integer from 0 to 6 as follows: 0 - No Rating 1 - All Ages 2 - Older Children 3 - Younger Teens 4 - Older Teens (15 & up) 5 - Adult Supervision Recommended 6 - Adults Only
Limitation:	If this option is not included in the command line or the property is specified as "Not Rated" then the Content Rating metadata field is not added to the output RealAudio/RealVideo file. For RealMedia: If this property is not passed on the command line or the property is specified as "Not Rated" then it is not set in the SDK and thus never gets added to the RealAudio/RealVideo file.
Example:	-r 0

Logs

The command-line will print out errors and warnings to the console as they are encountered. Each type of message will be preceded by a tag as shown in the examples below:

ERROR: "MyJob.hmpj" is not a valid job file.

WARNING: The input file duration is shorter than the specified end time. The encode will stop at the end of the file.

Specific options

Options in this section provide information about the system configuration. This information is useful in settings up an encoding session:

- -q Quite mode
- -h Print help
- -pd Print devices
- -v Print version

The quiet mode option is described in the following table:

Quiet options	
Syntax	-q
Description:	Prevents any output from being printed to the screen ever. this option can be useful if you are running the command line application from within another application that fails if data output to standard output or standard error.

The print help option is described in the following table:

Print Help Option	
Syntax	-h [Top] - [Top of Section]
Description:	Display short help on screen.

The print devices option is described in the following table:

Print Devices Option	
Syntax	-pd [Top] - [Top of Section]
Description:	This is an informational option that lists audio and video device information for the system. The intention is that the user can take this information to edit the devices specified in their job file. Note that only the first ten detected inputs are displayed.

Print Version Option	
Syntax	-v [Top] - [Top of Section]
Description:	This option prints version information for the command line application.

Stopping the encoding session,

To stop the current encoding session press **Ctrl+C**. This command stops the current encoding session and save the resulting output files (Note that some time may be required to merge SureStream files).

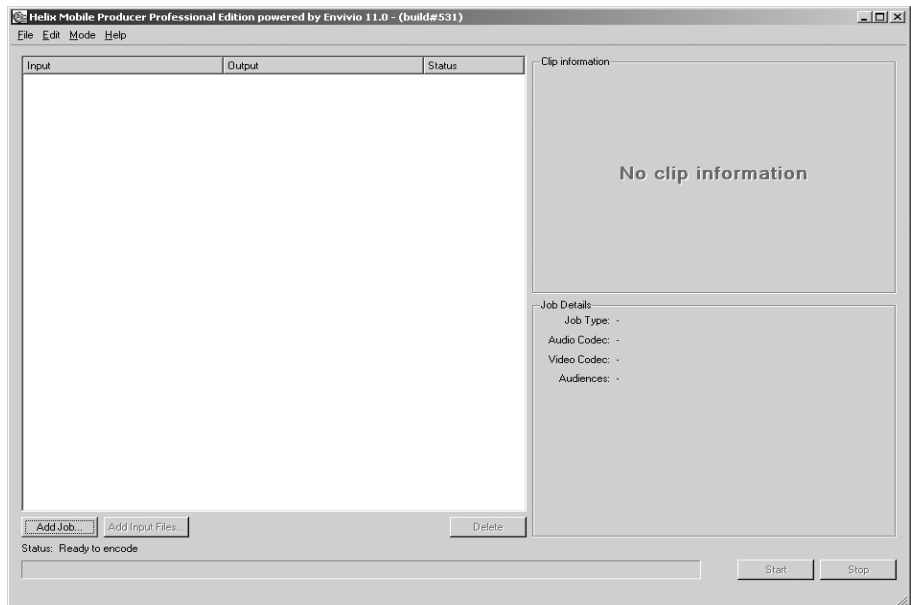
USING THE BATCH PROCESSOR

Helix Mobile Producer Professional gives you the ability to queue up encoding jobs and process them in batches.

Getting Started

To switch to batch processing mode, click **Mode>Batch Processor**. The following window is displayed:

Batch Encoder Window



Note: You can also toggle between the batch processor and the Helix Mobile Producer by pressing **Ctrl+M** (Main Interface) or **Ctrl+B** (Batch encoder).

Selecting a Job

You must have saved your own jobs if you want to use them in batch processing mode.

To select a job, click **Add Job...** and select the job you want to use.

Note: If the selected job contains an input file and output file, and if you have selected the **Respect job-defined input files...** option (see “Respect Job-Defined Input Files...” on page 72) they become the first task in the batch for that job.

Then you can:

- Add or delete input file(s)
- Specify the clip information for each file, or for all the files included in a job. See “Setting Metadata” on page 69 for more details on metadata parameters.

Tip: You can view the job properties by selecting the job name. The job properties are displayed in the **Job Details** panel.

Adding Input Files

To add media to the job, click **Add Input files...** and select the file you want to encode.

Deleting Input Files

To remove a media from the job, select the input file you want to delete then click **Delete**.

Specifying the Output File Name

You can specify the MPEG-4 output file name and path by clicking the input file name. Then you can change the file name and path.

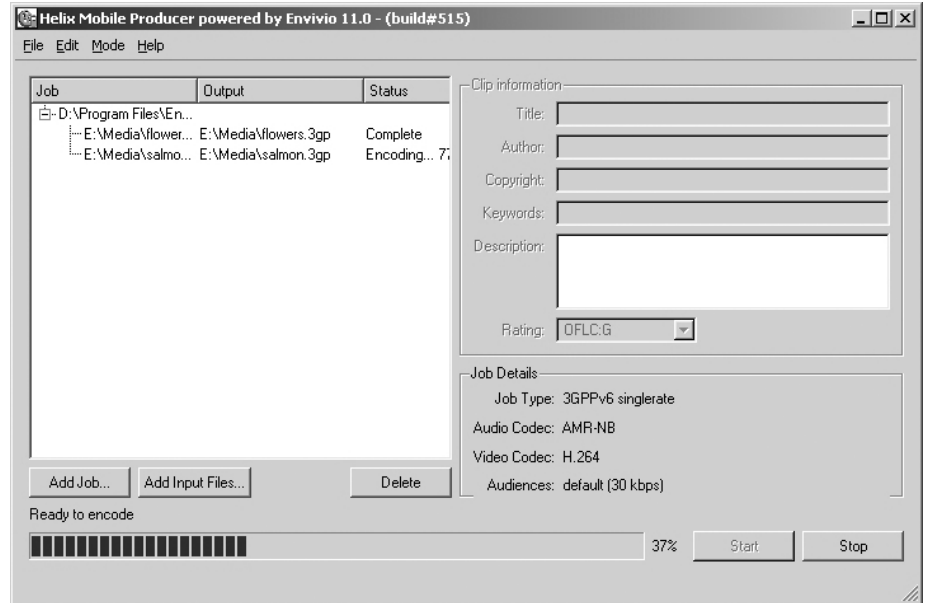
Note: By default, the output directory is the same as the input directory. You can specify a custom directory from the **Preferences** dialog box (see “Setting Preferences” on page 72).

Starting the batch encoding

To start the batch encoding, click the **Start** button.

Understanding the Batch Encoding Process

As each job is run, Helix Mobile Producer applies the selected job options, and starts encoding.



The progress bar indicates the global progression of the jobs.

The progression of each job is described in the **Status** column. The status of the job may be **Queued**, **Encoding** (followed by the encoded ratio) or **Complete**.

If an error occurs, Helix Mobile Producer stops the operation, records the error, and starts the next job.

AUDIENCE FILE REFERENCE

This appendix describes the audience syntax. Each audience file records settings about how clips are encoded. The information in this appendix allows you to edit audience information to modify encoding settings.

Understanding Audiences

Each audience file defines a single audience for which a clip is encoded. Helix Mobile Producer predefines a number of audiences that appear within the graphical user interface when you click the **Audio/Video streams** tab. For example, one audience may be for 20K. Each audience specifies the streaming rates at which audio and video clips are encoded, along with other settings.

Audience Files

The audience files are stored in the audiences subdirectory under the RealNetwork directory in the main Helix Mobile Producer installation directory. Audience files use the file extension .xml. The beginning of their file names correspond to the audience name in the graphical interface, followed by the video encoder type, the audio encoder type and the total bit rate. For example, when you select 3GPPv5 as export type, the **20k GPRS Voice** audience in the graphical user interface stores its information in the file named **20k_GPRS_Voice-h264-amrnb_20kbps.xml**.

Some settings in the audience files, such as the video codec, the frame rate, keyframe interval, and buffer size, can be overridden using the GUI interface. You can also edit audience files manually. This allows you to change audience information used by the graphical interface or command line application using a text editor or any automated process that can modify text files.

Tip: When creating a new audience file, RealNetworks recommends that you start with an existing audience file that you have renamed.

Audience Section

The `<audience>` and `</audience>` tags encapsulate an audience section within an audience file.

```
<audience xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance">
  ...all audience parameters
</audience>
```

Audience example

```
<audience xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance">
  <version>11.1</version>
  <audienceName>20k GPRS Voice</audienceName>
  <audienceAvgBitRate>20000</audienceAvgBitRate>
  <information />
  <audienceMaxBitRate>25000</audienceMaxBitRate>
  <videoEncoder xsi:type="h264">
    <keyFramePeriodInMs>5000</keyFramePeriodInMs>
    <maxFrameRate>5</maxFrameRate>
    <encodingComplexity>high</encodingComplexity>
    <videoMode>normal</videoMode>
  </videoEncoder>
  <audioEncoder xsi:type="amrnb">
    <bitRate>5150</bitRate>
    <useDTX>false</useDTX>
  </audioEncoder>
</audience>
```

Video and Audio Streams

Each audience template has video and audio streams. The video section starts and ends with `<videoEncoder>` and `</videoEncoder>` tags. The audio section starts and ends with `<audioEncoder>` and `</audioEncoder>` tags.

Video Stream examples

For More Information: Appendix C lists the video codecs along with the parameters values.

MPEG-4 SP encoder

```

<videoEncoder xsi:type="mpeg4sp">
  <keyFramePeriodInMs>5000</keyFramePeriodInMs>
  <maxFrameRate>5</maxFrameRate>
  <encodingComplexity>high</encodingComplexity>
  <videoMode>normal</videoMode>
</videoEncoder>

```

H.263 encoder

```

<videoEncoder xsi:type="h263">
  <keyFramePeriodInMs>5000</keyFramePeriodInMs>
  <maxFrameRate>5</maxFrameRate>
  <encodingComplexity>high</encodingComplexity>
  <videoMode>normal</videoMode>
</videoEncoder>

```

H.264 encoder

```

<videoEncoder xsi:type="h264">
  <keyFramePeriodInMs>5000</keyFramePeriodInMs>
  <maxFrameRate>5</maxFrameRate>
  <encodingComplexity>high</encodingComplexity>
  <videoMode>normal</videoMode>
</videoEncoder>

```

Real Media encoder

```

<videoEncoder xsi:type="realVideo10">
  <videoMode>normal</videoMode>
  <keyFramePeriodInMs>5000</keyFramePeriodInMs>
  <encodingComplexity>high</encodingComplexity>
  <maxFrameRate>15</maxFrameRate>
</videoEncoder>

```

Audio Stream Examples

For More Information: Appendix C lists the audio codecs along with the parameters values.

AAC encoder

```

<audioEncoder xsi:type="aac">

```

```
<bitRate>128000</bitRate>
<useMS>true</useMS>
<advAudio>
  <channelMode>stereo</channelMode>
  <samplingRate>44.100000</samplingRate>
</advAudio>
</audioEncoder>
```

AAC Plus encoder

```
<audioEncoder xsi:type="aacPlus">
  <bitRate>16000</bitRate>
</audioEncoder>
```

Enhanced AAC Plus encoder

```
<audioEncoder xsi:type="enhancedAacPlus">
  <bitRate>48000</bitRate>
</audioEncoder>
```

AMR-NB encoder

```
<audioEncoder xsi:type="amrnb">
  <bitRate>12200</bitRate>
  <useDTX>false</useDTX>
</audioEncoder>
```

AMR-WB encoder

```
<audioEncoder xsi:type="amrwb">
  <bitRate>15850</bitRate>
  <useDTX>true</useDTX>
</audioEncoder>
```

QCELP encoder

```
<audioEncoder xsi:type="qcelp">
  <bitRate>14000</bitRate>
</audioEncoder>
```

MP3 encoder

```
<audioEncoder xsi:type="mp3">
  <bitRate>96000</bitRate>
</audioEncoder>
```

RealAudio encoder

```
<audioEncoder xsi:type="realAudio">  
  <codecFlavor>cook19</codecFlavor>  
</audioEncoder>
```


JOB FILE REFERENCE

This appendix provides a reference for job files created in Helix Mobile Producer, describing the format and syntax of job file encoding parameters for audio and video input. It shows the basic structure of a job file and gives you details about the different sections of a job file.

Creating Job Files

A job file is an XML file that uses the following format:

```
<job build="539" exportType="3gpp2" version="11.1"
xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance">
<clipInformation>
  <title />
  <author />
  <copyright />
  <mediaKeywords />
  <description />
  <mediaRating>MPAA:G</mediaRating>
</clipInformation>
<inputs>
  <input xsi:type="avFileInput">
    <fileName />
    <prefilters>
      <audioPreFilters />
      <videoPreFilters>
        <enableResize>true</enableResize>
        <resizeWidth>176</resizeWidth>
        <resizeHeight>144</resizeHeight>
      </videoPreFilters>
    </prefilters>
  </input>
</inputs>
<outputs>
```

```
<fileOutput>
<fileName />
</fileOutput>
</outputs>
<encodingParameters>
  <rateControlMode>cbr</rateControlMode>
  <numberOfPass>2</numberOfPass>
  <encodeVideo>>true</encodeVideo>
  <encodeAudio>>true</encodeAudio>
  <exportSettings>
    <progressiveDownload>>true</progressiveDownload>
    <hinted>>true</hinted>
    <MTUSize>1400</MTUSize>
  </exportSettings>
  <audiences>
    <audience>
      <version>11.1</version>
      <audienceName>44k CDMA Voice</audienceName>
      <audienceAvgBitRate>44000</audienceAvgBitRate>
      <information />
      <audienceMaxBitRate>44000</audienceMaxBitRate>
      <videoEncoder xsi:type="mpeg4sp">
        <keyFramePeriodInMs>5000</keyFramePeriodInMs>
        <maxFrameRate>5</maxFrameRate>
        <encodingComplexity>high</encodingComplexity>
        <videoMode>normal</videoMode>
      </videoEncoder>
      <audioEncoder xsi:type="qcelp">
        <bitRate>14000</bitRate>
      </audioEncoder>
    </audience>
  </audiences>
</encodingParameters>
</job>
```

Input Parameters

This section defines the properties, or parameters, that you can set for import file formats and live capture sources for audio and video input. Note that some of these properties are optional.

File Input

You can set the file name parameter for file input, as shown in the following table and example.

File Name Parameter

Parameter ID	Definition	Value
file name	Input file for offline encoding	Absolute path to the input file name

Example

```
<input>
  <file name="D:\movies\my_movie.avi" />
</input>
```

Note: Helix Mobile Producer does not support live encoding of file input for broadcast output.

Live Capture Input

The live capture input parameters you can set for Helix Mobile Producer are listed and defined in the following table. You can set these parameters as shown in the example after the table.

Live Capture Input Parameters

Parameter ID	Definition	Value	Optional
audioDeviceID	Input audio capture source	—	No
videoDeviceID	Input video capture source	_	No
videoDeviceWidth	Width of the input video image in pixels	—	Yes
videoDeviceHeight	Height of the input video image in pixels	—	Yes

Example

```
<input xsi:type="captureInput">
  <audioDeviceID>0</audioDeviceID>
  <videoDeviceID>0</videoDeviceID>
  <videoDeviceWidth>720</videoDeviceWidth>
  <videoDeviceHeight>576</videoDeviceHeight>
  <prefilters>
    <audioPreFilters />
```

```
<videoPreFilters>
  <enableResize>true</enableResize>
  <resizeWidth>352</resizeWidth>
  <resizeHeight>288</resizeHeight>
</videoPreFilters>
</prefilters>
</input>
```

Prefilters

The prefilters you can set for Helix Mobile Producer are shown in the following examples.

For More Information: The prefilters parameters are listed at the end of the Input parameters. For more information on these parameters, refer to “Setting Audio and Video Filters” on page 60.

Example

```
<inputs>
<input xsi:type="avFileInput">
  <fileName>E:\contents\my_movie.avi</fileName>
  <prefilters>
    <audioPreFilters>
      <audioGain>10</audioGain>
      <audioBalance>20</audioBalance>
    </audioPreFilters>
    <videoPreFilters>
      <enableCropping>true</enableCropping>
      <croppingLeft>0</croppingLeft>
      <croppingTop>0</croppingTop>
      <croppingWidth>720</croppingWidth>
      <croppingHeight>480</croppingHeight>
      <enableResize>true</enableResize>
      <resizeWidth>720</resizeWidth>
      <resizeHeight>480</resizeHeight>
      <enableDeinterlace>true</enableDeinterlace>
      <enableInverseTelecine>true</enableInverseTelecine>
      <inverseTelecineFieldOrder>1</inverseTelecineFieldOrder>
      <enableLogoInsertion>true</enableLogoInsertion>
      <logoInsertionFile>E:\Documents\logo.bmp</logoInsertionFile>
```

```

<logoInsertionLeft>25</logoInsertionLeft>
<logoInsertionTop>30</logoInsertionTop>
<enableNoiseReductionAdjustment>true</enableNoiseReductionAdjustment>
<noiseReductionAdjustmentLevel>2</noiseReductionAdjustmentLevel>
<enableBrightnessAdjustment>true</enableBrightnessAdjustment>
<brightnessAdjustmentLevel>10</brightnessAdjustmentLevel>
<enableContrastAdjustment>true</enableContrastAdjustment>
<contrastAdjustmentLevel>-10</contrastAdjustmentLevel>
<enableGammaAdjustment>true</enableGammaAdjustment>
<gammaAdjustmentRed>10</gammaAdjustmentRed>
<gammaAdjustmentGreen>-10</gammaAdjustmentGreen>
<gammaAdjustmentBlue>40</gammaAdjustmentBlue>
</videoPreFilters>
</prefilters>
</input>
</inputs>

```

Output Parameters

This section defines the parameters that you can set for audio or video output files or broadcasts. Note that some of these properties are optional.

File Output

You can set the file name parameter for file output, as shown in the following table and example.

File Name Parameter

Parameter ID	Definition	Value
file name	Output file for offline encoding	Absolute path to the output file name

Example

```

<output>
  <file name="output.3gp"/>
</output>

```

Helix Broadcast Output

There are a number of Helix broadcast output parameters that you can set, as shown in the following examples.

For More Information: For more information on Helix Broadcast parameters, refer to “Helix Broadcast Mode” on page 36.

Push Account-Based

Example

```
<outputs>
  <broadcastOutput xsi:type="pushServerAccountBased">
    <streamname>live.rm</streamname>
    <address>192.168.105.187</address>
    <port>80</port>
    <username>admin</username>
    <password>toto</password>
    <savePassword>true</savePassword>
    <transport>udp</transport>
    <listenAddress>0</listenAddress>
    <fecPercent>0</fecPercent>
    <fecOffset>0</fecOffset>
    <allowResends>true</allowResends>
    <enableTCPReconnect>true</enableTCPReconnect>
    <TCPReconnectInterval>30</TCPReconnectInterval>
  </broadcastOutput>
</outputs>
```

Push Password-Only

Example

```
<outputs>
  <broadcastOutput xsi:type="pushServerPasswordOnly">
    <streamname>live.rm</streamname>
    <address>192.168.105.187</address>
    <port>3020</port>
    <endPort>3050</endPort>
    <password>envivio</password>
    <savePassword>true</savePassword>
    <transport>udp</transport>
    <listenAddress>0</listenAddress>
    <fecPercent>0</fecPercent>
    <fecOffset>0</fecOffset>
    <allowResends>true</allowResends>
  </broadcastOutput>
</outputs>
```

```

        <enableTCPReconnect>true</enableTCPReconnect>
        <TCPReconnectInterval>30</TCPReconnectInterval>
    </broadcastOutput>
</outputs>

```

Push Multicast

Example

```

<outputs>
  <broadcastOutput xsi:type="pushServerMulticast">
    <streamname>live.rm</streamname>
    <port>3020</port>
    <endPort>3050</endPort>
    <password>envivio!</password>
    <savePassword>true</savePassword>
    <transport>udp</transport>
    <listenAddress />
    <multicastAddress>224.224.224.224</multicastAddress>
    <multicastTTL>3</multicastTTL>
    <fecPercent>0</fecPercent>
    <fecOffset>0</fecOffset>
    <metadataResendInterval>30</metadataResendInterval>
  </broadcastOutput>
</outputs>

```

Pull

Example

```

<outputs>
  <broadcastOutput xsi:type="pullServer">
    <streamname>live.rm</streamname>
    <listenAddress>0</listenAddress>
    <listenPort>3031</listenPort>
    <password>envivio!</password>
    <savePassword>true</savePassword>
    <serverTimeout>30</serverTimeout>
  </broadcastOutput>
</outputs>

```

RTP Broadcast Output

There are five RTP broadcast output parameters that you can set, as shown in the following table and example.

RTP Broadcast Output Parameters		
Parameter ID	Definition	Value
broadcastOutput xsi:type	Broadcast mode selection	rtp
sdpFileName	Session description protocol file (or <i>announce file</i>), which provides information about a streamed Web broadcast	Absolute path to the announce file
destinationAddress	IP address for a multicast	IP address
destinationPort	Destination port for RTP broadcast mode	Port number
TTL	Amount of time left until a live multicast begins	1 through 15

Example

```
<outputs>
  <broadcastOutput xsi:type="rtp">
    <sdpFileName>E:\workspace\hmp11_beta\zola2\bin\Release\live.sdp</sdpFileName>
    <destinationAddress>127.0.0.1</destinationAddress>
    <destinationPort>5050</destinationPort>
    <TTL>3</TTL>
  </broadcastOutput>
</outputs>
```

Export Settings

The export settings you can set for Helix Mobile Producer are shown in the following examples.

For More Information: The export settings parameters are listed in the encoding parameters. For more information on these parameters, refer to “Setting Export Parameters” on page 65.

Example

```

<clipInformation>
  <title />
  <author />
  <copyright />
  <mediaKeywords />
  <description />
  <mediaRating>Not rated</mediaRating>
</clipInformation>

```

Metadata

The metadata you can set for Helix Mobile Producer are shown in the following examples.

For More Information: For more information on these parameters, refer to “Setting Metadata” on page 69.

Example

```

<clipInformation>
  <title>My clip</title>
  <author>Unknown</author>
  <copyright />
  <mediaKeywords>Music</mediaKeywords>
  <description>Music clip</description>
  <mediaRating>All ages</mediaRating>
</clipInformation>

```


AUDIO CODEC REFERENCE

This appendix provides a reference for all codecs used by Helix Mobile Producer. The first part describes all RealAudio codecs, broken down into separate tables for voice, mono music, stereo and surround sound music codecs. The second part describes all the other codecs.

RealAudio Codecs

Using the RealAudio Codec Reference Tables

The codecs below are grouped into tables according to the type of application such as voice or music and provide detailed information about each codec.

RealAudio Codec Name

This column describes the name of the codec as it appears in the Audiences dialog. (To access the Audience Templates dialog, click the **Audio/Video streams tag** and click the **Template** name you want to use.)

The name gives you vital information about the codec: the bit rate for the codec, the type of audio the codec is suited for, and if the codec is a “high response” codec.

High response codecs cover high frequency sounds, such as a flute, than normal codecs at the same bit rate. But they don’t cover low frequencies, such as a bass drum, as well as normal codecs.

RealAudio Codec ID

The codec flavor column identifies the codec number you can use when creating XML job files.

Sampling Rate

The Sampling Rate column lists the codec’s optimum sample rate. Using a codec’s optimum sampling rate in your audio source file ensures that the

audio stays synchronized with other media and prevents pitch shifting during audio resampling. Audio quality degrades if you use lower than the optimum sampling rate.

Audio input should be set at the same sample rate as required by the desired output codec. In the case of SureStream encoding, the input sample rate should be equal or greater than the largest sample rate of all codecs included in the output.

If you use a higher sampling rate when creating the source audio, it is best to use a multiple of the optimum rate. If the optimum rate is 8 kHz, for example, use a higher rate of 16 kHz or 32 kHz. When in doubt, use a CD-quality sampling rate of 44.1 kHz for your source audio.

Frequency Response

A codec with a higher frequency response reproduces a wider range of sound than a codec with a lower response. **High Response** is indicated after the coder name

Frequency response measures how much of the original audio's frequency range will be used in creating the RealAudio clip. Thus, the codec does not enhance produced audio's quality; it always results in a clip of equal or lower quality than the original audio. For example, if the original audio has an 8 kHz frequency response, encoding it with a codec that has a frequency response of 10 kHz produces a clip that still has a response of 8 kHz.

For More Information: The audio preparation chapter of *RealNetworks Production Guide* has an expanded list that covers all RealAudio codecs, including obsolete codecs no longer used by Helix Producer.

Compatibility

Some codecs are available for use with early versions of the Real Player. Compatibility is indicated below each codec type.

Voice Codecs

Helix Mobile Producer uses a voice codec when you encode a voice-only or voice-with-music clip. The lowest-speed voice codec normally used with RealAudio is 16 kbps. The lower-speed codecs can be used as "duress" streams in SureStream clips. They are also used to encode soundtracks for low-bandwidth RealVideo clips.

Available Voice Codecs

Helix Mobile Producer offers several codecs designed specifically for encoding voice.

RealAudio Voice Codecs

Codec Name	Codec ID	Sampling Rate
5 Kbps voice	sipr2	8 kHz
6.5 Kbps Voice	sipr0	8 kHz
8.5 Kbps Voice	sipr1	8 kHz
16 Kbps Voice	sipr3	16 kHz
32 Kbps Voice	cook7	22.05 kHz
64 Kbps Voice	cook14	44.1 kHz

Player Compatibility

- The sipr 6.5Kbps and 8.5Kbps codecs are compatible with RealPlayer 4 and later.
- The sipr 5Kbps and 15 Kbps codecs are compatible with RealPlayer 5 and later.
- The cook 32Kbps and 64Kbps codecs are compatible with RealPlayer G2 and later.

Music Audio Codecs

Music codecs create a higher quality of sound for audio recordings than voice codecs. Music Audio codecs are designed to encode audio with a larger pitch variance than voice. You will capture a broader, fuller sound with codecs designed for higher bit rates, but your files will be larger.

The lower-speed codecs can be used as “duress” streams in SureStream clips, and to encode soundtracks for low-bandwidth RealVideo clips. When there are two versions of a codec, Helix Producer uses the high response version by default.

About High-Response Codecs

The 20 kbps and 32 kbps music codecs come in two flavors. Helix Producer by default uses the “high response” versions, which are the better codecs for most situations. But you can also use the “normal response” versions.

The high response codecs cover a larger frequency spectrum than the normal response versions. Sometimes, the high response version has twice the range as the normal codec. This means it provides crisper sound and is better at capturing high frequencies. With symphonic music, for example, the high response codec gets more of the flute and piccolo. It can produce more distortion than the normal response codec with voices and loud sounds such as drums, though.

If you are encoding music with a diverse range of frequencies, stick with the high response codecs. If you notice distortion, compare your results with a clip that uses the normal response codecs. The best tool for determining which codec to use is your ear. Listen carefully for minute differences in how the clip sounds. It also helps to have other people listen. Our own ears have different frequency responses, too.

The slowest stereo codec is 12 kbps. Stereo codecs don't go lower than that because they would not have enough frequency response for adequate sound.

Available Audio Codecs

The wide of available codec bit rates allow you encode your files for your exact audience needs.

Music Audio Codecs

Codec Name	Codec ID	Sampling Rate
6 Kbps Music	cook8	8kHz
8 Kbps Music	cook0	
11 Kbps Music	cook1	8 kHz
16 Kbps Music	cook2	8 kHz
20 Kbps Music	cook3	11.025 kHz
20 Kbps Music High Response	cook15	22.05 kHz
32 Kbps Music	cook4	22.05 kHz
32 Kbps Music High Response	cook16	44.1 kHz
44 Kbps Music	cook5	44.1 kHz
64 Kbps Music	cook6	44.1 kHz

Player Compatibility

- All Music Audio Codecs are compatible with RealPlayer G2 and later.

Stereo Music Audio Codecs

Helix Mobile Producer contains an array of RealAudio codecs that are optimized for stereo music. These RealAudio codecs enable you to stream high-quality stereo music at a wide range of bandwidths.

Because the output is designed for dual speaker systems, your music will reach a large audience with varying systems.

Available Music Audio Codec

A variety music audio codecs are now available in Helix Mobile Producer.

Stereo Music Audio Codecs

Codec Name	Codec ID	Sampling Rate
12 Kbps Stereo Music	cook26	11.025 kHz
16 Kbps Stereo Music	cook17	22.05 kHz
20 Kbps Stereo Music (older version)	cook9	11.025 kHz
20 Kbps Stereo Music	cook18	22.05 kHz
20 Kbps Stereo Music High Response	cook19	22.05 kHz
32 Kbps Stereo Music (older version)	cook10	22.05 kHz
32 Kbps Stereo Music	cook20	22.05 kHz
32 Kbps Stereo Music High Response	cook21	44.1 kHz
44 Kbps Stereo Music (older version)	cook11	22.05 kHz
44 Kbps Stereo Music	cook22	44.1 kHz
44 Kbps Stereo Music High Response	cook23	44.1 kHz
64 Kbps Stereo Music (older version)	cook12	44.1 kHz
64 Kbps Stereo Music	cook24	44.1 kHz
96 Kbps Stereo Music (older version)	cook13	44.1 kHz
96 Kbps Stereo Music	cook25	44.1 kHz
64 kbps Stereo Music - RealAudio 10	raac0	44.1 kHz
96 kbps Stereo Music - RealAudio 10	raac1	44.1 kHz
128 kbps Stereo Music - RealAudio 10	raac2	44.1 kHz
160 kbps Stereo Music - RealAudio 10	raac3	44.1 kHz
192 kbps Stereo Music - RealAudio 10	raac4	44.1 kHz
256 kbps Stereo Music - RealAudio 10	raac5	44.1 kHz
320 kbps Stereo Music - RealAudio 10	raac6	44.1 kHz
44 kbps Stereo Surround	cook29	22.05 kHz

Stereo Music Audio Codecs (continued)

Codec Name	Codec ID	Sampling Rate
64 kbps Stereo Surround	cook27	44.1 kHz
96 kbps Stereo Surround	cook28	44.1 kHz
128 kbps Stereo Surround - RealAudio 10	raac7	44.1 kHz
160 kbps Stereo Surround - RealAudio 10	raac8	44.1 kHz
192 kbps Stereo Surround - RealAudio 10	raac9	44.1 kHz
256 kbps Stereo Surround - RealAudio 10	raac10	44.1 kHz
320 kbps Stereo Surround - RealAudio 10	raac11	44.1 kHz
96 kbps 5.1 Multichannel - RealAudio 10	cook30	22.05 kHz
128 kbps 5.1 Multichannel - RealAudio 10	cook31	44.1 kHz
180 kbps 5.1 Multichannel - RealAudio 10	cook32	44.1 kHz
264 kbps 5.1 Multichannel - RealAudio 10	cook33	44.1 kHz

Player Compatibility

- All RealAudio 10 codecs are compatible with RealPlayer 10 and later.
- All other stereo music codecs are compatible with RealPlayer G2 and later.

Other Audio Codecs

Several AMR, AAC, QCELP and MP3 audio codecs are now available in Helix Mobile Producer.

AMR, AAC, QCELP and MP3 Audio Codecs

Codec Name	Codec ID	Sampling Rate
7.4 kbps AMR Narrow Band	amrnb	8 khz
7.95 kbps AMR Narrow Band	amrnb	8 khz
10.2 kbps AMR Narrow Band	amrnb	8 khz
12.2 kbps AMR Narrow Band	amrnb	8 khz
6.6 kbps AMR Wide Band	amrwb	16 khz
8.85 kbps AMR Wide Band	amrwb	16 khz
12.65 kbps AMR Wide Band	amrwb	16 khz
14.25 kbps AMR Wide Band	amrwb	16 khz
15.85 kbps AMR Wide Band	amrwb	16 khz
18.25 kbps AMR Wide Band	amrwb	16 khz
19.85 kbps AMR Wide Band	amrwb	16 khz

AMR, AAC, QCELP and MP3 Audio Codecs (continued)

Codec Name	Codec ID	Sampling Rate
23.05 kbps AMR Wide Band	amrwb	16 khz
23.85 kbps AMR Wide Band	amrwb	16 khz
AAC Low Complexity	aac	variable
aacPlus	aacPlus	variable
Enhanced aacPlus	enhancedAacPlus	variable
QCELP half-rate	qcelp	8 khz
QCELP full-rate	qcelp	8 khz
MP3	mp3	variable

FILE AND DEVICE COMPATIBILITY

This appendix provides a reference for compatibility among the various file formats and numerous codecs supported by Helix Mobile Producer. It also briefly addresses the compatibility of these formats and codecs with the wireless mobile devices currently available.

File Format and Codec Compatibility

Helix Mobile Producer supports six audio codecs (not including the RealAudio codecs), three video codecs, and seven file formats, not all of which are compatible with one another. Refer to the following tables to determine whether a given audio or video codec is supported within a particular file format. An X in a codec's column means that the codec is compatible with the file type listed on the left in that row.

Audio Codec Compatibility

The following table provides a reference for audio codec compatibility.

Audio Codec Compatibility with Supported File Formats

File formats	Audio codecs							
	AAC	AAC Plus	Enhanced		AMR-WB	QCELP	RealAudio	MP3
			AAC Plus	AMR-NB				
ISMA Profile 0	X	-	-	-	-	-	-	-
3GPP2						X		
3GPP v5	X	-	-	X	X	-	-	-
3GPP v6 singlerate	X	X	X	X	X			
3GPP v6 multirate	X	X	X	X	X	-	-	-
AMR				X	X			
QCELP	-	-	-	-	-	X	-	-
MP3								X
RealMedia	-	-	-	-	-	-	X	-

Video Codec Compatibility

The following table provides a reference for video codec compatibility.

Video Codec Compatibility with Supported File Formats

File formats	Video codecs			
	RealVideo	MPEG-4 SP	H.264	H.263
ISMA Profile 0	-	X	-	-
3GPP2		X		
3GPP v5	-	X	-	X
3GPP v6 singlerate		X	X	X
3GPP v6 multirate	-	X	X	X
AMR				
QCELP	-	-	-	-
MP3				
RealMedia	X	-	-	-

Device Compatibility

Not all wireless mobile devices are compatible with all of the file formats and codecs listed in the preceding tables. To determine which formats and codecs a particular mobile device supports, see the manufacturer's documentation for that device.

SNMP MONITORING

This appendix explains how to set up Simple Network Management Protocol (SNMP) support for Helix Mobile Producer Professional and Helix Mobile Producer Professional Live on Windows operating systems. It also includes a reference table for supported SNMP objects that provide management information about Helix Mobile Producer Professional and Helix Mobile Producer Professional Live.

The SNMP Service

Helix Mobile Producer Professional and Helix Mobile Producer Professional Live support the Windows SNMP service. You can use an SNMP browser to monitor (remotely) computers running Helix Mobile Producer Professional or Helix Mobile Producer Professional Live. Note that the management information base (MIB) gets installed on your computer *only* if the Windows SNMP service is already installed there.

If you decide to install the Windows SNMP service after installing Helix Mobile Producer Professional or Helix Mobile Producer Professional Live, you must uninstall Helix Mobile Producer, install the SNMP service, and then reinstall Helix Mobile Producer.

Note: When you install Helix Mobile Producer Professional or Helix Mobile Producer Professional Live, select the **Install SNMP agent** check box on the final page of the Setup wizard.

Installing the Windows SNMP Service

The following procedure shows you how to do an SNMP installation on a computer running Windows 2000.

► To install the SNMP service:

1. Insert the Windows 2000 or Windows XP CD into your disk drive.
2. Click **Install Additional Components** (Windows 2000) or **Install Optional Windows Components** (Windows XP).
3. Click **Management and Monitoring Tools**.
4. Click **Next**.

This starts the installation process for the SNMP service.

Accessing the Helix Mobile Producer MIB

The MIB file is located in the Helix Mobile Producer Professional or Helix Mobile Producer Professional Live installation directory, **HMP-MIB11.mib**.

SNMP Object Description

The following table provides the names and descriptions of all of the monitored objects in the Helix Mobile Producer Professional MIB file.

Monitored SNMP Objects	
SNMP object	Description
cpuUsage	CPU usage for the machine
inputFile	Full path of the input source file, or "none"
audioDevice	Source audio device and port, or "none"
videoDevice	Source video device and port, or "none".
targetPlayer	Player that this encode is targeted for (Players list)
compatiblePlayers	Players that are compatible with this encode (Players list)
outputFile	Full path and name of output file, or "none"
outputStream	Live broadcast destination
outputSDPFile	Full path and name of the output SDP file, or 'none'
clipDuration	Duration of the clip in hh:mm:ss format
timestamp	Current time stamp in hh:mm:ss format
clipSize	Total file size of the clip in bytes
totalBitRate	Total bit rate of all streams in bits per second

Monitored SNMP Objects (continued)

SNMP object	Description
streamCount	number of stream pairs in the active job.
The following objects are repeated for each audience. X represents the audience index (from 1 to 8).	
streamX_audioCodecName	Name of the audio codec; see audio codec tables in Appendix C.
streamX_audioCodecID	ID of the audio codec; see audio codec tables in Appendix C.
streamX_audioBitRate	Audio codec bitrate in bits per second
streamX_audioFrequency	Frequency of the audio codec in Hz.
streamX_audioLatency	audio latency in milliseconds
streamX_audioCPU	Percentage of cpuUsage used for encoding this audio stream.
streamX_videoCodecName	The name of the video codec
streamX_videoCodecID	The ID of the video codec.
streamX_videoTargetBitRate	The target video bit rate in bits per second
streamX_videoBitRate	The current video bit rate in bits per second
streamX_videoLatency	the latency of the video stream in milliseconds
streamX_videocpu	Percentage of cpuUsage used for encoding this video stream.
streamX_videoQualityIndex	Quality of the video as a percentage of the source quality
streamX_videofps	Average frame rate in frames per second x 100.
streamX_videofpsMax	Maximum frame rate in frames per second x 100.
streamX_videofpsMin	Minimum frame rate in frames second x 100.

GLOSSARY

A AAC

Advanced Audio Coding, for MPEG-2 or MPEG-4 files.

ADSL

Asymmetric digital subscriber line. A high-speed Internet connection carried over telephone lines.

AIFF, AIFC

Audio Interchange File Format. A format for storing digital audio samples in a file. AIFC, a newer version of this format, can be used for compressing audio files.

artifacts

Blemishes, noise, spots, or other flaws in video, audio, or image production in MPEG files.

AVI

Audio/Video Interleaved. A multimedia file format.

B bandwidth

Bandwidth has many meanings, depending on the context. It started as a radio term and has been expanded to include other kinds of communications. It is the numeric difference between the highest and lowest frequencies of a radio band or channel.

Bandwidth means the same thing in relation to audio data, although in most applications, the lowest frequency is sufficiently low enough to be considered equal to zero. Hence, audio bandwidth usually means the highest frequency of an audio signal or the highest frequency that can be carried by an audio system.

Typical examples of bandwidth are 300–3,400 Hz for telephone lines, 20–20,000 Hz for hi-fi systems, and 50–15,000 Hz for FM radio signals.

BIFS

Binary format for scenes. A set of nodes, based on VRML (Virtual Reality Modeling Language), that make it possible to combine 2D and 3D

graphics, natural and synthetic sound, audio and video information, and stored and streamed material in one environment.

bit rate

The rate of data transmission over a computer network. Helix Mobile Producer uses several bit rates. The target bit rate is the channel capacity of the network. The maximum bit rate is a limit, less than the target bit rate, that Helix Mobile Producer uses to fit data into the actual bandwidth for a streaming server. The audio encoder uses an average target bit rate and a maximum target bit rate for variable-bit-rate encoding.

broadband

A general term for high-speed Internet connections provided by DSL lines, cable modems, and T1 lines.

C

CBR

Constant bit rate.

CELP

Code Excited Linear Prediction. An audio-encoding scheme used for very low-bit-rate encoding, mainly for speech.

CIF (pronounced "siF")

Common Intermediate Format. A video display size (352 x 288 pixels) well suited to higher bit rates.

codec

The software that encodes and decodes an audio or video file. The word is a shortened form of coder-decoder or compression-decompression.

E

entropy

A measure of the degree of disorder or chaos in a system. In the context of streaming media, as the entropy value increases, data compression becomes more difficult.

H

hinted movies

Movies that can be streamed for broadcast over the Internet. A "hint track" contains information about the media file that enables the server to stream it properly.

I

I-frames

An MPEG-4 term for video key frames.

Internet

An interconnected system of networks that connects computers around the world, using the TCP/IP protocol.

ISO

International Organization for Standards. ISO is a network of the national standards institutes of 146 countries. ISO standards are technical agreements that provide the framework for compatible technology worldwide.

K Kbps

Kilobits per second. A standard measure of the rate of data transmission over electronic devices.

key frame

A video frame that fully refreshes its contents and can be used as a reference point for other frames in the video sequence. In MPEG terminology, this is known as an I-frame.

M motion estimation

For video, a method that estimates changes between frames.

.mov

The Apple QuickTime file name extension for movie files.

MPEG (pronounced “m-peg”)

As defined on the MPEG Web site, MPEG stands for Moving Pictures Experts Group and is the name given to a family of international standards for coding audiovisual information in a compressed digital format. The MPEG family of standards includes MPEG-1, MPEG-2, MPEG-4, MPEG-7, and MPEG-21, which are formally known as ISO/IEC-11172, ISO/IEC-13818, ISO/IEC-14496, ISO/IEC-15938, and ISO/IEC 21000, respectively.

multimedia

A media presentation that contains some combination of text, graphics, sound, video, and animation. Most personal computers can now display multimedia content, and the source is typically a CD because the files are large and require considerable storage capacity. The MPEG-4 format enables broadcasters to deliver multimedia presentations over the Internet.

- N** **noise**
Random flaws that occur in audio or video content.
- P** **pixel**
A picture element, which is one point—or the smallest unit—in a graphic image.
- profile**
A subset of the MPEG-4 standard that enables an MPEG-4 file to be as complex as is necessary for a given software application. Vendors can use profiles to implement only the part of the MPEG-4 standard that they need, knowing that their files will be compatible with products from other vendors.
- Q** **QCIF (pronounced “q siif”)**
Quarter Common Intermediate Format. A video display size (1/4 CIF = 144 x 176 pixels) well suited to lower bit rates.
- R** **RVLC**
Reversible Variable Length Coding.
- S** **scene description**
A description of the media in a streamed presentation and how the content is to be reconstructed. The technical term for this is BIFS (binary format for scenes).
- set-top box**
A device that connects a TV to the Internet so that Internet information can be displayed on the TV screen. This term also is used to refer to a cable or satellite television receiver.
- SIF**
Standard Image Format. For video, the SIF NTSC is 352 x 240 pixels, and the SIF PAL is 352 x 288 pixels.
- smoother**
An algorithm that Helix Mobile Producer uses to regulate bit rates in streaming media and prevent unexpected peaks or bursts in the data streams. The smoother prevents media encoders from generating too much data or overly complex data.

stream

The encoded media or scene description data in an MPEG-4 file. A media stream consists of audio, video, or other multimedia content that is transmitted across a computer network in a streaming or continuous manner.

streaming

The delivery of a media stream from a steaming server to a media player (such as RealOne Player) as a steady, continuous flow of audio, video, or other data. The player reconstructs each media scene by using information stored in the scene description.

U URL

Uniform Resource Locator. A standard, worldwide protocol for naming documents or sites (such as Web pages) on the Internet.

V VBR

Variable bit rate.

VTR

Videotape recorder.

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