GENESYS
AN ALCATEL-LUCENT COMPANY

Framework 7.6

Stream Manager

Deployment Guide

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Table of Contents

List of Procedures

Preface

Welcome to the *Framework 7.6 Stream Manager Deployment Guide*. This guide introduces you to the concepts, terminology, and procedures relevant to Stream Manager 7.6.

Note: This guide is valid only for the 7.6 release of this product. For versions of this guide created for other releases, please visit the Genesys Technical Support website, or request the Documentation Library DVD, which you can order by e-mail from Genesys Order Management at `orderman@genesyslab.com`.

This chapter contains the following sections:

Intended Audience

This guide is primarily intended for system administrators who will be installing and operating Stream Manager 7.6. This guide assumes that you have a basic understanding in the following areas:

- Computer-telephony integration concepts, processes, terminology, and applications.
- Network design and operation.
- Familiarity with your own network configurations.

You should also be familiar with Genesys Framework architecture and functionality.

Chapter Summaries

The *Stream Manager 7.6 Deployment Guide* provides information on configuring, installing, and starting Stream Manager 7.6. To help you locate information, the guide begins with a Table of Contents and ends with an Index. In addition to this opening chapter, this guide contains these chapters:

- Chapter 1, "Overview," on page 15, defines Stream Manager and provides information on changes since the 7.5 release.
- Chapter 2, "Deployment Planning," on page 17, provides information when preparing to install Stream Manager in your environment.
- Chapter 3, "Quick Start Deployment of Stream Manager," on page 23, provides basic configuration information for using the Stream Manager Wizard to set Stream Manager.
- Chapter 4, "Configuration and Installation of Stream Manager," on page 29, further describes the procedures for manually deploying Stream Manager, using the Configuration Layer.
- Chapter 5, "Supported Functionality in Stream Manager," on page 39 provides information about Stream Manager abilities.
- Chapter 6, "Stream Manager Configuration Options," on page 57 describes how to configure Stream Manager for your environment.

Document Conventions

This document uses certain stylistic and typographical conventions—introduced here—that serve as shorthands for particular kinds of information.

Document Version Number

A version number appears at the bottom of the inside front cover of this document. Version numbers change as new information is added to this document. Here is a sample version number:

75fr_sm_06-2008_v7.6.001.00

You will need this number when you are talking with Genesys Technical Support about this product.

Type Styles

Italic

In this document, italic is used for emphasis, for documents' titles, for definitions of (or first references to) unfamiliar terms, and for mathematical variables.

Examples:

- Please consult the *Genesys 7 Migration Guide* for more information.
- *A customary and usual practice* is one that is widely accepted and used within a particular industry or profession.
- Do *not* use this value for this option.
- The formula, $x + 1 = 7$ where x stands for . . .

Monospace Font

A monospace font, which looks like `teletype or typewriter text`, is used for all programming identifiers and GUI elements.

This convention includes the *names* of directories, files, folders, configuration objects, paths, scripts, dialog boxes, options, fields, text and list boxes, operational modes, all buttons (including radio buttons), check boxes, commands, tabs, CTI events, and error messages; the values of options; logical arguments and command syntax; and code samples.

Examples:

- Select the `Show variables on screen` check box.
- Click the `Summation` button.
- In the `Properties` dialog box, enter the value for the host server in your environment.
- In the `Operand` text box, enter your formula.
- Click `OK` to exit the `Properties` dialog box.
- The following table presents the complete set of error messages T-Server distributes in `EventError` events.
- If you select true for the `inbound-bsns-calls` option, all established inbound calls on a local agent are considered business calls.

Monospace is also used for any text that users must manually enter during a configuration or installation procedure, or on a command line:

Example:

- Enter `exit` on the command line.

Screen Captures Used in This Document

Screen captures from the product GUI (graphical user interface), as used in this document, may sometimes contain a minor spelling, capitalization, or grammatical error. The text accompanying and explaining the screen captures corrects such errors *except* when such a correction would prevent you from installing, configuring, or successfully using the product. For example, if the name of an option contains a usage error, the name would be presented exactly as it appears in the product GUI; the error would not be corrected in any accompanying text.

Square Brackets

Square brackets indicate that a particular parameter or value is optional within a logical argument, a command, or some programming syntax. That is, the parameter's or value's presence is not required to resolve the argument, command, or block of code. The user decides whether to include this optional information. Here is a sample:

```
smcp_server -host [/flags]
```

Angle Brackets

Angle brackets indicate a placeholder for a value that the user must specify. This might be a DN or port number specific to your enterprise. Here is a sample:

```
smcp_server -host <confighost>
```

Related Resources

Consult these additional resources as necessary:

- The *Framework 7.6 Deployment Guide*, which will help you configure, install, start, and stop Framework components.
- The *Framework 7.6 Configuration Options Reference Manual*, which will provide you with descriptions of configuration options for other Framework components.
- The *Framework 7.6 Configuration Manager Help*, which will help you use Configuration Manager.
- The *Genesys 7 Migration Guide*, also on the Genesys Documentation Library DVD, which contains a documented migration strategy from Genesys product releases 5.x and later to all Genesys 7.x releases. Contact Genesys Technical Support for additional information.
- The *Genesys 7 Events and Models Reference Manual*, which contains the T-Library API, information on `TEvents`, and an extensive collection of call models.
- The *Genesys Technical Publications Glossary,* which ships on the Genesys Documentation Library DVD and which provides a comprehensive list of the Genesys and CTI terminology and acronyms used in this document.
- The Release Notes and Product Advisories for this product, which are available on the Genesys Technical Support website at:

http://genesyslab.com/support.

Information on supported hardware and third-party software is available on the Genesys Technical Support website in the following documents:

- *Genesys 7 Supported Operating Systems and Databases*
- *Genesys 7 Supported Media Interfaces*

Genesys product documentation is available on the:

- Genesys Technical Support website at http://genesyslab.com/support.
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Chapter

1 Overview

This chapter provides an overview of Stream Manager 7.6. It includes:

Introduction

Stream Manager 7.6 is a media server that generates and processes media streams in Real-time Transport Protocol (RTP) format, providing such functionalities as the following:

- Playing announcements
- Recording streams into files
- Making conference calls
- Silent voice monitoring
- Whisper coaching

New in This Release

The following new features are introduced in release 7.6 of Stream Manager:

- The off-line transcoding utility in Stream Manager, `smzip`, can now save audio file conversions as `.wav` files. See "Saving the Conversion as a Single WAVE File" on page 45 for more information.
- Stream Manager can now reject new dialogs based on a configured load threshold. See "Load Control in Stream Manager" on page 21 for more information.
- Stream Manager can now send two types of DTMF tones—digitized tones and named telephone events (RFC 2833). See "DTMF Generation in Release 7.6" on page 47 for more information.

See “Changes from 7.5 to 7.6” on page 78 for information regarding Stream Manager release 7.6 option changes.

Chapter

2 Deployment Planning

This chapter provides information that will help you plan your deployment of Stream Manager. It includes these sections:

Overview

This chapter is written for system administrators, contact center operations heads, and developers who are planning to deploy Genesys Stream Manager. Much of the information you gather now for planning purposes will also be useful when you later configure and install Stream Manager.

Deploying Stream Manager is similar in many ways to deploying other components of the Genesys Framework, with the significant exception that the voice signal is carried over the data network. This has serious implications for network planning and server sizing. The primary focus of this chapter is to highlight the major planning and resource concerns you face in rolling out Stream Manager and to explain how Stream Manager overlaps with the underlying data network. However, this chapter is not intended to be an exhaustive guide to network planning.

Because most users are probably deploying Stream Manager in conjunction with a previously deployed Genesys Framework, they can refer to the *Framework 7.6 Deployment Guide* for help on most points. The present document describes only those areas of deployment planning in which Stream Manager differs significantly from other Framework components. It does not discuss the following topics that can be found in the Framework documentation: Configuration Layer, Management Layer, Services Layer, solution availability, and security considerations.

Network Considerations

The performance of Stream Manager is directly linked to that of the underlying data network. It is essential that you perform a proper network audit to ensure that the data network has been properly sized and tuned for real-time (voice) packet transport. This section looks at the factors that affect overall performance of an IP-based configuration, and provides some general rules to follow when deploying Stream Manager.

Voice Quality

The following factors have a direct impact on voice quality:

- Network latency—overall network delay.
- Packet loss—voice packets that are dropped for various reasons (physical media error, time-out due to network congestion, and so on).
- Packet jitter—variation in voice packet arrival times. For example, in a system in which packets are emitted at 20-millisecond (msec) intervals, some packets actually arrive at intervals ranging from 0 to 32 msec.

To minimize network latency and ensure acceptable voice quality, you need to tune the network to prioritize real-time voice packets. There are various available schemes for prioritizing voice packets, depending on the IP router vendor. As a general rule, end-to-end network latency should not exceed 250 msec.

Packet loss is a function of several factors, including network bandwidth. As a general rule, maximum sustained packet loss should not exceed 5 percent.

You can minimize packet jitter by using a jitter buffer at the endpoint device. As a general rule, you should set the buffer size to the maximum anticipated deviation from the typical interpacket emission time.

Other factors that influence voice quality include:

- Packet misordering—packets arrive in the wrong order (similar to packet loss).
- Codec used—codecs that do not compress the audio signal produce better voice quality but use greater bandwidth.
- Silence suppression—this can save bandwidth but may also impact voice quality.

Bandwidth Requirements

Determining the bandwidth requirements for the underlying data network is another critical step in achieving proper performance and voice quality (bandwidth requirements for a video connection are much higher). Genesys recommends that you verify network performance and acceptable voice quality by conducting performance tests and measurements in a lab environment prior to production rollout.

For an IP/Ethernet network, factors that affect bandwidth requirements are:

- Codec utilized.
- Protocol headers.

Based on Genesys' experience, a full duplex voice conversation using the G.723.1 codec requires approximately 20 Kbps (kilobits per second). When estimating actual network bandwidth needs, you should also consider such factors as network efficiency and utilization.

Remote Agent Topology

Remote agent capabilities range from a single remote agent, to a group of remote agents in a branch office environment. The distributed nature of branch office or remote agent architectures adds to the complexity of network sizing and tuning. You should consider the following factors in planning Stream Manager deployment.

Bandwidth and Network Tuning

Just as in local network deployment of a VoIP-based system, you should, if at all possible, allot proper bandwidth for voice communication and tune the underlying network for real-time media. Remote agents using a dial-up connection will require greater bandwidth because of the extra network overhead (at least 33 Kbps, with 56 Kbps recommended). This assumes use of the G.723.1 codec, although some dial-up connections may accommodate G.729. A Digital Subscriber Line (DSL) connection is a better alternative than a dial-up connection. Choice of remote access is important to avoid sending voice over an unmanaged data network, such as the public Internet, where voice quality cannot be guaranteed.

For a branch office, network bandwidth requirements depend on the number of agents. Again, WAN connectivity to the corporate LAN must be tunable for real-time voice communications. You need to ensure that service-level agreements from your Virtual Private Network provider detail such requirements. As noted earlier, end-to-end network latency should not exceed 250 milliseconds.

Network Locations

Stream Manager performance, and the network configuration required to optimize performance, depends primarily on two factors—the scenario implemented, and the codecs used.

Low-processing Scenarios

Low-processing scenarios consist of cases where encoding/decoding the media stream is either not required, or the operation uses a trivial codec. These scenarios include Announcement and IVR Service without transcoding (including music-on-hold and Recording Announcement), as well as Conference Service with the G.711 codec. For these scenarios, Stream Manager performance is limited primarily by how fast the RTP packets can be transmitted, which is determined by a combination of the operating system (OS) kernel, the network driver, and Stream Manager itself. Another less significant factor is the total network traffic.

Since both transmission speed and network traffic depend on codec selection, Stream Manager performance in these scenarios varies depending on which codec is selected, and what packet size the codec uses. With a typical computer configuration and 20-msec packet size, Stream Manager can handle about 320 simultaneous media streams. Increasing packet size (by using the G.723.1 or MS-GSM codec, or by increasing the packet-size option for other codecs) will improve the performance, but may negatively affect the voice quality on end-points that have smaller jitter buffers.

High-processing Scenarios

High-processing scenarios require decoding and/or encoding of the media stream using a non-trivial codec. These scenarios include all types of Conference Service using codecs other than G.711 (regular conference, Silent Voice Monitoring, Whisper Coaching, and implicit conference in case of Manual Call Recording), as well as Announcement and IVR Service with transcoding and Call Recording in *mixed* mode. For these scenarios, Stream Manager performance is limited primarily by the amount of CPU power available. Table 1 shows the number of participants that Stream Manager can handle per codec, on a typically configured computer.

Table 1: Number of Participants Per Codec

Codec	Number of Participants
G.711	320 parties (included as a reference point)
G.729 on Intel platform and GSM	100 parties
G.729 on other platforms	40 parties
G.723.1	30 parties

PCAP Recording

Call Recording using the `pcap` mode is the most efficient scenario, since in this case Stream Manager does not have to perform any processing and does not need to time the outgoing packets—it just retransmits the RTP stream coming from one call participant to another, saving a copy into the file. With a typical computer configuration, disk operations are fast enough that they do not affect performance, enabling Stream Manager to handle approximately 500 call participants at one time (that is, 250 recorded calls).

Load Control in Stream Manager

Starting in release 7.6, load sharing is implemented to decrease the number of rejected scenarios that do not get processed. In an environment with distributed Stream Managers, if a dialog is rejected solely because the rejecting Stream Manager is overloaded, that dialog is then retried on another Stream Manager.

In addition, also starting in release 7.6, you can configure a threshold beyond which Stream Manager will reject new dialogs. This functionality is implemented using the configuration option `sip-load-threshold` (page 70). You configure this threshold for each Stream Manager in your network, further optimizing bandwidth usage.

Traffic Generated by Stream Manager

Table 2 provides basic data on link traffic between Stream Manager and Framework components. This data may further help you determine the components' optimal location on the network. The terms *gateway* and *desktop client application* here refer to third-party SIP- or H.323-compliant components.

Table 2: Traffic Generated by Stream Manager

Primary Data Types	Average Message Length	Number of Messages Per Transaction	Elements Determining Total Message	Total Traffic Volume	Timeliness of Message Delivery
Stream Manager–Desktop client application, Stream Manager–Gateway					
Real-time media (voice packets)	24–252 bytes, depending on codec	Variable	Voice packets	Very high	Critical

Table 3 on page 22 lists estimated per channel values for Stream Manager traffic consisting of 10 full-duplex conversations. Actual values will depend on the details of the configuration in use.

Table 3: Stream Manager Traffic Volume Per Channel

Codec	Volume per channel (both directions)
G.711	20 Kilobytes/sec (Kb/sec), 100 packets/sec
G.723.1	1.8 Kb/sec, 67 packets/sec
G.729	2.5 Kb/sec, 100 packets/sec

Chapter

3 Quick Start Deployment of Stream Manager

This chapter provides a simple example of configuring and installing Genesys Stream Manager in a Windows 2000/2003 environment. It includes:

Introduction

There are two ways to perform a first-time install of Stream Manager:

- Use the wizards that are packaged with Stream Manager.
- Use the Framework 7.6 Configuration Manager to create and configure the Stream Manager application before the installation process.

Genesys recommends that you begin your installation using the wizards, which provide a complete basic configuration. You can then add to or alter that configuration using Configuration Manager. If you are upgrading an existing configuration rather than performing a first-time install, you should use Configuration Manager.

This chapter first provides some basic information on starting and using the wizards. Once you launch the wizards, they are largely self-explanatory.

The rest of the chapter describes configuring and installing Stream Manager using Configuration Manager. This chapter assumes that at least the following Genesys Framework components are already installed and operational: DB Server (dedicated to the configuration database), Configuration Server, and Configuration Manager.

Quick Start Installation

Configuration Wizards facilitate component deployment.

Procedure: Configuring and installing Wizard Manager using Stream Manager Wizard

Start of procedure

1. Install and launch Genesys Wizard Manager. Refer to the *Framework 7.6 Deployment Guide* for detailed instructions for deploying the Wizard Manager.

   If you have already deployed the Configuration Layer, an instance of Wizard Manager already exists on your system.

   If you have not yet deployed the Configuration Layer, Wizard Manager suggests that you first set up the Configuration Layer and the Management Layer (if required).

2. Install the Stream Manager Wizard from the Stream Manager CD. This will install the necessary files so that the Stream Manager Wizard can be started from within the SIP Server Wizard or the T-Server for Cisco CallManager Wizard.

   Note: You should install the SIP Server Wizard or the T-Server for Cisco CallManager Wizard from their product CD before starting any wizard process.

3. Run the SIP Server Wizard or the T-Server for Cisco CallManager Wizard. The Stream Manager Wizard can be started from within either Wizard.

   Note: The `Switch` type in Configuration Manager should have a value of `SIP Switch or T-Server for Cisco CallManager` in order to begin the SIP Server Wizard or the T-Server for Cisco CallManager Wizard.

4. Run the Stream Manager Wizard and follow the instructions to configure the Stream Manager application and to prepare the installation package.

5. Copy the Stream Manager installation package that was prepared by the wizard to the host specified during the wizard process. Run the installation package from this location.

End of procedure

Basic Configuration

Procedure:
Quick Start—Configuring a Stream Manager Application object

Purpose: To set up a simple configuration in a single-tenant environment.

Prerequisites

- Configuration Manager is installed and running.

Notes

- For each configuration field, the tables in this section provide the value used in this example configuration, plus a description of what should go in the field so that you can tailor the example to your own environment.
- Do not change any field not mentioned in this Quick Start

Start of procedure

1. Open Configuration Manager.
2. If a `Host` object does not already exist for the computer on which this Stream Manager will be installed, create one as follows:
   a. Select the `Hosts` folder.
   b. Select `File > New > Host`. The `Properties` dialog box for the new `Host` object opens.
   c. Enter the information given in Table 4.

Table 4: Host Properties

Tab	Field	Example Value	Description
General	Name	Genesys_Server	Name of the computer running the Stream Manager applications[a]
	IP Address	198.168.8.1	IP address of the server[a]
	OS Information	Windows 2000	OS type and version running on the server

a. The name and IP address must be valid for your specific data network environment.

   d. Click `OK` to save the configured `Host` object.

3. In Configuration Manager, right-click the `Environment > Applications` folder and select `New > Application`, which opens the `Browse` dialog box with the available Application Templates.

   If a Stream Manager template file is not listed, do one of the following, and repeat this step:

   - Import the `VOIP_SM_<current-version>.apd` file from the product CD as follows:
     a. Right-click the `Environment > Application Templates` folder and select `Import Application Template`.
     b. In the `Open` dialog box, navigate to the product CD, select the template file, and click `Open`.
     c. (Optional) Enter a descriptive name for the template.
     d. Click `OK` to import the template.
   - Create a new template. Refer to the *Framework 7.6 Deployment Guide* for instructions regarding how to create a new template.

4. In the `Browse` dialog box, select the Stream Manager template file. The `Properties` dialog box opens for the new Stream Manager `Application` object.
5. Enter the information given in Table 5.

Table 5: Stream Manager Application Object Properties

Tab	Field	Example Value	Description
General	Name	SM_750	A descriptive name.
	Tenant	Resource	In a multi-tenant environment, the name of the Tenant. Stream Manager can be associated with only one Tenant. The Tenant must be the same as that specified for the T-Server that uses this Stream Manager application.
Server Info	Host	Genesys_Server	The host configured in Step 2.
	Ports	0	Any value (Stream Manager does not actually use this attribute, but it must be given a value.)
Start Info	Working Directory	c:\gcti\Stream Manager	Path name to the directory or folder where the Stream Manager application is installed.
	Command Line	sm.exe	Name of the Stream Manager executable file.

Table 5: Stream Manager Application Object Properties (Continued)

Tab	Field	Example Value	Description
Start Info (cont.)	Command-line Arguments	-host Genesys_Server -port 1111 -app “SM_750”	Host name and port number of Configuration Server, name of the Stream Manager application.
	Timeout (Startup, Shutdown)	120	The time interval, in seconds, during which the application is expected to start or shut down.

6. Click `OK` to save the configured Stream Manager `Application` object.

End of procedure

Next Steps

See “Starting and Stopping Stream Manager” on page 35 for more information about how to start and stop Stream Manager.

Chapter

4 Configuration and Installation of Stream Manager

This chapter describes how to deploy, configure, and install Stream Manager. This chapter includes:

Introduction

This chapter describes how to use Configuration Manager to deploy Stream Manager in a basic single-site configuration.

Note: This chapter assumes some familiarity with the Configuration Layer components of the Genesys Framework. Refer to the *Framework 7.6 Deployment Guide* and *Configuration Manager 7.6 Help* for information on deploying and using Framework components.

Basic Configuration Using Configuration Manager

Procedure:
Configuring a Stream Manager Application object

Prerequisites

- Configuration Manager is installed and running.

Notes

This chapter concentrates on fields in which you need to enter values that are specific to Stream Manager. Some other considerations are:

- Some fields and some entire `Application` objects are similar or identical to fields and objects in the conventional Genesys Framework, and should be configured in the same way.
- As a rule, if a field already has a default value, you can leave it unchanged. This chapter identifies the few cases where you may want to change a default setting.
- Some fields appear only if you are operating in a release 6.0 or later environment. This chapter identifies such fields so that users of earlier releases can skip the discussion.
- Most of the `Application` objects you create have a check box called `State Enabled` at the bottom of the `General` tab in the `Properties` window. Be sure that every application has a check mark in the box (see *Configuration Manager 7.6 Help* for possible reasons to leave the box empty).

Start of procedure

1. Open Configuration Manager.
2. If a `Host` object does not already exist for the computer on which this Stream Manager will be installed, create one as follows:
   a. Select the `Hosts` folder.
   b. Select `File > New > Host`. The `Properties` dialog box for the new `Host` object opens.
   c. Enter the following information on the `General` tab:
      - `Name`—must be unique to the Configuration Layer environment and valid within your data network configuration. See the Configuration Manager Help file for suggestions on name format.
      - `IP Address`—optional.

- `LCA Port`—enter `0` (zero) unless you are using the Management Layer (see "Configuring for Use with the Management Layer" on page 33).

d. Click `OK` to save the configured `Host` object.

3. In Configuration Manager, right-click the `Environment > Applications` folder and select `New > Application`, which opens the `Browse` dialog box with the available Application Templates.

If a Stream Manager template file is not listed, do one of the following, and repeat this step:

- Import the `VOIP_SM_<current-version>.apd` file from the product CD as follows:
  a. Right-click the `Environment > Application Templates` folder and select `Import Application Template`.
  b. In the `Open` dialog box, navigate to the product CD, select the template file, and click `Open`.
  c. (Optional) Enter a descriptive name for the template.
  d. Click `OK` to import the template.
- Create a new template. Refer to the *Framework 7.6 Deployment Guide* for instructions regarding how to create a new template.

4. In the `Browse` dialog box, select the Stream Manager template file. The `Properties` dialog box opens for the new Stream Manager `Application` object.

5. Enter the information given in Table 6.

Table 6: Stream Manager Application Settings

Tab	Field	Description
General	Name	Use any descriptive name.
	Tenant	In a multi-tenant environment, enter the name of the Tenant. Stream Manager can be associated with only one Tenant. The Tenant must be the same as that specified for the T-Server that uses this Stream Manager application.

Table 6: Stream Manager Application Settings (Continued)

Tab	Field	Description
Server Info	Host	Select the Host name configured in Step 2.
	Ports	0 (zero); Stream Manager does not use this attribute, but it must be given a value. If your Configuration Server does not accept the value 0 (zero), enter any other value.
	Backup Server (6.0 and later)	None.
	Reconnect Timeout	Set as for conventional T-Server.
Start Info	Working Directory Command Line Command-Line Arguments	Enter the placeholder character . (a period) in the first two fields, and leave the third blank. These fields will be updated automatically, if the installation package can connect to Configuration Server.
	Timeout (Startup, Shutdown)	60 recommended for both.
Connections		None.

6. Click `OK` to save the configured Stream Manager `Application` object.

End of procedure

Installing Stream Manager

On the product CD, the `solution_specific\voip_sm` directory includes subdirectories for each operating system that contains the necessary installation files.

Procedure: Installing Stream Manager on UNIX

Prerequisites

- A Stream Manager `Application` object exists.

Start of procedure

1. On the product CD, navigate to the `solution_specific\voip_sm` folder, and open the installation subdirectory for your environment.

> **Note:** Do not include spaces in the destination directory name when using a UNIX operating system.

2. Execute the installation script by typing `install.sh` on the command line.

End of procedure

Procedure: Installing Stream Manager on Windows 2000/2003

Prerequisites

- A Stream Manager `Application` object exists.

Start of procedure

1. On the product CD, navigate to the `solution_specific\voip_sm` folder, and open the subdirectory `windows`.
2. Double-click `Setup.exe` to start the InstallShield installation wizard.
3. Follow the InstallShield wizard instructions to install Stream Manager. InstallShield creates a batch file in the folder of the component it installs.
4. Open the batch file and ensure that it includes the parameter `-app <appname>`, where `<appname>` is the name of the application object you created in Configuration Manager.

End of procedure

Configuring for Use with the Management Layer

The Management Layer, part of Genesys Framework 7.6, centralizes many functions (such as starting and stopping, status monitoring, alarm processing, and processing of log records) for all components of the Genesys Solution. Detailed information about the Management Layer is available in the *Framework 7.6 Management Layer User's Guide.*

Note: Stream Manager always functions in primary (non-redundant) mode unless instructed otherwise by SIP Server.

Procedure:
Configuring Stream Manager to work with the Management Layer

Purpose: To enable Management Layer to monitor, start, and stop Stream Manager.

Prerequisites

- A Stream Manager `Application` object already exists, or is in the process of being created (see the procedure Configuring a Stream Manager Application object, page 30).

Start of procedure

1. Start Configuration Manager, if it is not already started.
2. Open the `Properties` window of the Stream Manager `Application` object, if it is not already open.
3. Enter the information as described in Table 7.

Table 7: Configuring the Stream Manager Application Object to work with the Management Layer

Object	Tab	Field	Values to Enter
Host	General	LCA Port	Number of the port on which LCA (Local Control Agent) is running
Stream Manager	Connections	Server: Add	Add the name of the Message Server
	Start Info	Working Directory	Path name to the directory or folder holding the Stream Manager application
		Command Line	sm.exe
		Command-line Arguments	-host <hostname> -port <portnumber> -app <appname>, where <hostname> and <portnumber> are the host and port of Configuration Server and <appname> is the name of the Stream Manager application
		Timeout (Startup, Shutdown)	120 recommended

Table 7: Configuring the Stream Manager Application Object to work with the Management Layer (Continued)

Object	Tab	Field	Values to Enter
Stream Manager (cont.)	Start Info (cont.)	Redundancy Type	Unspecified
		Auto-Restart	Put a check mark in the box if you want the Management Layer to automatically restart the application.

4. Click `OK` to save the configuration changes.

End of procedure

Starting and Stopping Stream Manager

Starting Stream Manager

You can start Stream Manager by using the Solution Control Interface or by starting it manually.

Procedure:
Starting Stream Manager using Solution Control Interface

Prerequisites

- Stream Manager is installed.
- Solution Control Interface (SCI) is installed.

Start of procedure

1. If it not already running, start SCI and log in.
2. Go to the `Applications` view.
3. Select the Stream Manager `Application` object in the `List` pane.
4. Click `Start` or select `Start` from the shortcut menu. The application's status changes from `Stopped` to `Started`.

End of procedure

Procedure: Starting Stream Manager manually

Prerequisites

- The Configuration Layer components DB Server and Configuration Server are installed and running.

Required Command-Line Parameters

The executable file must contain the three parameters listed in Table 8 in the command line.

Table 8: Required Command-Line Parameters

Parameter	Default	Possible Values	Description
-app	none	any text string	Name of the application as configured in the Configuration Database
-host	none	any host name	Configuration Server host
-port	none	any integer	Configuration Server port

Start of procedure

1. To start Stream Manager on UNIX, go to the directory where Stream Manager is installed and do one of the following:
   - To use only the required command-line parameters (listed in Table 8 on page 36), type the following at the command line:
     `sh run.sh`
   - To specify the command line yourself, or to use additional command-line parameters, type the following at the command line:
     `sm -port <Configuration Server host> -app <Stream Manager Application> -host <Configuration Server host>`
2. To start Stream Manager on Windows, do one of the following:
   - Start the `Genesys Stream Manager` service from the `Services` menu.
   - Select the shortcut `Start Stream Manager` from the `Start` menu.
   - Go to the directory where Stream Manager is installed and click the `startServer.bat` batch file.

End of procedure

Stopping Stream Manager

Procedure: Stopping Stream Manager using Solution Control Interface

Prerequisites

- Stream Manager is installed and running.
- Solution Control Interface (SCI) is installed.

Start of procedure

1. If it not already running, start SCI and log in.
2. Go to the `Applications` view.
3. Select the Stream Manager `Application` object in the `List` pane.
4. Click `Stop` or select `Stop` from the shortcut menu. The application's status changes from `Started` to `Stopped`.

End of procedure

Stopping Manually

Stream Manager can be stopped by any means that your operating system supports.

Graceful Shutdown

Stream Manager supports graceful shutdown when all of the following conditions apply:

- The Stream Manager `Application` object is marked as `Disabled` in Configuration Manager.
- The reaction `Shutdown application` is configured in Solution Control Interface.
- The Voice over IP Service DN is marked as `Disabled` and cannot be used by SIP Server.

In this scenario, when the last active RTP leg is completed, the log event message `53053 No more active dialogs - safe to shutdown` is generated, causing the `Shutdown application` reaction to execute.

Chapter

5 Supported Functionality in Stream Manager

This chapter describes the voice treatments and audio codecs supported by Stream Manager. It includes the following sections:

Support for Media Files

Stream Manager can handle multiple audio and video codecs. Stream Manager supports `*.wav` and `*.au` media files for audio playback, based on the value of the `audio-file-format` option (page 72). Stream Manager also supports binary files for video playback encoded by either the H.261, H.263, or H.264 codec types. Binary video files can contain pictures with CIF or QCIF sizes and arbitrary frame rates (up to a maximum of 30 frames per second) when encoded using either the H.261 or H.263 codec. (Exact picture sizes and frame rates are adjustable using the appropriate configuration options found in Chapter 6, "Stream Manager Configuration Options," on page 57).

Note: Starting with release 7.5, G.723.1 is a fully-supported codec so it can be used for transcoding and conference call services. However, because of performance and license issues, this codec and the G.729 codec are disabled by default. It can be enabled using the `sip-conf-codecs` option (page 63). The default list of values for this option and for the `sip-annc-codecs` option (page 62) allows for lower resource-intensive codecs to be used first.

Supported Codecs

Table 9 shows the audio codecs that are supported by Stream Manager, and their associated file names.

Table 9: Supported Audio Codecs and Audio File Names

Audio Codec Name	Name in SDP	File Name
G.711 mu-law	pcmu/8000	<name>_pcmu.<ext> or <name>_mulaw.<ext>
G.711 A-law	pcma/8000	<name>_pcma.<ext> or <name>_alaw.<ext>
G.723.1	g723/8000	<name>_g723.<ext> or <name>_g7231.<ext>
G.729/729a	g729/8000	<name>_g729.<ext> or <name>_g729a.<ext>
GSM Full Rate	gsm/8000	<name>_gsm.<ext> or <name>_gsmFR.<ext>
Microsoft GSM	-	<name>_msgsm.<ext> or <name>_gsmF.<ext>

Notes:

- The media file may be encoded with either the G.729 or G.729a codec. The format of the encoded data is exactly the same, although the coding/decoding algorithms for each codecs is different.
- If you are using Stream Manager with T-Server Cisco CallManager, the codec type of the music file in use must be a value specified in the `audio-codec` T-Server Cisco CallManager option. See the *Framework 7.6 T-Server Cisco Call Manager Deployment Guide* for more information.

Table 10 shows the video codecs supported by Stream Manager and their file naming conventions.

Table 10: Supported Video Codecs and Video File Name Conventions

Video Codec Name	Name in SDP	Stand-alone Name
H.261	H261/90000	<name>_h261_XCIF=mpi
H.263	H263/90000	<name>_h263_XCIF=mpi
H.264	H264/90000	<name>_h261_profile-level-id=mpi

Notes:

- For maximum compatibility, it is recommended that you prepare media for the H.264 codec using the Baseline profile with a `profile-level-id` value of `42e0xx`.
- For video codecs, the file name includes the negotiated size of the video (such as CIF and QCIF).

Support for Push Video

When using SIP Server, Stream Manager supports push video functionality. See the “Supported Functionality in SIP Server” chapter in the *Framework 7.6 SIP Server Deployment Guide* for more information about how to enable and manipulate this functionality.

Stream Manager 7.6 can play files that contain a raw video stream encoded with H.261, H.263, or H.264 video codecs only.

Note: Genesys does not provide any utility for converting either uncompressed video or compressed AVI files into these formats. You must use either commercial or open-source third-party converters for this purpose.

Audio Track Extraction

Extract the audio track from an AVI or MPEG video file by using the `mplayer` application with the following command line:

```
mplayer source_file -ao pcm:file=output_file.wav -vc null -vo null
```

Depending on the original file format, the extracted audio track may require further processing to convert it into an 8 KHz 16-bit mono PCM stream.

H.263 Video Preparation

Use the following command to create H.263 raw video with a QCIF=2 setting from any video file format supported by `mplayer`:

```
mencoder source_file -vf scale=176:144 -of rawvideo -ofps 15000/1001
-nosound -o output_file -ovc lavc -lavcopts
vcodec=h263:keyint=10:vpsize=1000:vqmin=3:vbitrate=100
```

Where:

- `vf`—applies a video filter (scale filter re-samples the picture to desired size, not necessary if the original file is already at desired resolution)
- `ofps`—specifies the desired playback speed (in frames per second, base NTSC rate = 30000/1001)
- `vbitrate`—controls the compression (higher bit rate provides better quality, but consumes more network traffic)

H.264 Video Preparation

Use the following command to create H.264 raw video with a 42e00a=2 setting and QCIF picture size:

```
mencoder source_file -vf scale=176:144 -of rawvideo -ofps 15000/1001
-nosound -o output_file -ovc x264 -x264encopts
keyint=10:bframes=0:nob_adapt:nob_pyramid:nocabac:level_idc=10
```

Where:

- `vf`—applies a video filter (same as for H.263)
- `ofps`—specifies the desired playback speed (in frames per second)
- `level_idc`—specifies the H.264 level (multiplied by 10)

Support for Media File Archives

Starting with release 7.5, Stream Manager supports `*.wav` media files packaged in an uncompressed file archive. File archives can be specified as:

- `music/in_queue.zip`
- `music/on_hold.zip`

Stream Manager can also support the `music/QTMF` file structure, which describes the Quad-tone Multi-frequency cadences that can be used in place of other media files.

Note: For backward compatibility, Stream Manager still supports the 7.2 file structure.

Stream Manager 7.6 searches for media files as follows:

1. If the file name specified in the `INVITE` request or the `PLAY` request includes `*.zip`, `*.wav`, or `*.au`, then that file is used for playback. File encoding does not have to match the value of the `sip-annc-codecs` option (page 62), and transcoding will be performed if necessary.

   Note: Stream Manager does not support specifying the full name for video files in the request. Since video playback is usually accompanied by an audio track, the request should either use the base name (resulting in Stream Manager constructing different names for video and audio components based on the codec selection), or the request must refer to a `*.zip` archive that contains both the audio and the video components.

2. If the name specified in the `INVITE` request or the `PLAY` request does not match any existing file, then Stream Manager assumes that the base name is specified and adds a suffix or extension based on the following conditions:
   - If a `*.zip` archive containing the base name exists, then Stream Manager extracts the media for the specified codec. Transcoding does not occur.
   - If the media file exists, Stream Manager adds the appropriate codec suffix or extension. The media file can be recorded in a different codec and used for playback, but the file suffix must match the codec specified in the file.

   Note: Stream Manager does not support a combination using a `*.zip` archive and non-archived media. In this scenario, a `*.zip` archive is used first.

3. If a media file is still not found, and the specified file name starts with `music/`, then Stream Manager attempts to retrieve the tone description from the `music/QTMF` file. If the tone description is found, Stream Manager generates the media stream with proper encoding based on that description.
4. If all previous steps fail, a `404 Not Found` response is sent.

Table 11, and Table 12, provide more information about archiving media files into a single uncompressed `*.zip` archive. Any folder information that is present in the `*.zip` archive is ignored by Stream Manager.

For audio data (Table 11), the file names specified in the archive must be the same as the short codec name that is specified in the SDP and in the Stream Manager options, with the `*.wav` extension:

Table 11: Audio Archive Names

Codec	ZIP Name	File Name
G.711 mu_law 8 KHz	pcmu.wav	<name>_pcmu.<ext> or <name>_mulaw.<ext>
G.711 A-law 8 KHz	pcma.wav	<name>_pcma.<ext> or <name>_alaw.<ext>
G.723.1 MP-MLQ ACELP	g723.wav	<name>_g723.<ext> or <name>_g7231.<ext>
G.729/729a CS-ACELP	g729.wav	<name>_g729.<ext> or <name>_g729a.<ext>
Standard GSM (33 bytes/20 msec)	gsm.wav	<name>_gsm.<ext> or <name>_gsmFR.<ext>
Microsoft GSM (65 bytes/40 msec)	mgsm.wav	<name>_msgsm.<ext> or <name>_gsmF.<ext>

For video data (Table 12), the file names specified in the archive must contain the codec name and format specification:

Table 12: Video Archive Names

Codec	ZIP Name	File Name
H.261	h261_<XCIF>=<mpi>	<name>_h261_<XCIF>=mpi
H.263	h263_<XCIF>=<mpi>	<name>_h263_<XCIF>=mpi
H.264	h264_<profile-level-id>=<mpi>	<name>_h264_<profile-level-id>=<mpi>

Support for Transcoding Audio Files

The `smzip` utility can be used to convert one given audio file and set of prepared video files to a set of media files with different encoding, and save the result as either an archived (`.zip`) file or as a standard WAVE (`.wav`) file. See "Off-line Transcoding" on page 45 for more information.

You can transcode your files in either real-time or offline.

Real-time Transcoding

Stream Manager can automatically converts from one codec to another when the encoding of a file does not match that of the negotiated codec. Starting with release 7.5, Session Description Protocol (SDP) generation has been changed to include all configured codecs and not only those codecs for which the media file is found. However, files that have the proper media available still take precedence. For more information, refer to `sip-annc-transcode`.

Off-line Transcoding

A separate `smzip` utility is provided to convert from one given audio file or set of prepared video files to a set of media files with different encoding. The results can be saved as one of the following:

- A single archived (`.zip`) file. See “Archiving the Conversion” on page 45.
- A single WAVE (.wav) file. See “Saving the Conversion as a Single WAVE File” on page 45.

Archiving the Conversion

To convert files and save the result in a single archived file, run the utility as follows:

```
smzip [ -ac audio_codec_list ] output_zip_name orig_file.wav [
raw_video_file ]
```

Where:

- `audio_codec_list` specifies a comma-separated list of codecs to be included into the archive. You can use the `all` value to include all supported codecs.
- `output_zip_name` specifies the name of the resulting archive file.
- `orig_file.wav` specifies the file to be converted with any of the supported codecs.
- `raw_video_file` specifies the corresponding video stream that is encoded with one of the supported codecs. The file name includes the video format suffix.

By default, only license-free codecs are included. If the option is not specified in the command line, the `pcmu`, `pcma`, `gsm`, and `msgsm` values are used by default.

Note: Starting in release 7.6, you can use other standard tools, such as WinZip, to extract files from an archive created using the `smzip` utility.

Saving the Conversion as a Single WAVE File

To convert files and save the result in a single WAVE file, run the utility as follows:

```
smzip -wav audio_codec output_wav_name orig_file.wav
```

Where:

- `audio_codec` specifies the codec to use for encoding the file, as given in Table 13.

Table 13: Codecs Supported for Converting to WAVE File

Name	Compatibility with Microsoft Media Player
pcmu	Compatible
pcma	Compatible
g723	Requires G.723.1 codec to be installed separately. Available only in commercial packages.
g729	Requires G.729/729a.1 codec to be installed separately. Available only in commercial packages.
gsm	Not compatible
msgsm	Compatible

- `output_wav_name` specifies the name of the resulting WAVE file.
- `orig_file.wav` specifies the file; this file may be either an uncompressed 8 KHz 16-bit PCM (1 channel) file or a file encoded with one of the supported codecs.

Support for Generation of Standard Telephone Tones

Instead of keeping pre-recorded standard telephone tones, Stream Manager 7.6 generates them as needed from the descriptions stored in the `music/QTMF` file, which is stored in the standard Genesys text configuration format.

Two sections must be present in the `music/QTMF` file, as follows:

- `file` section—describes the cadences for each allowed `music/name` parameter from the request. The element name does not include the `music/*` prefix. The value if formed as follows:

  ```
  file-value ::= ( cadence-description-list )
  ```

  ```
  cadence-description-list ::= cadence-description | cadence-description-list , cadence-description
  ```

  ```
  cadence-description ::= tone-name = duration | x repeat-counter = ( cadence-description-list )
  ```

  Where:

  - `tone-name` refers to the element in the `tone` section
  - `duration` is any integer that specifies the duration of that tone in 10 millisecond frames

    - `repeat-counter` is an integer that specifies how many times the sequence in the parenthesis should be repeated

- `tone` section—defines the multi-frequency tones that are used for cadence generation as following:

  ```
  tone-value ::= ( frequencies , amp = amplitude )
  frequencies ::= frequency | frequencies , frequency
  frequency ::= f n = frequency-value-in-Hz
  n ::= 1 | 2 | 3 | 4
  ```

  Where `amplitude` refers to the volume of the generated signal as a percentage of the maximum volume.

Note: The tone definition file included with Stream Manager is for the United States and Canada. For other countries, the definitions need to be changed, or alternative definitions must be supplied as pre-recorded files placed into the `music` folder. However, if a media file exists, Stream Manager will use that file instead.

Support for Dual-Tone Multi-Frequency

Upon receiving a DTMF event in an RTP stream, Stream Manager sends the `INFO` message with the `application/dtmf-relay` payload that contains the played digits and the duration (in milliseconds). For example:

```
Signal= 1
Duration= 160
```

This message is always sent in the context of the existing SIP dialog.

Stream Manager recognizes the `TELEPHONE-EVENT` mime-type as valid for telephony events listed in RFC 2833.

DTMF Generation in Release 7.6

Starting in release 7.6, you can:

- Instruct Stream Manager to send one of two types of DTMF tones: digitized tones, or named telephone events as described in RFC 2833. Selection of one or the other is made by using the configuration option `sip-dtmf-method` (see page 77), which sets the default type.
- Set the default duration of DTMF tones and events by using the configuration option `sip-dtmf-duration` (see page 77). The default duration can be overridden as required.
- Set the delay before starting DTMF generation by using the configuration option `sip-dtmf-delay` (see page 77).

Note: Digitized tones are reliably delivered only when a low-compression codec, such as G.711, is used. Therefore, when using the tone method, Stream Manager gives higher priority to `pcma` and `pcmu` codecs.

Use the following URI (which conforms to RFC 4240) in the `INVITE` message to request DTMF generation to the connection end point:

```
sip:annc@<SM_hostport>;play=dtmf:<digits>[,t<digit_duration>]
[;<annc_parameters>]
```

Where:

- `digits`—Specifies the DTMF sequence to be sent, using the following characters:

`0-9, *, #, A-D`	DTMF digits and special DTMF events
`p`	Pause at half the digits duration

- `digit_duration`—Sets the duration of each DTMF event, in milliseconds; if not specified, the configured default value (configuration option `sip-dtmf-duration`) is used.
- `annc_parameters`—Additional announcement parameters, such as `content-type`, can be specified. The following two announcement parameters are ignored:
  - `repeat`—The DTMF sequence is always generated just once.
  - `record`—DTMF generation cannot be combined with recording.

When Stream Manager receives the request, it will generate the requested sequence as an audio tone (as defined in ITU-T Recommendation Q.23), and as out-of-band RTP packets (as defined by RFC 2833).

Example For example, to request Stream Manager to send the DTMF sequence `*80,6504661410`, where the comma (`,`) designates a pause, using a duration of 100 msec and the default generation method, use the following URI:

```
sip:annc@hport:6050;play=dtmf:*80p6504661410,t100
```

Support for Call Recording

Starting in release 7.5, Stream Manager supports a new method for call recording. However, it still supports the 7.2 method. The following sections describe each method.

Regular Method

Starting in release 7.5, Stream Manager performs non-emergency call recording of a two-way call by converting the call into a two-party conference. Both call participants are re-invited to the conference call by sending the following URL in the `INVITE` messages to Stream Manager:

```
sip:conf = conf-ID @SM-hostport; record = record-URL
```

Where the parameter specifies the file name for recording. It is processed in the same method that the Recording Announcement Service uses. See page 52 for more information about this service.

Note: The parameter values must be identical for both parties.

For this scenario, a two-step process is used for codec negotiation.

First Step In the first step, the SIP Server sends an SDP message from the original conversation with the addition of the `a=inactive` attribute, in order to notify the Stream Manager of the end-point capabilities. Because codec negotiation for each party in the conference is done independently, the Stream Manager needs to know the end-point capabilities in advance, in order to avoid selecting different codecs for different parties (which, although acceptable for a mixed recording mode, will not work for the `pcap` recording mode).

Second Step In the second step of this process, the SIP Server sends a re-`INVITE` request without SDP, for which the Stream Manager responds in the same way that it would for a regular conference (except it uses only those codecs that are supported by both participants).

The `sip-call-record-mode` option controls which of the following files the call recording service produces:

- An audio file with mixed streams
- A file with all packets captured for future processing.

See page 76 for more information about this option.

smmix Utility

The `smmix` utility converts `pcap` file recordings into the `wav` file format that mixes both recorded streams into a single channel. Run the executable file as follows:

```
smmix [ -ac audio_codec ] input_file.cap [output_base_name]
```

Where:

- `input_file.cap`—specifies the full name of the input capture file that was produced in `pcap` mode during call recording.
- `output_base_name`—specifies the output file name. If it is not specified, the utility creates a name by removing the codec suffix and file extension from the input file name, and then creates a file name using the same rules as the Recording Announcement service on page 52.
- `audio_codec`—selects the codec for the output file. The default value is `G.711 mu-Law`.

Note: The `smmix` utility is designed to work with files recorded by Stream Manager only. Any captures produced by other tools may not be processed correctly.

Manual Method (Emergency Recording)

Call or Conference Call recording is implemented in SIP Server by initiating a conference call with a recording device, such as Stream Manager, or with a third-party recording device. A new SIP Server call leg is created for every recording performed by Stream Manager.

Note: Stream Manager can perform on different hosts under different operating systems, but still share a single storage area for media files. In this case, file accessibility with the shared storage for all hosts should be resolved at an Administrative level.

Audible Alert

Stream Manager is able to provide an audible alert when call recording starts. To enable this functionality, configure the `Voice over IP Service` DN in Configuration Manager by adding an alert sound to the `request-uri` option. You can use the following as an example:

```
request-uri = annc@SM;play=music/normal_5sec;repeat=1;record=recording/
call-
```

Note: The play and repeat parameters must be specified before the record parameter, because SIP Server includes the `Call UUID` attribute at the end of the `Request-URI` message.

File Creation

When recording calls, Stream Manager uses the following configuration options:

- `max-record-file-size`—the maximum size of the audio file used for recording.
- `max-record-time`—the maximum recording time, in seconds.
- `max-record-silence`—the maximum allowable amount of time (in seconds) that silence may be detected during a recording. Additional recording parameters should be specified in the `RecordUserAnnouncement` treatment Refer to the *Universal Routing 7.6 Reference Manual* for more information.

File recording ceases when any of the following events occur:

- SIP Server sends a `STOP` command (for example, after entering a key stroke combination that indicates that the caller has finished recording the announcement).
- The `max-record-time` interval has expired.
- The `max-record-silence` or the `max-record-file-size` limits are reached.

SIP Server will send a `STOPPED` notification if the recording is interrupted and issues `EventTreatmentEnd` or `EventTreatmentNotApplied`.

When recording an incoming audio stream, Stream Manager combines the announcement file with the recording file. The URL in the SIP Server `INVITE` request should be created as follows:

```
sip:annc@SM-hostport; record= record-URL [; play= prompt-URL ]
```

The optional `play` parameter may refer to a media file that can contain a pre-recorded "beep" sound or other announcement. The `record-URL` parameter is used to generate the output file name based on the following:

- If the file name specified in the URL includes the correct file extension, then Stream Manager uses this name to record the file without any modifications. If the file name already exists, it will be overwritten.
- If the file name is not specified, then Stream Manager adds a segment number (starting with 1) and the codec-specific suffix or extension

The files are always recorded with the same codec used during the call.

Note: Stream Manager can record the audio portion of a video call, but the recording leg in the conference call must contain the `confrole=monitor` attribute.

Recommendations

Although Stream Manager supports recording with any audio codec, Genesys recommends that you use the G.711 codec only when recording calls, unless there are specific reasons to use another codec. The `sip-record-codecs` option (page 66) must be set to `pcmu` (mu-Law) or `pcma` (A-law). Either value requires significantly less CPU resources for processing, and it also provides the best quality and the best compatibility with other software.

If call recording services are used occasionally, or if the majority of calls use the G.711 codec, then the `mixed` recording mode can be specified in the `sip-call-record-mode` option. See page 76 for more information about this option. This value is recommended only when the G.723 and G.729 codecs are disabled for conference calls and are not listed as values in the `sip-conf-codecs` option. See page 63 for more information about this option.

However, if call recording services are used frequently, and if the G.723, G.729, or GSM codecs are used, then the `sip-call-record-mode` option must be set to `pcap`. The recorded files can then be processed by the *smmix* utility to convert them into the G.711 codec in either mu-Law or A-law format.

Support for Record User Announcement

The call recording function is supported by the `RecordUserAnnouncement` treatment. When SIP Server receives a `RecordUserAnnouncement` request, it sends the message to Stream Manager. In cases where multiple Stream Managers are connected to SIP Server, the `RecordUserAnnouncement` request does not determine which Stream Manager is chosen by SIP Server because SIP Server chooses among all available Stream Managers in a round-robin fashion.

By default, the recorded user's announcement is saved into a `users` folder. The file name specified in the `RecordUserAnnouncement` treatment should match the Configuration Manager `Tenant` name. The format of the recorded file depends on the audio codec chosen during the negotiation procedure. See the `sip-record-codecs` option on page 66 for more information.

Support for SIP Extensions

Stream Manager uses the NETANN (Basic Network Media Services with SIP) Internet Draft (RFC 4240) for the following functionalities:

- Announcement Service (including music-in-queue and music-on-hold for both audio and video calls)
- Basic Conferencing Call support (including Silent Voice Monitoring, Voice Recording, and Whisper Coaching functionality).

Note: Prompt and Collect Service (using VoiceXML scripts) is not supported.

Simple Announcement Service

For a simple announcement service, the URL in the SIP `INVITE` request should be in the following format:

`sip:annc@SM-hostport` ; `play=` `prompt-URL` [; `annc-parameter`]... [; `URI-parameter`]...

Where:

- `SM-hostport`—identifies the Stream Manager location
- `prompt-URL`—identifies what should be played
- `annc-parameter`—an optional parameters
- `URI-parameter`—the SIP Request-URI parameter

prompt-URL for File Announcement

The parameter `prompt-URL` must be constructed in accordance with RFC 2396 (either in a full or abbreviated form). For announcements stored in local files, the `file://` scheme may be used in the URL. The file name use absolute or relative paths, and with or without codec-specific suffix/extension. Stream Manager searches for the file in this order:

1. Using the original name provided in the request (only for single-media announcements).
2. Adding a suffix based on the codecs specified in Table 9 on page 40 and adding a file extension based on the option `audio-file-format` as specified on page 72.

annc-parameter

Stream Manager recognizes only the following parameters:

- `mode-param ::= mode=stream` enables `stream` mode for announcements obtained through the HTTP interface. The specified URL is assumed to correspond to a real-time media streaming device that provides an endless stream. Stream Manager does not cache the received data (except for what is necessary to provide jitter buffer). When a new request comes for the same URL, it will receive media stream when it connects to Stream Manager. and not from the beginning of media file as in case of regular mode.

  Note: This parameter is ignored for local file announcements.

- `repeat-param ::= repeat= ( forever | 1*DIGIT )`—specifies the number of repetitions. The value `repeat=forever` is assumed if this parameter is not specified.

  Note: This is not applicable when in `stream` mode.

- `content-param ::= content-type= MIME-type`—selects the particular codec for announcement. Only one single MIME-type can be specified.

  Note: This parameter cannot be used for announcements containing both audio and video media.

prompt-URL for Announcement through HTTP

If the media is received through an HTTP interface, the `http://` scheme should be used in the URL. If the file name part of the URL already includes a correct file extension, it is passed to the HTTP server without any modification.

Otherwise, Stream Manager adds a codec-specific suffix (found in Table 9 on page 40) and the `.wav` file extension to the specified name.

Recording Announcement Service

Stream Manager provides a non-standard extension that combines announcement and recording services to record an incoming stream. The URL provided in the SIP `INVITE` request should be in the following format:

`sip:annc@SM-hostport; record=record-URL [; play= prompt-URL ]`

In this case, the optional `play` parameter can refer to the media file with a pre-recorded "beep" sound or any other announcement. The `record-URL` parameter is used for generating the output file name as follows:

- If the filename portion of the URL already includes the correct file extension—either `.wav` or `.au`, depending on the configuration of the `audio-file-format` option as specified on page 72 (`.wav` is the default)—the Stream Manager uses this filename for the recording without modifications. If a file with this name already exists, it will be overwritten with the new file.
- If the filename portion of the URL does *not* include the correct file extension already, then the Stream Manager adds a codec-specific suffix/ extension (as it does for playback filenames), as well as a segment number in cases where a file with this name already exists. The complete file name takes the following format:

  `<name>_<codec>.ext`

  Or if segmented:

  `<name>(<n>)_<codec>.ext`

  Segments start at `2` and increase by increments of `1`, for as many segments as necessary to avoid one fragment overwriting another. Segmentation applies even for files recorded by different instances of Stream Manager. For example, if `record=sample` is specified in three requests, and the µ-Law codec is selected, then the Stream Manager instances will record to the following filenames:

  - `sample_pcmu.wav`
  - `sample(2)_pcmu.wav`
  - `sample(3)_pcmu.wav`

Note: Files are always recorded using the same codec used for transmission. Transcoding is not supported.

Basic Conference Calls

For basic conference calls (including Silent Voice Monitoring, Voice Recording, and Whisper Coaching), Stream Manager uses a protocol that is compatible with current conference call standards. In order to establish a conference call, the `INVITE` message is sent to Stream Manager for each participant as a URL:

`sip:conf=UniqueID` @`SM-hostport` [; confrole=`conf-Role`] [; `URI-parameter`]...

Where:

- `UniqueID`—any string that uniquely identifies the conference call. The first `INVITE` message with the previously unknown ID creates the conference call and all subsequent `INVITE` messages with the same ID includes the participants to the conference call.
- `SM-hostport`—the Stream Manager location (as required by RFC 3261).
- `conf-Role`—a non-standard extension that specifies coach or student roles for Whisper Coaching support.
- `URI-parameter`—the SIP Request-URI parameter as described in RFC 3261.

Stream Manager 7.6 also allows the following values for the `confrole` attribute:

`confrole=regular`	Customer call leg receives audio from the mixer (or student call leg in a Whisper Coaching scenario), and video from the agent/student call leg (or from the file in a Push Video scenario).
`confrole=agent` or `confrole=student`	Agent call leg receives audio from the mixer, and video from the regular or customer call leg (or from the file in a Push Video scenario).
`confrole=coach`	Supervisor call leg in a Whisper Coaching scenario receives audio from the mixer, and the same video stream as the agent/student leg.
`confrole=monitor`	Supervisor or recording device call leg in a Silent Call Monitoring scenario receives audio from the mixer, and the same video stream as the agent/student leg.
`confrole=push`	Media playback device call leg does not receive any media; incoming audio stream is pushed to the mixer, and video stream is pushed to a regular or customer call leg.
`confrole=push-all`	Media playback device call leg provides audio to the mixer, and video stream to all call legs in the conference call.

Silent Voice Monitoring

Calls using Silent Voice Monitoring is established with Stream Manager as a regular conference call but with the monitoring call leg muted. The SDP for the muted (monitoring) leg must include a `=recvonly` attribute for this audio stream to indicate that this end point is not going to send any RTP packets. If it does, Stream Manager will ignore these packets.

Barge-in functionality is supported for audio and video calls.

Whisper Coaching

Whisper Coaching functionality is controlled by a non-standard attribute `confrole` in the conference call leg URL. In order to establish a Whisper Coaching session between a customer, an agent, and a supervisor, the following `INVITE` messages is sent with these URLs including the same `ConfID`:

Customer:

`sip:conf=ConfID @SM-hostport [ ; URI-parameter ]...`

Agent:

`sip:conf=ConfID @SM-hostport ; confrole = student [ ; URI-parameter ]...`

Supervisor:

`sip:conf=ConfID @SM-hostport ; confrole = coach [ ; URI-parameter ]...`

In this case, Stream Manager will mix the voice streams so that the agent and the supervisor hear all call parties, but the customer hears only the agent. Starting with release 7.2, the conference call must include only one customer and one agent. However, there are no limitations on the number of supervisor call legs.

Stream Manager Log Events

Starting in release 7.6, log events for Stream Manager are described online in *Framework 7.6 Combined Log Events Help*.

Chapter

6 Stream Manager Configuration Options

This chapter describes the configuration options that are specific to Stream Manager. It includes these sections:

Overview

Setting configuration options in Configuration Manager is the primary means of configuring all Genesys applications.

Stream Manager options are grouped into the following sections:

- `contact` (see page 60)
- `codecs` (see page 61)
- `limits` (see page 68)
- `x-config` (see page 72)

Notes: In release 7.6, Stream Manager configuration options were reorganized into new or different groups. See "Changes from 7.5 to 7.6" on page 78 for details.

The old locations are still recognized by Stream Manager if you are using a version of Configuration Manager prior to release 7.6

Time Values

Some configuration options set time values for timeouts, durations, and delays. The following format allows you to use fractional values and various time units for these settings:

`[[[ hours:] minutes:] seconds][.milliseconds]`

or

`[ hours hr][ minutes min][ seconds sec][ milliseconds msec]`

A time unit value in italics (such as *hours*) must be replaced by an integer value for this time unit. Integer values with no measurement units are supported. When you do not specify any measurement unit, the default value applies.

Support for SIP Server

Enable SIP Server support in Stream Manager by using the following option:

- `sip-port` (page 61)

You may need to update the following options to match the end points configuration and desired operational environment:

- `sip-annc-codecs` (page 62)
- `sip-annc-transcode` (page 62)
- `sip-call-record-mode` (page 76)
- `sip-conf-codecs` (page 63)
- `sip-pcap-codecs` (page 66)
- `sip-record-codecs` (page 66)

Video Files

For video codecs, additional parameters such as picture size and frame rate must be negotiated between the end points. However, since this negotiation occurs before Stream Manager examines the media file, the following options are used to describe existing video files:

- `h261-annc-fmts` (page 72)
- `h263-annc-fmts` (page 72)
- `h264-annc-fmts` (page 73)

gsm or msgsm?

There are currently two methods of packaging the encoded stream in GSM, which are not binary-compatible with each other.

- Standard GSM packs one 20 msec frame into 33 bytes, leaving 4 bits unused. Most VoIP software and equipment (except that produced by

Microsoft) use this packaging for RTP transmission, and these streams are almost universally understood. However, newer Microsoft software does support this method, and it is rarely used for WAVE files.

- Microsoft GSM packs a 40 msec double-frame into 65 bytes. This method is used only by Microsoft, and is the method most used for WAVE files.

Starting in release 7.6, Stream Manager supports both methods, and allows you to specify which method to use for each service by adding the new value `msgsm` to the configuration options `sip-annc-codecs`, `sip-conf-codecs`, `sip-http-codecs`, `sip-pcap-codecs`, and `sip-record-codecs`. When `msgsm` is specified, and it is the default value in release 7.6, Stream Manager uses the Microsoft GSM method for the given operation. In particular:

- An Announcement service uses media files encoded with Microsoft GSM, with the `msgsm.wav` suffix. The use of `gsm.wav` files is also possible if both `gsm` and `msgsm` are specified by the `sip-annc-transcode` option, (which turns on transcoding from `gsm` to `msgsm`), although this approach is not recommended.
- For all RTP transmissions, Stream Manager uses 65-byte double-frame packets by default.
- However, if a standard GSM 33-byte frame is received from the endpoint, Stream Manager switches to transcoding mode and converts all subsequent packets into that format.

  Converting one Microsoft GSM double-frame into two GSM frames requires only the repacking of the same bits differently, so it is not too resource-intensive, and does not, by itself, affect voice quality. However, because converted frames are still sent in one packet, transcoding may cause problems if the receiver's jitter buffer is too small.
- For recording operations, files are recorded in a Microsoft-compatible format that is readable by most third-party tools, even if the incoming stream uses standard GSM. This is the only case where Stream Manager may perform any codec conversion in recording.

For comparison, when the value `gsm` is specified, Stream Manager works as follows:

- An Announcement service uses media files encoded with standard GSM with the `gsm.wav` suffix, unless transcoding from `msgsm` to `gsm` is enabled.
- For all RTP transmissions, Stream Manager uses 33-byte frame packets, regardless of what is received from the other side.

  Switching to Microsoft GSM is disabled intentionally. This avoids a double format switch when the other side uses the same algorithm but is configured differently (for example, if there is another instance of Stream Manager for "emergency" recording or conference party on hold). However, there is limited rationale for this switch anyway, as almost all modern endpoints support standard GSM.

- However, Stream Manager still understands and properly decodes Microsoft GSM double-frames for recording and conferences, even though they do not cause any switch or diagnostic messages.
- For recording operations, files are recorded in standard GSM format. Since these WAVE files are not widely supported, using `gsm` in `sip-record-codecs` is not recommended.

In addition, when Stream Manager selects a codec for recording using the intersection of two lists, the values `gsm` and `msgsm` are interpreted as referring to the same codec. In other words, Call Recording will work with a configuration of:

```
sip-conf-codecs = pcmu,gsm
sip-record-codecs = msgsm
```

even though `msgsm` is not specified by the `sip-conf-codecs` option. In this case, Stream Manager uses standard GSM for communication with call participants, but records files in Microsoft GSM format, converting in real-time.

Contact Section

This section must be named `contact`. The name is case-sensitive.

This section contains options that specify Stream Manager contact information for SIP and RTP protocols.

rtp-address

Default Value: `$HOST`
Valid Values: `$HOST`, `$AUTO`, or any IP address
Changes Take Effect: Immediately

Specifies the IP address used by Stream Manager for RTP communication. When the value is set to `$AUTO`, Stream Manager automatically retrieves the IP address of the host where it is currently running. This option may be used on multi-homed hosts to select specific network interfaces for RTP traffic.

Note: The `$AUTO` value does not work in a default installation of Red Hat Linux, because the command `gethostbyname` returns `127.0.0.1` as the local host name. You must edit the `/etc/hosts` file for the `$AUTO` value to work.

max-ports

Default Value: `2000`
Valid Values: `4-65534`
Changes Take Effect: Immediately

Defines the maximum allowable number of ports for RTP/RTCP connections. This option and `rtp-port` together define a range of ports for use by Stream

Manager. To prevent overlap when using multiple Stream Managers, you can use `rtp-port` and `max-ports` to define a different range for each Stream Manager.

rtp-port

Default Value: `8000`
Valid Values: `1024-65535`
Changes Take Effect: Immediately

Specifies the initial port for RTP/RTCP connections. This option and the `max-ports` option together define a range of ports for use by Stream Manager.

sip-port

Default Value: `0`
Valid Values: Any valid port number (`5060` is recommended)
Changes Take Effect: After application restart
Specifies the port used for SIP Server connections. When the value is set to `0`, no SIP requests are processed. This option is not used when Stream Manager operates with T-Server Cisco CallManager.

Codecs Section

This section must be named `codecs`. The name is case-sensitive.

This section contains options for setting up codecs, applicable mostly to SIP server.

codec-choice priority

Default Value: `offer`
Valid Values: `offer, option`
Changes Take Effect: Immediately

Specifies in which order Stream Manager is to select a codec when it receives an SDP with more than one codec. When this option is set to `option`, Stream Manager ignores the order in which the codecs are listed in the SDP, and selects a codec in the order specified by the applicable `sip-annc-codecs`, `sip_conf_codecs`, `sip-pcap-codecs`, or `sip-record-codec` configuration option. This value provides more consistency in codec selection, with one exception. If the initial INVITE message arrives without SDP, and Stream Manager responds with a list of codecs, the remote party must still select the particular codec from the list.

If this option is set to `offer`, Stream Manager selects a codec in the order specified in the SDP.

packet-size

Default Value: `g711=20,gsm=20,g723=30,g729=20`
Valid Values: A comma-separated list of <codec_name>=<packet_size> pairs derived from the values in Table 14 (default values are underlined).

Table 14: Valid Packet Sizes for packet-size Configuration Option

Codec Name	Packet Size (ms)
g711 (both pcmu and pcma codecs)	10, <u>20</u>, 30, 40
gsm (does not affect msgm, which always uses 40 msec)	<u>20</u>, 40
g723	<u>30</u>, 60
g729	10, <u>20</u>, 30, 40

Changes Take Effect: Immediately (for new sessions)

Specifies the packet size (in milliseconds) that Stream Manager should use when generating an RTP stream with a particular audio codec. Using a larger than default packet size will increase Stream Manager performance, but may result in voice quality degradation on end points with smaller jitter buffers.

sip-annc-codecs

Default Value: `pcmu,pcma,msgsm,g729,g723`
Valid Value: A comma-separated list of any of the following codec names—`pcma, gsm` or `msgsm, g729, g723, h261, h263, h264=N`
Changes Take Effect: Immediately

Specifies the list of codecs to be used for the Announcement Service. A particular codec will be selected from this list based on the list of supported codecs supplied by the end point, the presence of a media file for the codec, and the list of codec transcoding allowed by the `sip-annc-transcode` option (below). Only one `gsm` or `msgm` value can be specified at the same time. See "gsm or msgsm?" on page 58.

Note: Starting in release 7.5, the default value lists less resource-intensive codecs first.

sip-annc-transcode

Default Value: `pcmu,pcma,msgsm`
Valid Value: A comma-separated list of any of the following codec names—`pcmu, pcma, gsm, msgsm, g723, g729`
Changes Take Effect: Immediately

Specifies the list of codecs to be used for any necessary transcoding for the Announcement Service. Stream manager will perform transcoding only when file and transmission codecs are both specified in this option. For this option, `gsm` and `msgm` can be specified at the same time.

For example, if configuration options `sip-annc-codecs` and `sip-annc-transcode` are set to their respective default values, and the requested media file exists in codecs `pcmu` and g729, then Stream Manager will use the following codecs (listed in order of priority):

- `pcmu`
- `g729`
- `pcma`—transcoded from `pcmu`

sip-conf-codecs

Default Value: `pcmu,pcma,msgsm`
Valid Values: A comma-separated list of any of the following codec names—`pcmu, pcma, gsm` or `msgsm, g729, g723, h261, h263, h264=N`
Changes Take Effect: Immediately

Specifies the list of codecs to be used for the Conference Service (which includes Silent Voice Monitoring, Voice Recording, Whisper Coaching, and Push Video) and Call Recording in mixed mode. Only one `gsm` or `msgm` value can be specified at the same time. See "gsm or msgsm?" on page 58.

Note: Starting in release 7.5, the default value lists less resource-intensive codecs first.

Table 15 lists the Session Description Protocol (SDP) codec names. The option values can consist of a list of SDP codec names separated by comma. Codecs that are not specified in the list will not be used for this service.

Table 15: SDP Codec Names

Full codec Name	Name in Option	annc	conf	http
ITU-T G.711 mu-law audio 8 KHz (64Kbit)	pcmu	+	+	+
ITU-T G.711 A--law audio 8 KHz (64Kbit)	pcma	+	+	+
ITU-T G.723.1 MP-MLQ ACELP audio	g723	+	+	+
ITU-T G.729/729a CS-ACELP audio	g729	+	+	+
GSM Full Rate audio	gsm	+	+	+

Table 15: SDP Codec Names (Continued)

Full codec Name	Name in Option	annc	conf	http
ITU-T H.261 video	`h261`	+	Support for Silent Monitoring, Whisper Coaching, and Push video only	`NO`
ITU-T H.263 video	`h263`	+	Support for Silent Monitoring, Whisper Coaching, and Push video only	`NO`
ITU-T H.264 video	`h264`	+	Support for Silent Monitoring, Whisper Coaching, and Push video only	`NO`

Note: For the H.264 video codec, a dynamic RTP payload type (consisting of any integer from 96 to 127) must be also specified.

sip-g723-fmtp

Default Value: An empty string
Valid Values: Any string with parameters in the same format as present in SDP (but without the `fmtp:N` prefix)
Changes Take Effect: Immediately

Specifies the parameter string for the G.723.1 audio codec that Stream Manager provides in the SDP sent in response to an `INVITE` message. This option does not affect the RTP stream generated by Stream Manager.

By default, Stream Manager does not provide any codec parameters in the SDP. Use this option if such parameters are necessary, such as to restrict the usage of low-rate and/or SID frames by the remote side.

RFC 3555 defines the following codec parameters:

`annexa=yes`	Annex A, Voice Activity Detection, is used or preferred.
`annexa=no`	Annex A, Voice Activity Detection, is not used or preferred.
`bitrate=5.3`	A data rate of 5.3 Kb/sec is used or preferred for the audio bit stream.
`bitrate=6.3`	A data rate of 6.3 Kb/sec is used or preferred for the audio bit stream.

sip-g729-fmtp

Default Value: An empty string
Valid Values: Any string with parameters in the same format as present in SDP (but without the `fmtp:N` prefix)
Changes Take Effect: Immediately

Specifies the parameter string for the G.729 audio codec that Stream Manager provides in the SDP sent in response to an `INVITE` message. This option does not affect the RTP stream generated by Stream Manager.

By default, Stream Manager does not provide any codec parameters in the SDP. Use this option if such parameters are necessary, such as to confirm that Voice Activity Detection is disabled, as is required by some gateways.

RFC 3555 defines the following codec parameters:

`annexb=yes`	Annex B, Voice Activity Detection, is used or preferred.
`annexb=no`	Annex B, Voice Activity Detection, is not used or preferred.

sip-h261-fmtp

Default Value: An empty string
Valid Values: Any string with parameters in the same format as present in SDP (but without the `fmtp:N` prefix)
Changes Take Effect: Immediately

Specifies the parameter string for the H.261 video codec that will be provided in the response to an `INVITE` message that does not contain Session Description Protocol (SDP). Starting with the 7.5 release, this option does not affect the selection of the video file for playback.

Note: The format parameters in the SDP always describe the receiver's capabilities, so the `sip-h26x-fmtp` options are not necessary for video playback operation. The purpose of these options is to select the video format for "push video" capabilities.

sip-h263-fmtp

Default Value: An empty string
Valid Values: Any string with parameters in the same format as present in SDP (but without the `fmtp:N` prefix)
Changes Take Effect: Immediately

Specifies the parameter string for the H.263 video codec that will be provided in the response to an `INVITE` message that does not contain Session Description Protocol (SDP). Starting with the 7.5 release, this option does not affect the selection of the video file for playback.

sip-h264-fmtp

Default Value: An empty string
Valid Values: Any string with parameters in the same format as present in SDP (but without the `fmtp:N` prefix)
Changes Take Effect: Immediately

Specifies the parameter string for the H.264 video codec that will be provided in the response to an `INVITE` message that does not contain Session Description Protocol (SDP). Starting with the 7.5 release, this option does not affect the selection of the video file for playback.

sip-http-codecs

Default Value: `pcmu`
Valid Value: A comma-separated list of any of the following codec names—`g723, g729, gsm, pcmu, pcma`
Changes Take Effect: Immediately

Specifies the list of codecs to be used for the Announcement Service using a media stream accessible through the HTTP protocol.

Note: Only audio streaming is supported.

sip-pcap-codecs

Default Value: `pcmu,pcma,msgsm,g729,g723,h261,h263,h264=108`
Valid Values: A comma-separated list of any of the following codec names—`pcmu, pcma, gsm, g723, g729, h261, h263, h264=N`
Changes Take Effect: Immediately

Specifies the list of codecs that are used when performing call recording in `pcap` mode.

Note: Starting with release 7.5, Stream Manager can record calls with video portion. However, only the audio portion of the call is recorded. Video codecs are included in this list so that Stream Manager can relay the video stream from one end point to another end point.

sip-record-codecs

Default Value: `pcmu,pcma,msgsm`
Valid Value: A comma-separated list of any of the following codec names—`pcmu, pcma, gsm` or `msgsm, g723, g729`
Changes Take Effect: Immediately

Specifies the list of codecs to be used for the Call or Conference Call Recording and Recording Announcement Service. Only one `gsm` or `msgsm` value can be specified at the same time. See “gsm or msgsm?” on page 58.

Codec Selection

Table 16 summarizes the options used for codec selection in a given scenario.

Table 16: Codec Selection Scenarios

Scenario	SIP URL in request	Codec selected based on this configuration option:

Music-on-hold and announcements from local file, IVR service, DTMF playback.	`annc@sm;play=file:<xxx>`
	or
	`annc@sm;play=dtmf:<xxx>`
		`sip-annc-codecs`
		When looking for the media file, Stream Manager also uses the `sip-annc-transcode` option. This does not affect the codec used for transmission.

Music-on-hold and announcements through HTTP interface	`annc@sm;play=http:<xxx>`	`sip-http-codecs`
Conference, (silent) monitoring, whisper coaching, and push video	`conf=<ID>@sm`	`sip-conf-codecs`
Recording treatment without playback	`annc@sm;record=<xxx>`	`sip-record-codecs`
Recording treatment with playback	`annc@sm;play=<xxx>;record=<yyy>`	`sip-record-codecs` Only values present also in the `sip-annc-codecs` or `sip-http-codecs` option (depending on the URL) are used.

Emergency recording (conference with `gcti::record` pseudo-DN)	`conf=<ID>@sm`
	and
	`annc@sm;record=<xxx>`
		`sip-record-codecs`
		Only values present also in the `sip-conf-codecs` option are used.

Call recording in `mixed` mode	`conf=<ID>@sm;record=<xxx>` two-step codec negotiation	`sip-record-codecs` Only values present also in the `sip-conf-codecs` option are used.
Call recording in `pcap` mode	`conf=<ID>@sm;record=<xxx>` two-step codec negotiation	`sip-record-codecs` Only values present also in the `sip-pcap-codecs` option are used.

Limits Section

This section must be named `limits`. The name is case-sensitive.

This section contains options for specifying timeouts and other constraints. For all of these options, the defaults options are usually adequate, but you may wish to change them to fine-tune performance or to troubleshoot a problem.

conf-cleanup-timeout

Default Value: `2 sec`
Valid Values: Any time value
Changes Take Effect: Immediately

Specifies the timeout before Stream Manager begins to clear the call legs of a conference call. This activity takes place after Stream Manager has detected that all call participants have disconnected (either normally or by inactivity timeout) and when only the announcement and recording call legs remain. This option frees resources when SIP call control is lost due to network or software failures, and when the call has been disconnected but not cleared from memory.

Notes: If network problems persist, the actual conference call completion may be delayed by the SIP call stack timeout.

If all conference call participants place the call on hold, and one of the participants disconnects from the call, Stream Manager may clear the conference call even if the SIP call control is active.

file-cache-size

Default Value: `10 Mb`
Valid Values: `200-204800`
Changes Take Effect: Immediately

Sets the size of the memory cache for playing media files. All files that are currently being played are always kept in memory, so this option affects only files that are kept after the play operation has completed. The unit can be specified in KB, MB, or GB units. If the unit is not specified, then kilobytes are assumed.

max-mixer-delay

Default Value: `60 msec`
Valid Values: Any time value
Changes Take Effect: Immediately

Defines the maximum time delay (in milliseconds) for incoming audio data from conference participants. It can be used to define the maximum size of the jitter buffer. Increasing the value may help to eliminate “choppy” audio that can be caused by dropped RTP packets. The increased value can increase the delay in voice packets dropped because of the buffer overflow, but it may also introduce delays in voice propagation. Since Stream Manager uses a dynamic jitter buffer, this option does not affect voice quality or minor delays when receiving packets.

max-record-file-size

Default Value: `0`
Valid Values: Any non-negative integer
Changes Take Effect: Immediately for a new file

Defines the maximum size of the audio file used for recording. The default value `0` (zero) means that the file size is unlimited. The unit can be specified in KB, MB, or GB units. If the unit is not specified, then bytes are assumed.

max-record-silence

Default Value: `0`
Valid Values: Any time value
Changes Take Effect: Immediately for a new file

Defines the maximum allowable amount of time that silence can be detected during a recording. If the value is set to `0` (zero), silence detection is not used.

Note: This option does not apply to regular, non-emergency Call Recording.

max-record-time

Default Value: `0`
Valid Values: Any time value
Changes Take Effect: Immediately for a new file

Defines the maximum recording time, in seconds. The default value `0` (zero) means that the recording time is unlimited.

rtcp-inactivity-timeout

Default Value: `30 sec`
Valid Values: Any time value
Changes Take Effect: Immediately

Defines the maximum time Stream Manager sends out RTP packets in a call without receiving any RTCP reports. If this time is exceeded, Stream Manager drops the call. The purpose of this mechanism is to delete calls that did not terminate properly. Missing RTCP reports are usually a consequence of a hardware or software failure (for example, PC or softphone failures, loss of network connectivity, loss of power to hardphone, and so on). If `rtcp-inactivity-timeout` is set to `0` (zero), this option is disabled.

rtp-stream-delay

Default Value: `60 msec`
Valid Values: Any time value
Changes Take Effect: After application restart

Defines the maximum acceptable delay when Stream Manager is playing an audio file. Stream Manager will drop the voice packet if it cannot send it for more than this interval.

Note: This option takes effect only when under heavy load, such as when Stream Manager is hitting the available resource limit. This option does not take effect during normal operation, such as when voice packets are sent according to the specified codec parameters.

sip-http-delay

Default Value: `120 msec`
Valid Values: Any time value
Changes Take Effect: Immediately

Specifies the initial delay when processing treatment and music-on-hold services provided by an HTTP interface to a streaming device.

sip-load-threshold

Default Value: `0`
Valid Values: Any non-negative integer
Changes Take Effect: Immediately

Specifies the maximum allowed load for Stream Manager in a load-sharing environment with multiple Stream Managers running under control of SIP Server. The load is stated in Resource Usage Estimation (`RUE`) units that approximate load based on the operation performed and the codecs used. The default value `0` (zero) disables load control; a positive integer specifies the maximum load after which Stream Manager rejects new service requests.

For an estimated number of participants that Stream Manager can handle per codec, on a typically configured computer, see Table 1, “Number of Participants Per Codec,” on page 20.

The recommended value for this option depends on processing power of the computer. Since accurate measurement of CPU utilization is too complex and too platform-dependent, Stream Manager 7.6 uses load estimates based on the currently active operation. These estimates are calculated as the sum of Resource Usage Estimates (RUE) units for all active RTP legs (audio and video, if applicable), using the numbers in Table 17 on page 71 for each leg.

Table 17: Resource Usage Estimate (RUE) Units for Call Legs

Operation (leg)	Codec	RUE
Announcement Service (playback, recording, or both) Playback performance depends mostly on the number of packets sent per second, which is a function of packet size. Microsoft GSM always uses 40-ms frames (or the number of packets if half of what is required for 20-ms frames); for most other codecs, the option `packet-size` controls the packet size.	Microsoft GSM, or any audio codec, with a packet size 40 ms or greater	2
	Any audio codec with a packet size less than 40 ms	3
	Any video codec	4
Party in regular Conference Full mixing without optimization is used for this particular leg; both a decoder and encoder is deployed.	G.711 (`pcmu` or `pcma`)	3
	GSM (both Microsoft and standard)	8
	G.729a on Intel platform	10
	G.729a on other platforms	25
	G.723.1	30
Party in special Conference with optimization (Call Monitoring, Whisper Coaching, or Call Recording in `mixed` mode) Passthrough mode is used for the leg in question, so a decoder is deployed only for the audio stream; there is no processing of the video stream.	G.711 (`pcmu` or `pcma`)	3
	GSM (both Microsoft and standard)	5
	G.729a on Intel platform	5
	G.729a on other platforms	10
	G.723.1	7
	Any video codec	3
One party in release 7.5 Call Recording with `pcap` recording mode. SIP Server 7.6 does not perform recording when video is present, so data for video codecs is provided only for reference.	Any audio codec	2
	Any video codec	3

X-Config Section

This section must be named `x-config`. The name is case-sensitive.

This section contains all other Stream Manager configuration options.

audio-file-format

Default Value: `.wav`
Valid Values: `.au` or `.wav`
Changes Take Effect: Immediately

Specifies the file format (`.au` or `.wav`) used for media files in Announcement and Music-on-Hold services. Windows and UNIX platforms support both `.au` and `.wav` files.

beep-on-rtp-nte

Default Value: `false`
Valid Values: `true, false`
Changes Take Effect: Immediately

When this option is set to `true`, Stream Manager will generate an audible multi-frequency tone (a combination of 600 Hz and 800 Hz tones) when an out-of-band Dual Tone Multi-Frequency (DTMF) signal is received from one of the conference participants in an RTP packet using an NTE payload (according to RFC 2833). The tone duration matches the duration of the DTMF payload. This tone will be heard by all other conference participants, but not by the party that invoked the DTMF signal.

h261-annc-fmts

Default Value: `QCIF=2`
Valid Values: A comma-separated list of H.261 video formats (maximum 8 formats)
Changes Take Effect: Immediately

Specifies the list of video formats (in order of preference) that are used by Stream Manager for video purposes. When Stream Manager receives a request to play a video file, it selects the first format from the list that has the corresponding media file and is supported by the end point. See Table 18 on page 73 for a list of video formats.

h263-annc-fmts

Default Value: `QCIF=2`
Valid Values: A comma-separated list of H.263 video formats (maximum 16 formats)
Changes Take Effect: Immediately

Specifies the list of video formats (in order of preference) that are used by Stream Manager for video purposes. When Stream Manager receives a request to play a video file, it selects the first format from the list that has the corresponding media file and is supported by the end point.

Table 18 on page 73 lists the H.261 and H.263 video codecs format specified as a pair of pre-defined picture size and Minimum Picture Interval (MPI) values separated by an equal sign. The video frame rate is calculated as NTSC base rate (29.97_fps) divided by the MPI value. Stream Manager supports only MPI values from 1 to 6.

Table 18: H.261/H.263 Video Formats

Picture Size/ Allowed MPI	SQCIF (128x96)	QCIF (176x144)	CIF (352x288)	CIF (704x576)	CIF16 (1408x1152)
H.261 codec	-	1 to 4	1 to 4	-	-
H.263 codec	1 to 6	1 to 6	1 to 6	1 to 6	1 to 6

h264-annc-fmts

Default Value: `42e00a=2`
Valid Values: A comma-separated list of H.264 video formats (up to a maximum of 16 formats)
Changes Take Effect: Immediately

Specifies the list of video formats (in order of preference) that are used by Stream Manager for video purposes. When Stream Manager receives a request to play a video file, it selects the first format from the list that has the corresponding media file and is supported by the end point.

Table 19 lists the H.264 video codecs format specified as a pair of profile-level-id values and MPI values separated by an equal sign. Stream Manager supports MPI values from 1 to 6 only.

Table 19: H.264 Video Formats

Picture Size/ profile-level-id	Level	SQCIF (128x96)	QCIF (176x144)	QVGA (320x240)	CIF (352x288)	CIF4 (704x576)
xxxx0a	1	1 to 6	2 to 6	-	-	-
xxxx0b	1.1	1 to 6	1 to 6	3 to 6	4 to 6	-
xxxx0c	1.2	1 to 6	1 to 6	2 to 6	2 to 6	-
xxxx0d	1.3	1 to 6	1 to 6	1 to 6	1 to 6	-
xxxx14	2	1 to 6	1 to 6	1 to 6	1 to 6	-

Table 19: H.264 Video Formats (Continued)

Picture Size/ profile-level-id	Level	SQCIF (128x96)	QCIF (176x144)	QVGA (320x240)	CIF (352x288)	CIF4 (704x576)
xxxx15	2.1	1 to 6	1 to 6	1 to 6	1 to 6	-
xxxx16	2.2	1 to 6	1 to 6	1 to 6	1 to 6	3 to 6
xxxx1e	3	1 to 6	1 to 6	1 to 6	1 to 6	2 to 6

log-trace-flags

Default Value: `+rtcp`
Valid Values: Any combination of the following, separated by spaces:

`+ping`	Turns on printing of `PING` and `PING ACK` messages between Stream Manager and T-Server Cisco Call Manager (all other messages are always printed into trace stream). This value does not apply when Stream Manager works with SIP Server.
`+rtp/nte`	Turns on printing of all RTP packets with NTE (Named Telephony Event) payload, even if they are not used. Packets used for `TONE NOTIFY` or `SIP INFO` messages are is always printed.
`+rtp/dump`	Adds hexadecimal dump of the payload contents to each RTP packet printed.
`+rtcp`	Turns on printout of RTCP packets (both incoming and outgoing).
`-rtcp`	Turns off printout of RTCP packets.

Changes Take Effect: Immediately

Specifies the types of information that are written into the log files.

rtp-close-delay

Default Value: `2 sec`
Valid Values: Any time value
Changes Take Effect: Immediately

Specifies the interval that Stream Manager delays closing the RTP port after the session is terminated. The value should be sufficient enough to avoid ICMP `Port unreachable` messages that are generated on Windows when RTP packets cannot be delivered. Because of possible delays with delivering signalling information, the end point may continue trying to communicate with Stream Manager after the session is terminated.

rtp-ip-tos

Default Value: `0`
Valid Values: Integer `0-255`, in decimal format or in hexadecimal format with `0x` prefix
Changes Take Effect: Immediately for a new RTP stream

Defines the value of the Type of Service (TOS) byte in the IP header of RTP packets sent by Stream Manager. Depending on the network configuration, the TOS byte is treated as either:

- 3-bit IP precedence field, followed by a 4-bit type-of-service. The least significant bit (LSB) is unused and set to 0. (RFC 1349)
- 6 bit DiffServ, with the two least significant bits unused. (RFC 2774)

For example, the following values may be used to assign a higher priority to RTP packets:

- `0x10`—IPTOS_LOWDELAY, low-delay type of service
- `0x20`—IPTOS_PREC_PRIORITY, priority precedence
- `0x40`—IPTOS_PREC_CRITICAL, critical precedence
- `0xB8`—DiffServ EF (Expedited Forward)

Note: On most operating systems, applications running on behalf of non-privileged user accounts are not permitted to set a non-zero TOS value, so you may have to perform additional actions to enable this functionality. In particular:

- On Linux, the application must have CAP_NET_ADMIN capability (that is, run from the `root` account).
- On Windows, the following registry setting must be set (see also http://support.microsoft.com\kb/248611):

```
HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet\Services\TcpIp\Pa
rameters
DisableUser TOSSetting = (DWORD) 0
```

Refer to operating system documentation for further information.

sip-call-record-mode

Default Value: `mixed`
Valid Values:

`pcap`	Stream Manager creates a file with all RTP packets captured. That file is created in standard PCAP format, and no conversion of the RTP stream between the parties is performed. This value has no performance penalty for any codec.
mixed	An audio file containing the recording of the mixed conversation is created. Stream Manager creates a regular conference call and adds an internal recording party to the call.

Changes Take Effect: Immediately

Specifies the mode of call recording when SIP Server creates a two-party conference call that requires Stream Manager recording capabilities.

sip-conf-gain

Default Value: An empty string
Valid Values: A comma-separated list of pairs where `orig-name = gain`
Changes Take Effect: When a new conference is created

Provides gain control for conference participants based on the names provided in the Session Description Protocol (SDP) message received from a particular end point. The values can be specified as follows:

- `orig-name`—the originator name that is taken from the first words of the `o=` or `s=` lines received in the SDP message. The value is case-sensitive and must be in single or double quotes. For example, if the received SDP message contains the following lines:

  `o=Genesys 137 137 IN IP4 192.168.7.60`

  `s=StreamManager 7.2.002.06`

  The `orig-name` will be created as Genesys/StreamManager.

- `gain`—a floating point number that specifies the multiplier to be applied to the voice stream from that particular end point.

  For example, the option may be set as follows to reduce the volume of a `music-in-hold` stream in a conference call when it is played by Stream Manager:

  `sip-conf-gain = "Genesys/StreamManager"=0.5`

Note: Values greater than 1.0 cause Stream Manager to increase the sound volume and may result in poor sound quality because of waveform clipping.

sip-dtmf-delay

Default Value: `120 msec`
Valid Values: Any time value
Changes Take Effect: Immediately

Specifies the delay before Stream Manager starts DTMF generation. The default value is sufficient in most cases, but if the first DTMF digit is being clipped (or not played or recognized), the delay can be increased.

sip-dtmf-duration

Default Value: `240 msec`
Valid Values: `100 msec–8 sec`
Changes Take Effect: Immediately

Specifies the default duration used for DTMF generation, if no duration is specified in the request. If this option is not specified as a 40-msec interval (that is, evenly divisible by 40), Stream Manager will round the value down to the nearest valid interval. The duration specified by this option includes a fixed 40 msec after-digit pause. For example, the default duration of `240 msec` consists of a tone duration of 200 msec followed by a 40 msec pause.

sip-dtmf-method

Default Value: `both`
Valid Values:

`rfc2833`	RTP packets with Named Telephony Event (NTE) payload as specified by RFC 2833
`tone`	In-band audio tones
`both`	Simultaneous RTP packets and in-band audio tones, as described above
`alt`	Alternating RTP packets and in-band audio tones, as described above, in sequence (one for half the duration, followed by the other for the rest of the duration)

Changes Take Effect: Immediately

Specifies the default method used for DTMF generation

Note: In-band DTMF is reliably delivered only when a low-compression codec, such as G.711, is used. Therefore, when using the tone method, Stream Manager gives higher priority to `pcma` and `pcmu` codecs.

sip-send-info

Default value: `auto`
Valid values:

`true`	`INFO` message is generated
`false`	`INFO` message is not generated
`auto`	`INFO` message is generated if the `Allowed` header is in the `INVITE` message, and it contains the `INFO` parameter. If the `Allowed` header is not present, the message is assumed to be allowed.

Changes Take Effect: Immediately

Controls whether Stream Manager generates SIP Server `INFO` messages containing the `application/dtmf-relay` parameter when receiving NTE packets over an RTP connection.

x-type

Default Value: `sm`
Valid Value: `sm`
Changes Take Effect: Not applicable

Specifies the component to which the `x-config` options apply.

Changes from 7.5 to 7.6

Table 20 on page 78 lists configuration options that have changed between Stream Manager releases 7.5 and 7.6.

Notes: For release 7.6, many Stream Manager configuration options are reorganized into new or different groups, and two options are renamed. This is also described in Table 20.

The former locations and names are still recognized by Stream Manager for purposes of backwards compatibility.

Table 20: Option Changes from Release 7.5 to 7.6

Option Name	Option Values	Type of Change	Details
Call Section			
call (section)		Removed	Contained options moved to new section or removed completely.

Table 20: Option Changes from Release 7.5 to 7.6 (Continued)

Option Name	Option Values	Type of Change	Details
call-protocol	sm	Removed	Used by other components in IPMX architecture; never used by Stream Manager itself.
call-address	$HOST, $AUTO, or IP address	Modified, moved	Renamed to rtp-address; moved to new section contact.
Contact Section			
contact (section)		Added	See page 60.
rtp-address	$HOST, $AUTO, or IP address	Modified, moved	Formerly call-address; moved from call section. See page 60.
max-ports	4-65534	Moved	Moved from x-config section. See page 60.
rtp-port	1024-65535	Moved	Moved from x-config section. See page 61.
sip-port	Valid port number (5060 is recommended)	Moved	Moved from x-config section. See page 61.
Codecs Section			
codecs (section)		Added	See page 61.
codec-choice-priority	offer, option	Added	See page 61.
packet-size	<codec_name>=<packet_size>	See Details	Added in release 7.5, documented in 7.6. See page 62.
sip-annc-codecs	List of codecs	Modified, moved	Default and valid values changed; moved from x-config section. See page 62.
sip-annc-transcode	List of codecs	Added	See page 62.
sip-conf-codecs	List of codecs	Modified, moved	Default and valid values changed; moved from x-config section. See page 63.
sip-g723-fmtp	Empty string or Annex A and/or bit rate	Added	See page 64.

Table 20: Option Changes from Release 7.5 to 7.6 (Continued)

Option Name	Option Values	Type of Change	Details
sip-g729-fmtp	Empty string or Annex D and/or bit rate	Added	See page 64.
sip-h261-fmtp	Empty string or SDP parameters	Moved	Moved from `x-config` section. See page 65.
sip-h263-fmtp	Empty string or SDP parameters	Moved	Moved from `x-config` section. See page 65.
sip-h264-fmtp	Empty string or SDP parameters	See Details	Added in earlier release, documented in 7.6. See page 65.
sip-http-codecs	List of codecs	Moved	Moved from `x-config` section. See page 66.
sip-pcap-codecs	List of codecs	Modified, moved	Default and valid values changed; moved from `x-config` section. See page 66.
sip-record-codecs	List of codecs	Modified, moved	Formerly `sip-record-codec`; default and valid values changed; moved from `x-config` section. See page 66.
Limits Section			
limits (section)		Added	See page 68.
conf-cleanup-timeout	Time value	Moved	Moved from `x-config` section. See page 68.
file-cache-size	`200-204800`	Moved	Moved from `x-config` section. See page 68.
max-mixer-delay	Time value	Moved	Moved from `x-config` section. See page 68.
max-record-file-size	Non-negative integer	Moved	Moved from `x-config` section. See page 69.
max-record-silence	Time value	Moved	Moved from `x-config` section. See page 69.
max-record-time	Time value	Modified, moved	Default value changed; moved from `x-config` section. See page 69.

Table 20: Option Changes from Release 7.5 to 7.6 (Continued)

Option Name	Option Values	Type of Change	Details
rtcp-inactivity-timeout	Time value	Moved	Moved from `x-config` section. See page 69.
rtp-stream-delay	Time value	Moved	Moved from `x-config` section. See page 70.
sip-http-delay	Time value	Moved	Moved from `x-config` section. See page 70.
sip-load-threshold	Non-negative integer	Added	See page 70.
X-Config Section			
conf-cleanup-timeout	Time value	Moved	Moved to new section `limits`. See page 68.
file-cache-size	`200-204800`	Moved	Moved to new section `limits`. See page 68.
log-trace-flags	`+ping`, `+rtp/nte`, `+rtp/dump`, `+rtcp`, `-rtcp`	Modified	Valid values changed; see page 74.
max-mixer-delay	Time value	Moved	Moved to new section `limits`. See page 68.
max-ports	`4-65534`	Moved	Moved to new section `contact`. See page 60.
max-record-file-size	Non-negative integer	Moved	Moved to new section `limits`. See page 69.
max-record-silence	Time value	Moved	Moved to new section `limits`. See page 69.
max-record-time	Time value	Modified, moved	Default value changed; moved to new section `limits`. See page 69.
rtcp-inactivity-timeout	Time value	Moved	Moved to new section `limits`. See page 69.
rtp-close-delay	Time value	Added	Added in earlier release, documented in 7.6. See page 74.
rtp-ip-precedence	`0,1,2,3,4,5,6,7`	Removed	Replaced by `rtp-ip-tos`. See page 75.

Table 20: Option Changes from Release 7.5 to 7.6 (Continued)

Option Name	Option Values	Type of Change	Details
rtp-ip-tos	Integer from `0-255` in decimal format or in hexadecimal format with `0x` prefix	Added	See page 75.
rtp-port	`1024-65535`	Moved	Moved to new section `contact`. See page 61.
rtp-stream-delay	Time value	Moved	Moved to new section `limits`. See page 70.
sip-annc-codecs	List of codecs	Modified, moved	Default and valid values changed; moved to new section `codecs`. See page 62.
sip-conf-codecs	List of codecs	Modified, moved	Default and valid values changed; moved to new section `codecs`. See page 63.
sip-dtmf-delay	Time value	Added	See page 77.
sip-dtmf-duration	Time value	Modified	Default value and description changed. See page 77.
sip-dtmf-method	`rfc2833, tone, both, alt`	New	See page 77.
sip-h261-fmtp	Empty string or SDP parameters	Moved	Moved to new section `codecs`. See page 65.
sip-h263-fmtp	Empty string or SDP parameters	Moved	Moved to new section `codecs`. See page 65.
sip-h264-fmtp	Empty string or SDP parameters	See Details	Added in earlier release, documented in 7.6; moved to new section `codecs`. See page 65.
sip-http-codecs	List of codecs	Moved	Moved to new section `codecs`. See page 66.
sip-http-delay	Time value	Moved	Moved to new section `limits`. See page 70.
sip-pcap-codecs	List of codecs	Modified, moved	Default and valid values changed; moved to new section `codecs`. See page 66.

Table 20: Option Changes from Release 7.5 to 7.6 (Continued)

Option Name	Option Values	Type of Change	Details
sip-port	Valid port number (`5060` is recommended)	Moved	Moved to new section `contact`. See page 61.
sip-record-all-conf	`true, false`	Removed	
sip-record-base-name	Valid filename	Removed	
sip-record-codec	List of codecs	Renamed, moved	Renamed to `sip-record-codecs`; moved to new section `codecs`.

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