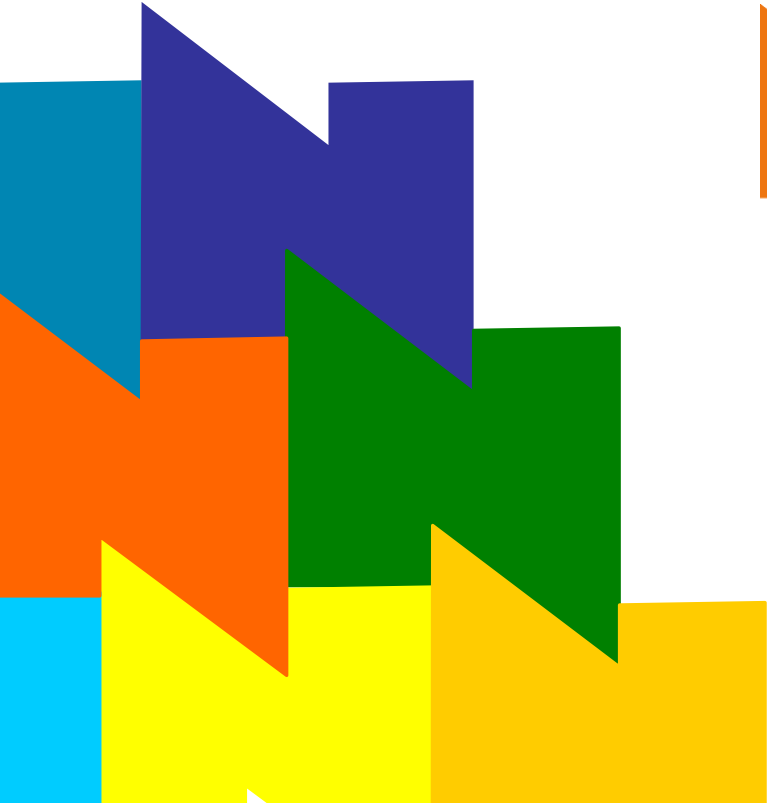


A series of overlapping geometric shapes in the top right corner, including a blue square, a purple square, a green square, an orange square, and a yellow square, all arranged in a stepped, descending pattern.

Video Mail Application Demonstration Program Manual

9000-62480-18

A series of overlapping geometric shapes in the bottom left corner, including a blue square, a purple square, a green square, an orange square, and a yellow square, all arranged in a stepped, ascending pattern.

100 Crossing Boulevard
Framingham, MA 01702-5406 USA
www.nmscommunications.com

No part of this document may be reproduced or transmitted in any form or by any means without prior written consent of NMS Communications Corporation.

© 2009 NMS Communications Corporation. All Rights Reserved.

Alliance Generation is a registered trademark of NMS Communications Corporation or its subsidiaries. NMS Communications, Natural MicroSystems, AG, CG, CX, QX, Convergence Generation, Natural Access, Natural Access MX, CT Access, Natural Call Control, Natural Media, NaturalFax, NaturalRecognition, NaturalText, Fusion, Open Telecommunications, Natural Platforms, NMS HearSay, AccessGate, MyCaller, and HMIC are trademarks or service marks of NMS Communications Corporation or its subsidiaries. Multi-Vendor Integration Protocol (MVIP) is a registered trademark of GO-MVIP, Inc. UNIX is a registered trademark in the United States and other countries, licensed exclusively through X/Open Company, Ltd. Windows NT, MS-DOS, MS Word, Windows 2000, and Windows are either registered trademarks or trademarks of Microsoft Corporation in the United States and/or other countries. Clarent and Clarent ThroughPacket are trademarks of Clarent Corporation. Sun, Sun Microsystems, Solaris, Netra, and the Sun logo are trademarks or registered trademarks of Sun Microsystems, Inc. in the United States and/or other countries. All SPARC trademarks are used under license and are trademarks or registered trademarks of SPARC International, Inc. in the United States and/or other countries. Products bearing SPARC trademarks are based upon an architecture developed by Sun Microsystems, Inc. Linux is a registered trademark of Linus Torvalds. Red Hat is a registered trademark of Red Hat, Inc. All other marks referenced herein are trademarks or service marks of the respective owner(s) of such marks. All other products used as components within this product are the trademarks, service marks, registered trademarks, or registered service marks of their respective owners.

Every effort has been made to ensure the accuracy of this manual. However, due to the ongoing improvements and revisions to our products, NMS Communications cannot guarantee the accuracy of the printed material after the date of publication or accept responsibility for errors or omissions. Revised manuals and update sheets may be published when deemed necessary by NMS Communications.

P/N 9000-62480-18

Revision history

Revision	Release date	Notes
1.0	March 2005	SRG, Video Access 1.0
1.1	September 2005	SRG, Video Access 2.0 Beta 1
1.2	October 2005	DEH, Video Access 2.0 Beta 2
1.3	December 2005	DEH, Video Access 2.0
1.4	February 2007	DEH, Video Access 3.0 Alpha
1.5	March 2007	SRG, Video Access 3.0 Beta
1.6	May 2007	PJP, Video Access 3.0 Beta 2
1.7	July 2007	PJP, Video Access 3.0
1.8	February 2009	DEH, Video Access 3.2
Last modified: January 19, 2009		

Refer to www.nmscommunications.com for product updates and for information about support policies, warranty information, and service offerings.

Table Of Contents

Chapter 1: Introduction	5
Chapter 2: Overview of vmsamp	7
vmsamp overview	7
vmsamp environment	8
Video Access document set.....	9
Chapter 3: Configuring vmsamp.....	11
Overview of configuring vmsamp.....	11
Creating the gateway configuration file	11
Creating the server configuration file.....	11
Configuring the 3G-324M Interface	12
IP interface parameters	13
System and error processing parameters	13
Video transcoder parameters.....	15
TDM interface parameters.....	15
Audio and video codec parameters.....	16
3G-324M and call setup parameters	16
Configuring the Video Messaging Server Interface.....	18
IP interface parameters	18
System parameters	19
Play and record parameters	19
Silence detection parameters	20
Audio and video codec parameters.....	22
3GP parameters	23
Chapter 4: Using vmsamp	25
Running vmsamp	25
vmsamp menu commands.....	26
Endpoint menu.....	27
H.324M Middleware menu.....	28
Call control menu	28
Play and record menu	29
Jitter buffer menu.....	29
Random access menu	30
Video transcoding menu	30
Video tromboning menu	30
Miscellaneous menu	30
Video transcoding.....	31
Switching between video pass-through and video transcoding.....	32
Setting up and tearing down a call	33
Video tromboning	34
H.245 relay	34
RTP endpoint control	36
Bridging video media streams	38
Bridging audio media streams	38
Switching back tromboned calls	39

1

Introduction

The *Video Mail Application Demonstration Program Manual* is one in a set of manuals that describe the Video Access product. This manual documents a video mail program called *vmsamp* that is included with Video Access. *vmsamp* is an executable demonstration application that can also serve as a reference on the use of Video Access components and constructs.

This manual targets video application developers who use Natural Access. It assumes that you are familiar with telephony concepts, Natural Access, and the C programming language. If you are not familiar with Natural Access, read the *Video Access Overview Manual* to learn about the Natural Access features that relate to Video Access before reviewing this manual.

2

Overview of vmsamp

vmsamp overview

The Video mail application demonstration program (*vmsamp*) is a menu-driven application that demonstrates how to:

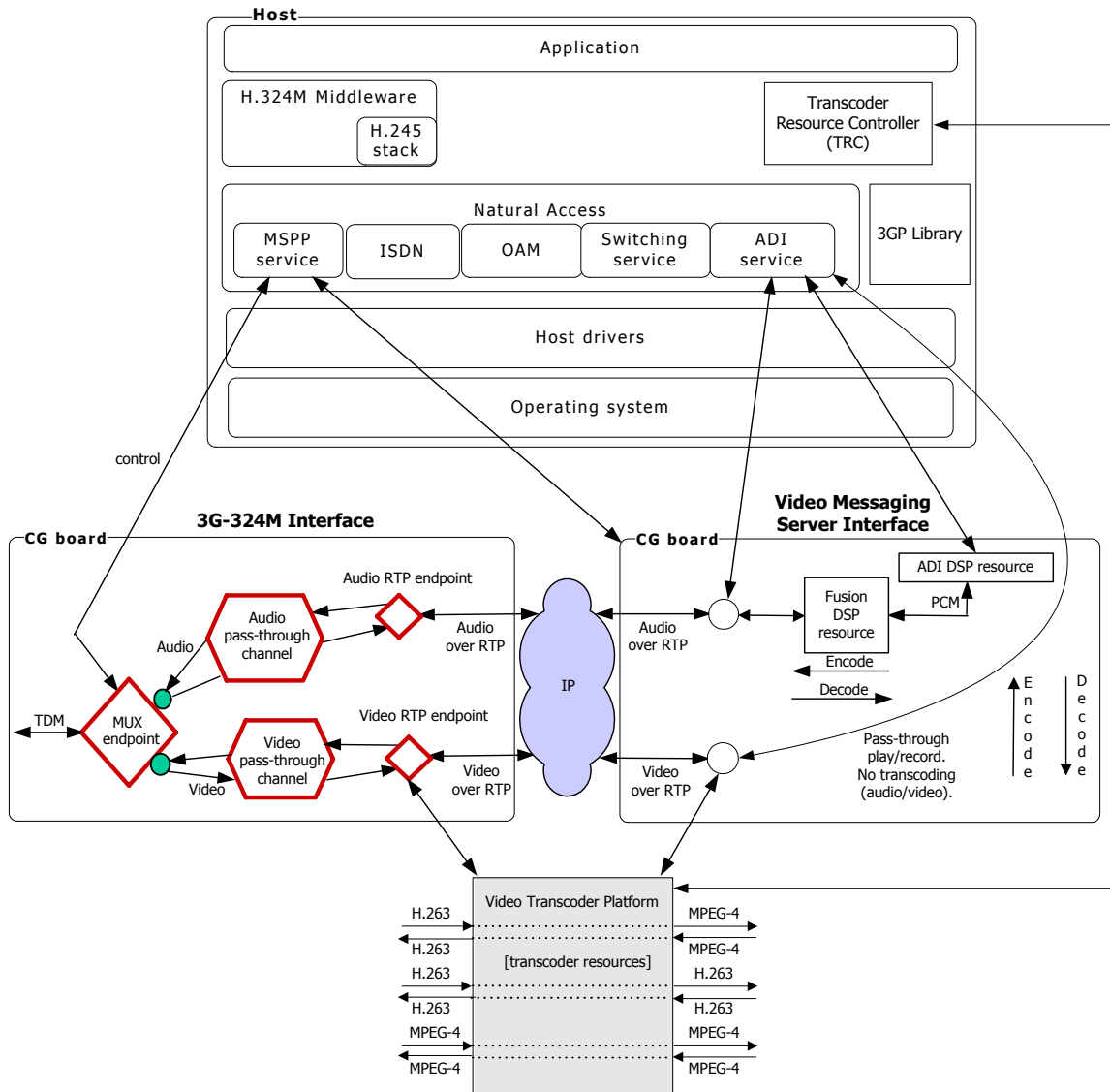
- Receive a call from a 3G terminal to the 3G-324M Interface or place a call to a 3G terminal.
- Record audio and video into a 3GP file using the Video Messaging Server Interface.
- Play 3GP files to the 3G terminal with or without video transcoding.
- Trombone two 3G-324M calls together.

You must configure *vmsamp* before you run it. For information, see *Overview of configuring vmsamp* on page 11.

vmsamp environment

The following illustration shows the environment in which the *vmsamp* functions. For clarity, the illustration shows the 3G-324M Interface and the Video Messaging Server Interface operating on independent boards. However, both interfaces can reside on the same CG board.

Natural Access services control and configure the gateway and server components except for the H.245 stack that runs on the host outside of Natural Access.



The H.324M Middleware abstracts the H.245 stack to negotiate session control with remote wireless devices using an H.223 multiplexer as the data link layer of the transport mechanism.

For demonstration purposes, PRI call control is achieved using Q.931 signaling by the onboard ISDN stack. TDM trunk connections are created using the H.110 bus on the CG board and are managed by the Natural Access Switching service. For more information, refer to the *Switching Service Developer's Reference Manual*.

Video Access document set

The following table describes each of the manuals in the Video Access documentation set, along with guidelines for their use:

Manual	Description	Use this manual if...
<i>Video Access Overview Manual</i>	A general introduction to Video Access and its features.	You are new to Video Access. Start with this manual before proceeding to the <i>Video Mail Application Demonstration Manual</i> .
<i>Video Mail Application Demonstration Program Manual</i>	How to use <i>vmsamp</i> , a functional video mail application built on Video Access and supplied with the product.	You are new to Video Access and want to gain hands-on experience with Video Access technology and code before you start writing your own applications. The <i>vmsamp</i> application includes reference code for most of the data structures and API features described in the other Video Access manuals.
<i>3G-324M Interface Developer's Reference Manual</i>	How to use the 3G-324M Interface to connect with 3G-324M terminals capable of audio and video. This manual also describes the 3G-324M Interface capabilities and functions.	You are developing gateway functionality based on the 3G-324M Interface.
<i>Video Messaging Server Interface Developer's Reference Manual</i>	How to play and record audio and video RTP media, and how to use the Video Messaging Server Interface.	Your application will use the Video Messaging Server Interface to process video and audio streams.
<i>Video Access Utilities Manual</i>	How to use the Video Access utilities that are available for manipulating 3GP files and monitoring 3G-324M calls.	You are responsible for Video Access content capture and analysis, or for the manipulation or troubleshooting of data generated or received by Video Access components. The utilities documented here can also be used to manipulate content created outside of Video Access.

Note: For an additional layer of detail about Video Access structures, refer to the Video Access header files.

3

Configuring vmsamp

Overview of configuring vmsamp

Before running *vmsamp*, ensure that you have created both the 3G-324M Interface (gateway) and Video Messaging Server Interface (server) configuration files.

Creating the gateway configuration file

To create the gateway configuration file (*gw.cfg*) in an IPv4 network, specify the following settings:

```
useVideoXc = 0
Video_Format = 2
srcIPAddr_video = <CG board IP address>
destIPAddr_video = <CG board IP address>
srcIPAddr_audio = <CG board IP address>
destIPAddr_audio = <CG board IP address>
```

To create the gateway configuration file (*gw.cfg*) in an IPv6 network, specify the following settings:

```
useVideoXc = 0
Video_Format = 2
srcIPAddr_video = <CG board IPv6 address>
destIPAddr_video = <CG board IPv6 address>
srcIPAddr_audio = <CG board IPv6 address>
destIPAddr_audio = <CG board IPv6 address>
useIPv6=1
```

For more information, refer to *Configuring the 3G-324M Interface* on page 12.

Creating the server configuration file

To create the server configuration file (*srv.cfg*) in an IPv4 network, specify the following settings:

```
3gpFileName = test.3gp
srcIPAddr_video = <CG board IP address>
destIPAddr_video = <CG board IP address>
srcIPAddr_audio = <CG board IP address>
destIPAddr_audio = <CG board IP address>
```

To create the server configuration file (*srv.cfg*) in an IPv6 network, specify the following settings:

```
3gpFileName = test.3gp
srcIPAddr_video = <CG board IPv6 address>
destIPAddr_video = <CG board IPv6 address>
srcIPAddr_audio = <CG board IPv6 address>
destIPAddr_audio = <CG board IPv6 address>
useIPv6=1
```

For more information, refer to *Configuring the Video Messaging Server Interface* on page 18.

Configuring the 3G-324M Interface

Create a gateway configuration file (*gw.cfg*) to set up the 3G-324M Interface. Each line of the configuration file must specify a different parameter value. The format of a line is:

Parameter = Value

Any text on a line appearing after a hash character (#) is ignored as a comment.

The following tables describe the types of parameters in the *gw.cfg* file:

- IP interface
- System and error processing
- Video transcoder
- TDM interface
- Audio and video codec
- 324M and call setup

IP interface parameters

The parameters in the following table define IP interface settings for the 3G-324M Interface:

Parameter	Type	Default	Description
destIPAddr_audio	string	10.1.8.94	IP address of the system (server) receiving the audio RTP stream from the 3G-324M Interface.
destIPAddr_video	string	10.1.8.94	IP address of the system (server) receiving the video RTP stream from the 3G-324M Interface.
destPort_audio	DWORD	3000	UDP port on the remote system (server) receiving the audio RTP stream.
destPort_video	DWORD	4000	UDP port on the remote system (server) receiving the video RTP stream.
h263encapRfc	unsigned int	2429	H.263 encapsulation RFC number to employ on H.263 video endpoints created: 2190: RFC 2190 2429: RFC 2429
simplexEps	int	0	Type of endpoints and channels to use for inbound and outbound audio and video paths: 0: Uses separate simplex endpoints and channels. 1: Uses duplex endpoints and channels.
srcIPAddr_audio	string	10.1.8.94	IP address of the 3G-324M Interface sending or receiving the audio RTP stream.
srcIPAddr_video	string	10.1.8.94	IP address of the 3G-324M Interface sending or receiving the video RTP stream.
srcPort_audio	DWORD	1000	UDP port on the 3G-324M Interface receiving the audio RTP stream from the server.
srcPort_video	DWORD	2000	UDP port on the 3G-324M Interface receiving the video RTP stream from the server.
useIPv6	int	0	IP network type for the gateway: 0: IPv4 1: IPv6

System and error processing parameters

The parameters in the following table define system-level and error processing settings for the 3G-324M Interface:

Parameter	Type	Default	Description
enableAVsync	int	0	Enables or disables audio and video RTP endpoints to perform incoming skew calculations: 0: Disables RTP endpoints from performing incoming skew calculations. 1: Enables endpoints from performing incoming skew calculations.
h324LogFile	string	h324.log	3G-324M Interface log file name.

Parameter	Type	Default	Description
nCalls	int	0	Load test parameter for the number of calls. A value of -1 runs the test indefinitely.
nPorts	int	1	Number of ports to run.
notifyDemuxErrors	int	0	Enables or disables DEMUX error detection on the 3G-324M interface: 0: Disables DEMUX error detection. 1: Enables DEMUX error detection.
notifyIframe	int	0	Enables or disables I-Frame reporting with DCI from both the 324M-to-IP and IP-to-324M directions: 0: Disables I-Frame reporting. 1: Enables I-Frame reporting.
outOfBandDCI mode	unsigned char	1	0: No change. 1: Replaces only initial in-band DCI with out-of-band DCI. 2: Replaces all in-band DCI with out-of-band DCI. 3: Replaces all in-band DCI with out-of-band DCI and inserts out-of-band DCI before every I-frame.
SendDciTo3gpLib	unsigned char	0	Enables or disables sending out-of-band video decoder information to the 3GP library: 0: Disables sending out-of-band video decoder information (MPEG4 or H.264) to the 3GP library. 1: Enables sending out-of-band video decoder information to the 3GP library.
StopPlayRecOnUII	int	0	Enables or disables stopping of simultaneous play/record operation when a user input indication command is received and then starts to play the recorded file. 0: Disables the automatic stopping of simultaneous play/record operation when user input is received. 1: Enables the automatic stopping of simultaneous play/record operation when user input is received.
terminalType	int	201	Determines whether the gateway is master or slave for H.245 negotiations. Range: 0-255
vidSkewTime	int	0	Video skew value (in ms) sent to a video RTP endpoint. This value is used to adjust video with respect to audio. A positive value indicates that the video lags audio. A negative value indicates that the video leads audio. The maximum value is 4095.

Video transcoder parameters

The parameters in the following table define video transcoder settings for the 3G-324M Interface:

Parameter	Type	Default	Description
trcConfigFile	string	trc.cfg	Video transcoder configuration file name. Refer to the <i>Video Transcoder Installation and Developer's Reference Manual</i> for information.
trcFileTraceMask	int	0x20	Video transcoder trace mask into file.
trcLogFile	string	trc.log	Video transcoder log file name. This file is created only when the video transcoder is in use (useVideoXc = 1).
trcTraceMask	int	0x20	Video transcoder trace mask. Refer to the <i>Video Transcoder Installation and Developer's Reference Manual</i> for information.
useVideoXc	int	0	Enables or disables use of the video transcoder in the application. 0: Does not use the video transcoder. 1: Uses the video transcoder.

TDM interface parameters

The parameters in the following table define TDM settings for the 3G-324 Interface:

Parameter	Type	Default	Description
interfaceType	string	E1	3G-324M trunk interface type: E1: E1 interface type T1: T1 interface type This value must match the value of the NetworkInterface.T1E1[x].Type keyword in the CG board's board keyword file.
isdn_country	unsigned int	1000 (Europe)	Country mode for the network operator variant in which to start. The behavior of a variant can change depending on the country specified. For valid operator and country combinations, refer to the <i>NMS ISDN Messaging API Developers Reference Manual</i> .
isdn_dNT_TE	DWORD	2 (EQUIPMENT_NT)	ISDN partner equipment setting: 1: Board acts as a network. 2: Board connects to a network.
isdn_operator	unsigned int	11 (Euro)	Network operator variant in which to start. For valid operator and country combinations, refer to the <i>NMS ISDN Messaging API Developers Reference Manual</i> .
mux_timeslot	int	0	Timeslot for creating the MUX endpoint.

Audio and video codec parameters

The parameters in the following table define audio and video codec settings for the 3G-324M Interface:

Parameter	Type	Default	Description
Audio_Format	int	1 (N_AMR)	Audio format: 1: AMR 2: G.723
enableH263Profile3	int	0 (disable)	Enables or disables support for H.263+ profile 3: 0: Disables support for H.263+ profile 3. 1: Enables support for H.263+ profile 3.
h263_maxBitRate	unsigned int	43000	H.263 maximum bit rate specified in local capabilities.
h263Resolution	int	0 (QCIF)	H.263 video resolution specified in local capabilities. Possible values are QCIF or SQCIF.
Video_AlternateFormat	int	0 (NO_VIDEO)	Secondary video format: 0: No video. 1: MPEG-4 2: H.263 4: H.264
Video_Format	int	2 (H.263)	Preferred video format: 1: MPEG-4 2: H.263 4: H.264

3G-324M and call setup parameters

The parameters in the following table define 3G-324M and call setup settings for the 3G-324M Interface:

Parameter	Type	Default	Description
advanced264Parameters	int	0	Enables or disables setting of the ProfileIOP and AcceptRedundantSlices parameters in local capabilities: 0: Disables setting the parameters. 1: Enables setting the parameters (ProfileIOP set to 0, AcceptRedundantSlices set to TRUE).
AutoHangup	int	0	Selects whether or not to disconnect the call after receiving the second ADIEVN_PLAY_DONE event. 0: Does not disconnect the call. 1: Disconnects the call.
AutoPlaceCall	int	0	0: Places a call. 1: Places a first call automatically. Place a new call when disconnected.
BetweenCallsTime	unsigned int	3000	Timeout (in milliseconds) to place a new call when disconnected in AutoPlaceCall mode.

Parameter	Type	Default	Description
DialNumber	string	2000	Number to dial when placing a call in AutoPlaceCall mode.
max_nal_unit_size	int	0	Enables or disables setting of max_nal_unit_size parameter in local capabilities per the H.241 specification. 0: Does not set max_nal_unit_size parameter in local capabilities. Any positive integer: Sets max_nal_unit_size parameter in local capabilities to specified value.
NalAlignedMode	int	0	Enables or disables setting of NalAlignedMode parameter in local capabilities: 0: Does not set NalAlignedMode parameter in local capabilities. 1: Sets NalAlignedMode in local capabilities.
specifyDci	int	1	Indicates whether to specify MPEG-4 VO/VOL or H.264 DCI in the video OLC: 0: Does not specify MPEG-4 VO/VOL or H.264 DCI in the video OLC. 1: Specifies MPEG-4 VO/VOL or H.264 DCI in the video OLC.
TromboneOnUUI	int	0	Enables or disables tromboning when a terminal sends a UUI command: 0: Tromboning is not performed. 1: Two 3G-324M ports are tromboned together when a terminal sends a UUI command. Only applies if both ports negotiated H.263 video codecs.
videoWithAL2	int	1	Turns on video with AL2 support in the terminal capabilities set: 0: Does not support video with AL2. Use video with AL3. 1: Supports video with AL2.

Configuring the Video Messaging Server Interface

Create a configuration file (*srv.cfg*) to set up the Video Messaging Server Interface. Each line of the configuration file must specify a different parameter value. The format of a line is:

<Parameter> = <Value>

Any text on a line appearing after a hash character (#) is ignored as a comment.

The following tables describe the types of parameters in the *srv.cfg* file:

- IP interface
- System
- Play and record
- Silence detection
- Audio and video codec
- 3GP

IP interface parameters

The parameters in the following table define IP interface settings for the Video Messaging Server Interface:

Parameter	Type	Default	Description
destIPAddr_audio	string	10.1.8.94	IP address of the system (3G-324M Interface) receiving the audio RTP stream from the server.
destIPAddr_video	string	10.1.8.94	IP address of the system (3G-324M Interface) receiving the video RTP stream from the server.
destPort_audio	DWORD	1000	UDP port on the remote system (3G-324M Interface) receiving the audio RTP stream.
destPort_video	DWORD	2000	UDP port on the remote system (3G-324M Interface) receiving the video RTP stream.
frame_quota	int	2	Number of audio frames assembled in each packet.
srcIPAddr_audio	string	10.1.8.94	IP address of the server sending or receiving the audio RTP stream.
srcIPAddr_video	string	10.1.8.94	IP address of the server sending or receiving the video RTP stream.
srcPort_audio	DWORD	3000	UDP port on the server receiving the audio RTP stream from the 3G-324M Interface.
srcPort_video	DWORD	4000	UDP port on the server receiving the video RTP stream from the 3G-324M Interface.
useIPv6	int	0	IP network type for the server: 0: IPv4 1: IPv6

System parameters

The parameters in the following table describe the system parameters for the Video Messaging Server Interface:

Parameter	Type	Default	Description
enableAVsync	int	0	Enables or disables audio and video RTP endpoints to perform incoming skew calculations: 0: Disables RTP endpoints from performing incoming skew calculations. 1: Enables endpoints to perform incoming skew calculations.
nBoardType	int	6000	CG board type (6000, 6060, 6500, 6565).
nPorts	init	1	Number of ports to run.
vidSkewTime	int	0	Video skew (in ms) that is set by the application and sent to the board to adjust video skew with respect to audio. A positive value indicates that the video lags audio. A negative value indicates that the video leads audio. The maximum value is 4095.

Play and record parameters

The parameters in the following table define play and record audio and video settings for the Video Messaging Server Interface:

Parameter	Type	Default	Description
audio_buffer_size	int	4000000	Maximum audio file size (in bytes) that can be recorded.
AutoPlay	DWORD	0	Enables or disables an immediate play after completion of H.245 negotiation. 0: Disables immediate play. 1: Enables immediate play.
AutoPlayList	int	0	Enables or disables automatic play of files from the list after the call setup. The files are listed in the ListFileName file. 0: Does not start automatically playing files from the list. 1: Automatically starts playing files from the list after the call setup.
AutoPlayRec	int	0	Enables or disables simultaneous play/record after call setup. 0: Does not start simultaneous play/record after call setup. 1: Starts simultaneous play/record after call setup. This setting automatically affects the AutoRecord, AutoPlay, and AutoPlayList parameters when they are set to 0.
AutoRecord	DWORD	0	Enables or disables immediate start of recording after completion of H.245 negotiation. 0: Does not automatically start recording. 1: Starts recording immediately.

Parameter	Type	Default	Description
enable_Iframe	DWORD	1	Determines when recording on audio and video channels begins. 0: Recording begins immediately after the ADI multimedia recording function is called. 1: Recording begins upon receipt of next I-frame.
ListFileName	int	autoplay.lst	Name of file that contains the list of files to start playing after the call is set up. Only used if AutoPlayList = 1.
Maxtime	DWORD	0	Maximum allowed time for recording (in hundreds of milliseconds). If Maxtime is exceeded, recording is immediately terminated and all files are closed. 0: Disables this time limit.
PartialBuffers	int	0	Specifies either synchronous or asynchronous play/record. 0: Synchronous play/record. 1: Asynchronous play/record.
video_buffer_size	int	4000000	Maximum video file size (in bytes) that can be recorded.

Silence detection parameters

The parameters in the following table define silence detection settings for the Video Messaging Server Interface:

Parameter	Type	Default	Description
audioStop_InitTimeout	DWORD	0	Enables or disables automatic stop of audio and video recording when audio silence (for example, no voice activity) is detected at the start of the audio recording. 0: Audio and video recording is not automatically stopped. 1-65535: Number of milliseconds of silence needed to trigger the stop. This option has no effect on playback.
audioStop_SilenceLevel	DWORD	-45	Audio level (in dBm) required to trigger audioStop_initTimeout or audioStop_SilenceTimeout. Range: -51 to -15 dBm
audioStop_SilenceTimeout	DWORD	0	Enables or disables automatic stop of audio and video recording when audio silence (for example, no voice activity) is detected while recording audio. 0: Audio and video recording is not automatically stopped. 1-65535: Number of milliseconds of silence needed to trigger the stop. This option has no effect on playback.
native_silence_detect	DWORD	0	Enables or disables silence detection: 0: Disables silence detection. 1: Enables silence detection.

Parameter	Type	Default	Description
videoStop_InitTimeout	DWORD	0	<p>Enables or disables automatic stop of audio and video recording when the video signal is not present at the start of a recording session (for example, if the RTP packet stream is not transmitting).</p> <p>0: Audio and video recording is not automatically stopped. 1-65535: Number of milliseconds of no video needed to trigger the stop.</p> <p>This option has no effect on playback.</p>
videoStop_NosignalTimeout	DWORD	0	<p>Enables or disables automatic stop of audio and video recording when the video signal is not present during a recording session (for example, if the RTP packet stream stops transmitting).</p> <p>0: Audio and video recording is not automatically stopped. 1-65535: Number of milliseconds of no video needed to trigger the stop.</p> <p>This option has no effect on playback.</p>

Audio and video codec parameters

The parameters in the following table define audio and video codec settings for the Video Messaging Server Interface.

Parameter	Type	Default	Description
Audio_Record_Format1	enum	None	Audio record format: 0: NO_AUDIO 1: N_AMR (passthrough or native AMR) 2: N_G723 (passthrough or native G.723) 3: F_G711U 4: F_G711A 5: F_G723_5 6: F_G723_6 7: F_GSM_FR 8: F_AMR_475 9: F_AMR_515 10: F_AMR_590 11: F_AMR_670 12: F_AMR_740 13: F_AMR_795 14: F_AMR_102 15: F_AMR_122
Audio_RTP_Format	enum	11 (AMR_122)	Audio RTP format (only applies if audio transcoding is performed): 0: NO_AUDIO_RTP 1: G723_5 2: G723_6 3: G711 4: AMR_475 5: AMR_515 6: AMR_590 7: AMR_670 8: AMR_740 9: AMR_795 10: AMR_102 11: AMR_122
Video_Format	enum	1	Video format: 0: NO_VIDEO Non-zero: Video record format is based on the video codec negotiated in 3G-324M.

3GP parameters

The parameters in the following table define 3GP settings for the Video Messaging Server Interface.

Parameter	Type	Default	Description
3gpDirectWrite	unsigned long	0	Enables the 3GP library to wait until mmCloseFile is invoked before writing 3GP data to disk: 0: 3GP library waits until mmCloseFile is invoked before writing 3GP data to disk. 1: 3GP library writes data when mmWriteStream is invoked.
3gpFileName	string	record.3gp	3GP file name.
3gpFileName_play	string	play.3gp	3GP file name containing audio and video media used for play during simultaneous play/record operation.
createHintTracks	int	0	Enables or disables whether the 3GP library should create hint tracks for each track in the 3GP file. 0: Disables the creation of hint tracks. 1: Enables the creation of hint tracks.
getSDP	int	0	Enables or disables the retrieval of SDP-related parameters from 3GP file to be played. Data will be written to an output SDP file using outputSDPFileName . 0: Disables retrieval/storage of SDP information. 1: Enables retrieval/storage of SDP information.
inputSDPFileName	string	input.sdp	Input SDP file for setting of SDP-related parameters in recorded 3GP file. Enable the capability using setSDP .
outputSDPFileName	string	output.sdp	Output SDP file for storage of SDP-related parameters from 3GP file to be played. Enable the capability using getSDP .
seekTime	int	0	Uses mmSeekToTime to seek to the specified time during a play operation. The seek time is in ms and is relative to the start of the 3GP file. Valid values lie between the first sync point time and the last sync point time. A value of 0 disables seeking.
setSDP	int	0	Enables or disables the setting of SDP-related parameters in a recorded 3GP file, based on information in an input SDP file using inputSDPFilename . 0: Disables setting SDP information. 1: Enables setting SDP information.

Parameter	Type	Default	Description
skewCorrection	int	0	<p>Enables the 3GP library to perform skew correction for an audio stream during a record operation.</p> <p>The application sets the skew correction time by using mmSetSkewCorrection. The 3GP library adjusts audio skew with respect to video by inserting or removing audio frames.</p> <p>skewCorrection is in ms. The range is -4095 to 4095 ms.</p> <p>0: Disables skew correction. A positive value indicates that audio leads video. A negative value indicates that audio lags video.</p>
useHintTracks	int	0	<p>Enables or disables the 3GP library from using hint tracks for each track in the 3GP file.</p> <p>0: Disables the use of hint tracks. 1: Enables the use of hint tracks.</p>
writeSyncPoints	int	0	<p>Enables or disables the 3GP library from creating sync points for random access support.</p> <p>0: Disables the writing of sync points as part of the 3GP file. 1: Enables the writing of sync points as part of the 3GP file.</p>

4

Using vmsamp

Running vmsamp

When *vmsamp* is configured to run a single channel, it is a menu-driven interactive program. Each menu is invoked by a single letter.

When you enter a menu command, *vmsamp* either executes the function associated with the menu, or it prompts for additional information. In the latter case, you must enter a valid submenu command (also one letter) to execute the associated function. To get help, enter **b**. To exit from a *vmsamp* session, enter **a**.

For a comprehensive list of *vmsamp* menu commands, see *vmsamp menu commands* on page 26.

To run *vmsamp*, follow these steps:

Step	Description						
1	Navigate to the <code>\nms\vaccess\demos\vmsamp</code> directory.						
2	<p>Enter the following command at the command line:</p> <pre>vmsamp -g gw.cfg -s srv.cfg</pre> <p><i>gw.cfg</i> is the name of the 3G-324M Interface (gateway) configuration file and <i>srv.cfg</i> is the name of the Video Messaging Server (server) configuration file.</p> <p>You can also specify the following options:</p> <table><tr><th>Option</th><th>Description</th></tr><tr><td>-b boardnum</td><td>Board number. Default: 0</td></tr><tr><td>-I trace level</td><td>Application trace level. Default: 0xF Refer to <i>vm_trace.h</i> for possible values.</td></tr></table> <p>If you start <i>vmsamp</i> without any command line parameters, the default parameters apply.</p>	Option	Description	-b boardnum	Board number. Default: 0	-I trace level	Application trace level. Default: 0xF Refer to <i>vm_trace.h</i> for possible values.
Option	Description						
-b boardnum	Board number. Default: 0						
-I trace level	Application trace level. Default: 0xF Refer to <i>vm_trace.h</i> for possible values.						
3	Place a call from the 3G terminal (or the 3G terminal simulator) to the 3G-324M Interface.						
4	Talk to the 3G terminal in front of the camera.						
5	<p>Start audio and video recording on the Video Messaging Server Interface:</p> <ul style="list-style-type: none">• Enter z for play and record commands.• Enter r to start audio and video recording.						
6	<p>Stop audio recording on the Video Messaging Server Interface:</p> <ul style="list-style-type: none">• Enter z for play and record commands.• Enter u to stop audio recording. Video recording automatically stops at the same time.						

Step	Description
7	Play the recorded 3GP file from the Video Messaging Server Interface: <ul style="list-style-type: none">• Enter z for play and record commands.• Enter s to play a recorded file.
8	Ensure that both audio and video were played back to the 3G terminal.

Note: Ensure that the gateway configuration file (*gw.cfg*) and the server configuration file (*srv.cfg*) are accurate and that you choose the menu options carefully. The *vmsamp* demonstration program provides minimal protection against invalid entries, range checking, or attempts to use invalid modes.

vmsamp menu commands

vmsamp contains the following menus:

- Endpoint
- H.324M Middleware
- Call control
- Play and record
- Jitter buffer
- Random access
- Video transcoding
- Video tromboning
- Miscellaneous

This topic describes each of these menus.

Endpoint menu

The following table describes the *vmsamp* menu commands that execute MSPP endpoint commands and queries:

Menu command	Function	Description
p	Audio pass-through commands	Submenu for the audio pass-through commands: a: Exits from the submenu. b: Reprints the menu. c: Plays RFC 2833 event. d: Updates DTMF mode. e: Sets audio transmit payload ID.
r	Video channel query	Calls mspSendQuery for the video channel (MSP_QRY_JITTER_VIDEO_GET_STATE).
s	AMR bypass endpoint commands	Submenu for the AMR RTP endpoint commands: a: Exits from the submenu. b: Reprints the menu. e: Calls mspEnableEndpoint . f: Calls mspDisableEndpoint . g: Calls mspSendCommand to change the payload ID map (MSP_CMD RTPFDX_MAP). n: Calls mspSendCommand to enable receive skew calculations (MSP_CMD RTPFDX_CALC_SKEW_OFFSET). o: Calls mspSendCommand to disable receive skew calculations (MSP_CMD RTPFDX_CALC_SKEW_OFFSET).
t	AMR bypass endpoint query	Reserved for future use.
u	AMR bypass channel commands	Calls mspSendCommand for the AMR bypass channel to change the jitter depth (MSP_CMD_JITTER_CHG_DEPTH).
v	AMR bypass channel query	Calls mspSendQuery for the AMR bypass channel (MSP_QRY_JITTER_GET_STATE).

H.324M Middleware menu

The following table describes the *vmsamp* menu command that invokes H.324M functions:

Menu command	Description
y	Submenu for the H.324M Middleware menu: a: Exits from the submenu. b: Reprints the menu. c: Calls h324VideoFastUpdate . d: Calls h324RoundTripDelay . e: Calls h324EndSession . f: Calls h324UserIndication (alphanumeric, 128 bytes). g: Calls h324UserIndication (alphanumeric, 512 bytes). h: Calls h324UserIndication (non-standard, 128 bytes). i: Calls h324UserIndication (non-standard, 512 bytes). j: Calls h324UserIndication (signal). l: Calls h324CloseChannel . m: Calls h324LineErrorReporting . n: Calls h324_h223SkewIndication . o: Calls h324VendorIDIndication . p: Calls h324VideoTemporalSpatialTradeoff .

Call control menu

The following table describes the *vmsamp* menu command that invokes call control functions:

Menu command	Description
x	Submenu for the call control menu: a: Exits from the submenu. b: Reprints the menu. c: Calls nccPlaceCall . d: Calls nccDisconnectCall . e: Calls h324CloseChannel , h324EndSession , and nccDisconnectCall .

Play and record menu

The following table describes the *vmsamp* menu command that invokes ADI play and record functions:

Menu command	Description
z	<p>Submenu for the play and record menu:</p> <p>a: Exits from the submenu.</p> <p>b: Reprints the menu.</p> <p>i: Calls adiRecordMMToMemory or adiRecordMMAsync and adiPlayMMFromMemory or adiPlayMMAsync to start simultaneous play/record functionality.</p> <p>l: Calls adiPlayMMFromMemory or adiPlayMMAsync for audio and video multiple times based on the specified list of files.</p> <p>m: Calls adiStopPlaying to stop playing audio and video in simultaneous play/record mode.</p> <p>n: Calls adiStopRecording to stop recording audio and video simultaneous play/record mode.</p> <p>r: Calls adiRecordMMToMemory or adiRecordMMAsync to record audio and video using the default file names.</p> <p>s: Calls adiPlayMMFromMemory for playing audio and video using the default file names.</p> <p>t: Calls adiPlayMMFromMemory or adiPlayMMAsync for audio and video using the specified file name.</p> <p>u: If application is recording audio, calls adiStopRecording to stop recording audio. If application is playing audio, calls adiStopPlaying to stop playing audio.</p> <p>v: If application is recording video, calls adiStopRecording to stop recording video. If application is playing video, calls adiStopPlaying to stop playing video.</p> <p>y: Sets new video jitter buffer latency.</p>

Jitter buffer menu

The following table describes the *vmsamp* menu commands that manage the video jitter buffer:

Menu command	Description
n	Normalizes the video jitter buffer.
g	Purges the video jitter buffer.

Random access menu

The following table describes the menu command that performs random access operations:

Menu command	Description
c	Submenu for the random access menu: a: Exits the submenu. b: Reprints the menu. c: Seeks to a time relative to the start of a file. d - Seeks to a time relative to the current position in a file. e: Seeks to the next sync point. f: Seeks to the previous sync point. g: Prints the sync point table. h: Prints the current time.

Video transcoding menu

The following table describes the *vmsamp* menu command that performs video transcoding operations:

Menu command	Description
0	Submenu for the video transcoder commands: a: Exits from the submenu. b: Reprints the menu. c: Calls trcGetStatus . d: Calls trcResetVTP . e: Calls trcStartVideoChannel . f: Calls trcStopVideoChannel .

Video tromboning menu

The following table describes the *vmsamp* menu commands that perform video tromboning operations:

Command	Description
f	Stops video tromboning.
o	Starts video tromboning.

Miscellaneous menu

The following table describes miscellaneous *vmsamp* menu commands:

Function	Description
a	Quits the application.
b	Reprints the Help menu.
c	Prints the system configuration for the 3G-324M Interface and the Video Messaging Server Interface (IP addresses and port numbers for audio and video).

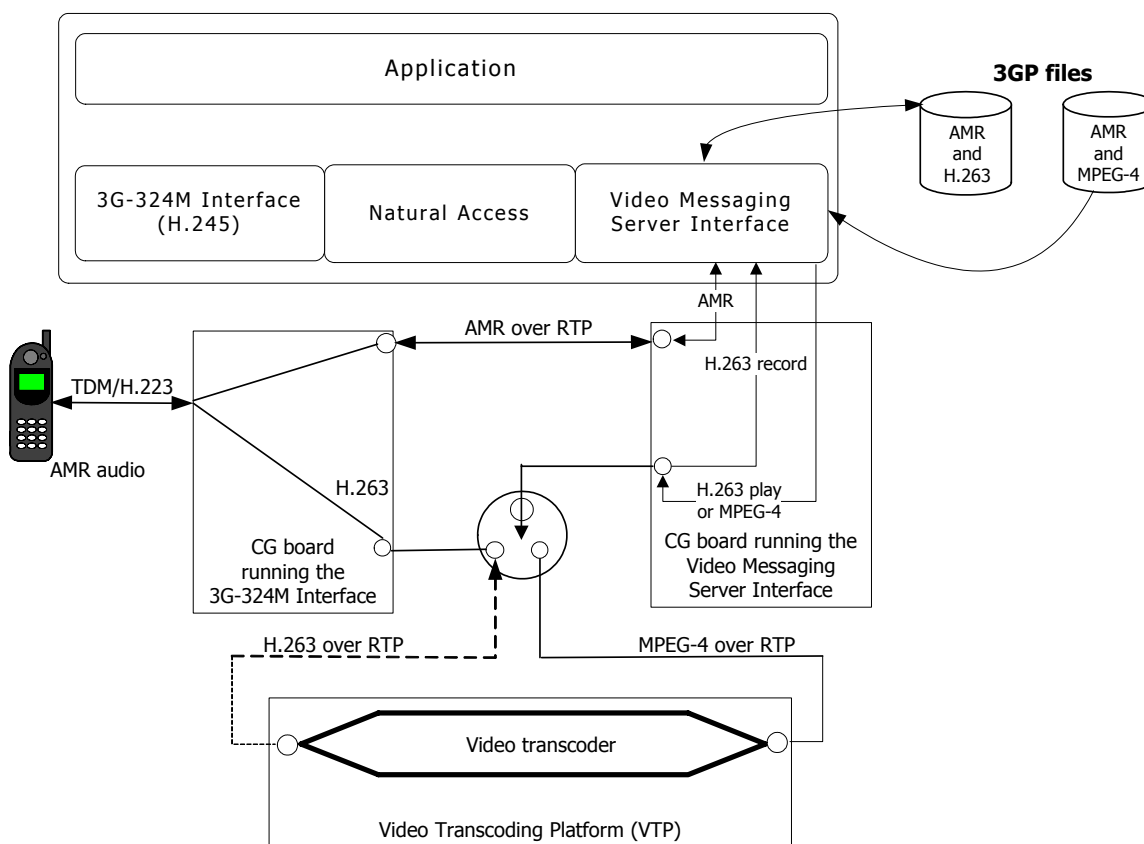
Video transcoding

This topic describes typical messaging application call data flow sequences involving video transcoding. The video transcoder is available separately from NMS.

vmsamp demonstrates the following functions:

- Play and record video and audio natively (no transcoding) using the video format selected during H.245 negotiation with the 3G mobile terminal.
- Real-time video transcoding during playing of the messages whose video format does not match the one selected during H.245 negotiation with the calling terminal.

The following illustration shows the media flow in a video mail system when H.263 baseline is the negotiated format in the 3G-324M connection:



All MPEG-4 video content played to the terminal must go through the video transcoder so that it can be rendered by the terminal. H.263 messages can be passed from the Video Messaging Server Interface to the 3G-324M Interface directly.

The application is responsible for configuring the server's RTP endpoint correctly so it sends RTP packets from the Video Messaging Server Interface either:

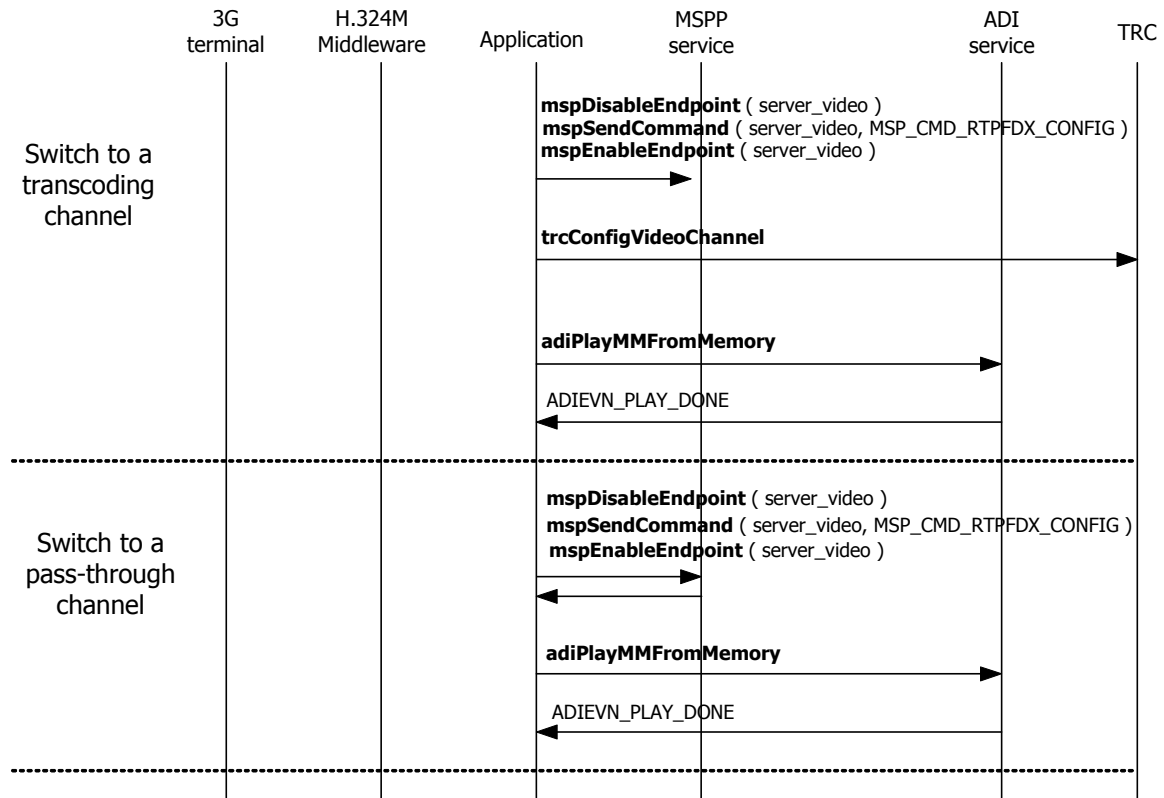
- Through the video transcoder platform, which then relays the packets to the 3G-324M Interface, or
- Directly to the 3G-324M Interface (internally on the CG board).

For recording, H.263 video is passed directly from the 3G-324M Interface to the Video Messaging Server Interface over RTP for 3GP storage.

Switching between video pass-through and video transcoding

In the call-connected state, the application reconfigures the destination IP and port addresses of the server video endpoint to match the desired destination platform of either the 3G-324M Interface (gateway) or video transcoding.

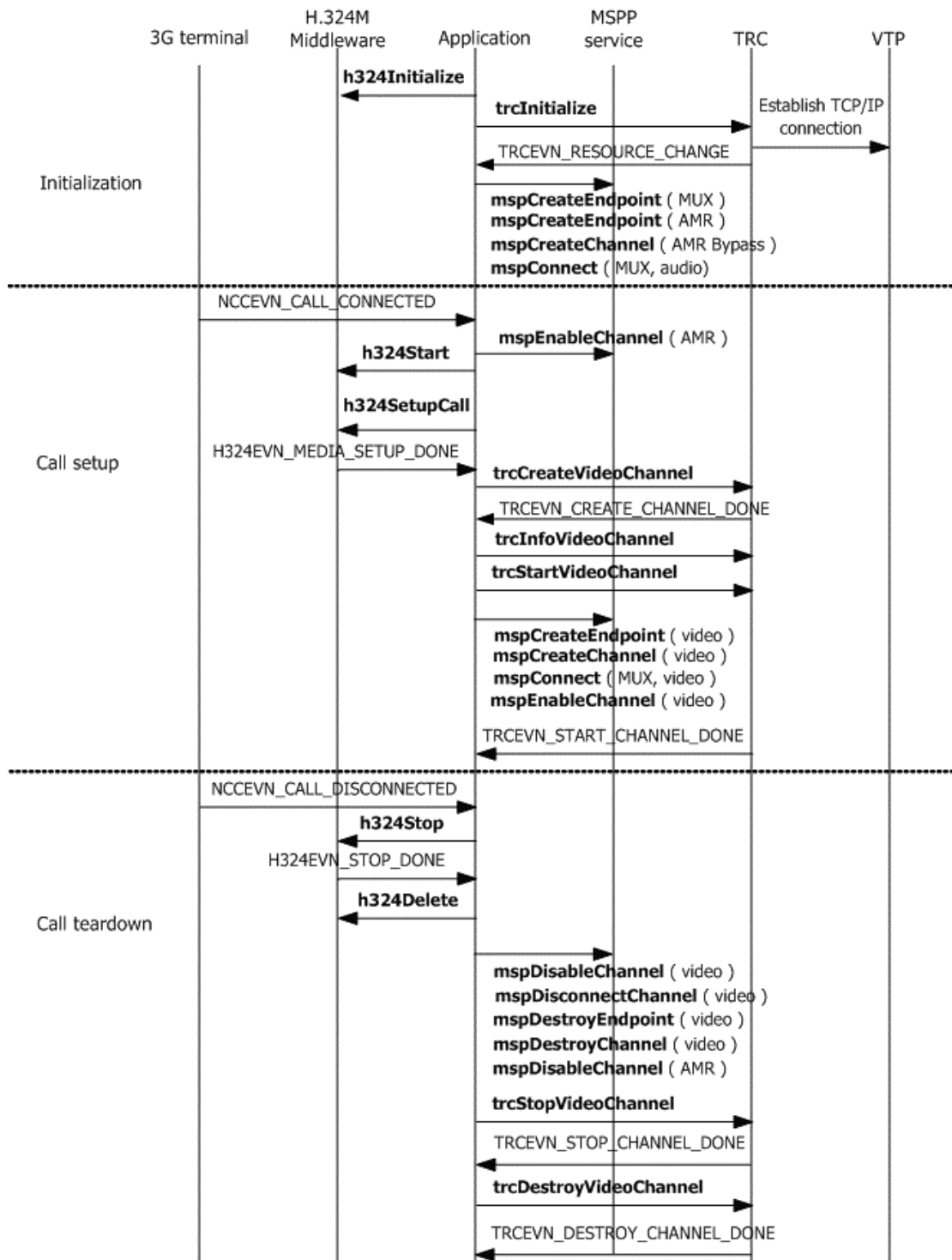
The application also requires an I-frame to be generated by the video transcoder whenever it switches the server to play through the transcoder:



Setting up and tearing down a call

The following call flow describes the application logic for setting up and tearing down an H.324M video call that involves video transcoding.

Note: The video transcoding channel can be created and started at any point after the video codec is selected on the 3G-324M Interface.



Video tromboning

Video tromboning can be used by a variety of service applications to interconnect two RTP streams originating from 3G-324M terminals that have similar audio and video characteristics. Supported characteristics include:

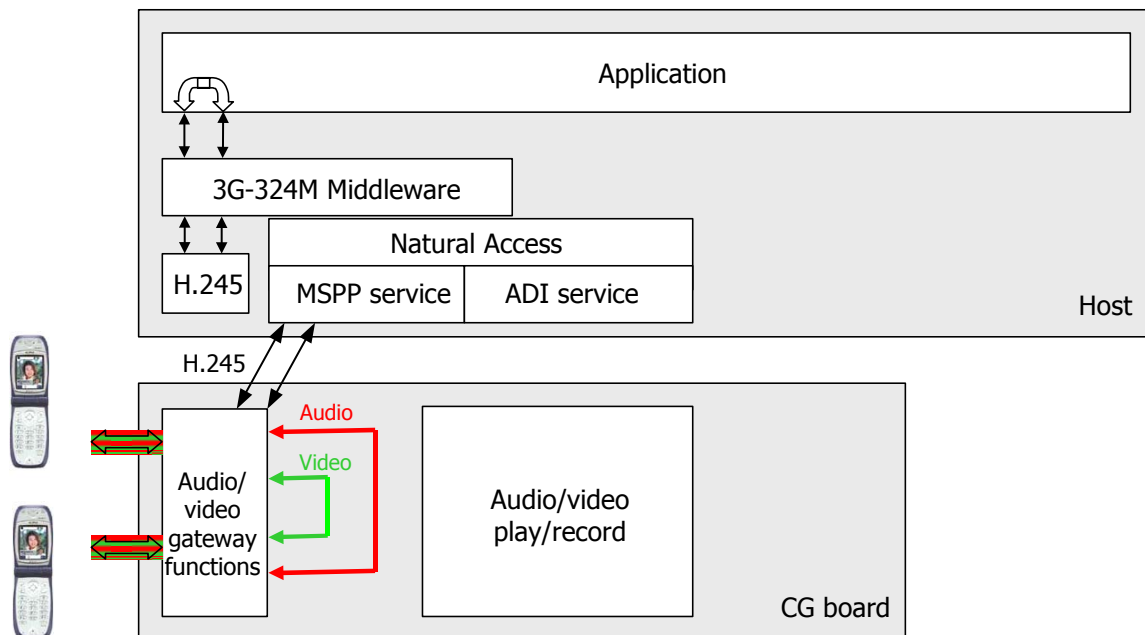
Characteristic	Description
Video	H.263 baseline level 10
Audio	AMR
General	Bit rates consistent with the 3G-324M Interface specification

This topic describes the typical API call flows required to trombone together two 3G-324M calls.

vmsamp demonstrates how to perform the following tasks:

- Trombone together two calls handled by *vmsamp*.
- Switch back to video messaging functions applied to both calls.

The following illustration shows the video tromboning configuration data flow:



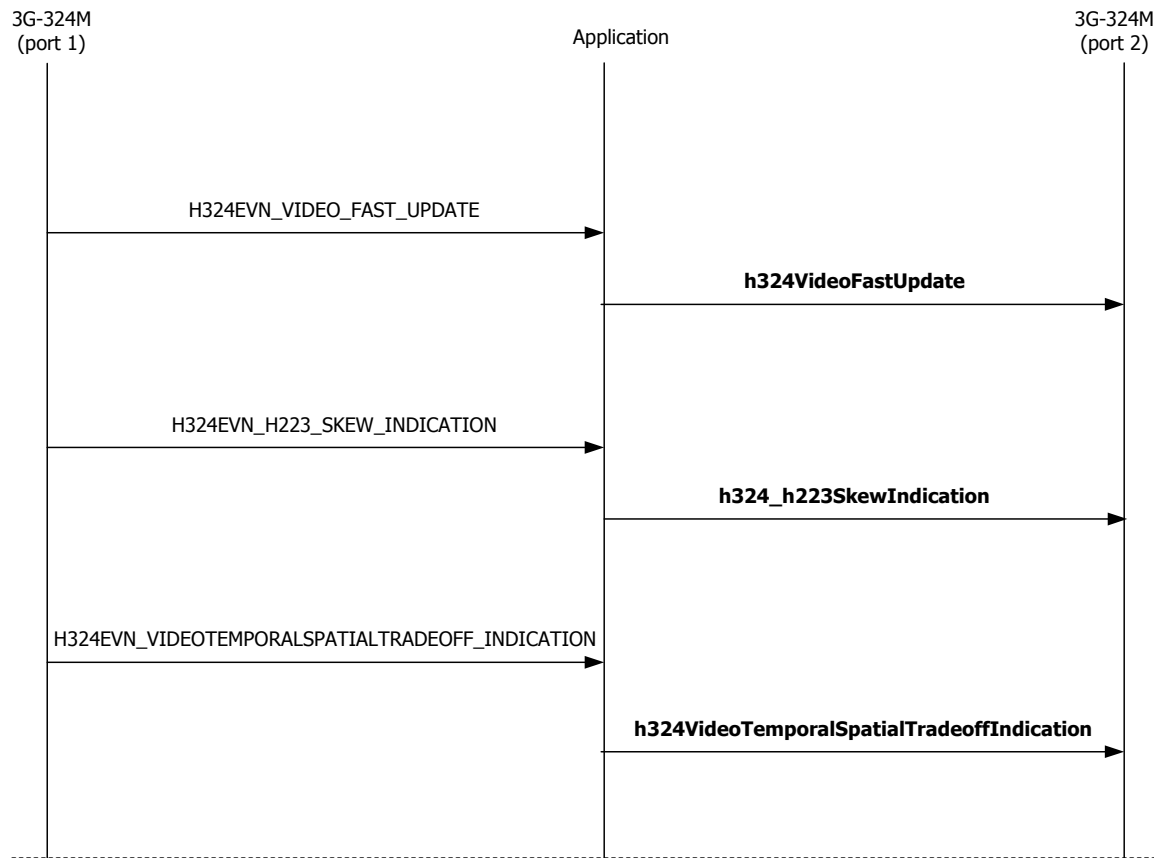
H.245 relay

The application is responsible for relaying H.245 messages, such as a FastVideoUpdate message, that are exchanged between both tromboned terminals.

Note: If the H.245 messages sent by the terminal during the initial H.245 negotiation or messaging state are significant for the tromboned terminal (such as a SkewIndication message), it is the application's responsibility to store and relay these messages to the partner terminal when the tromboning phase is started.

To relay H.245 messages between tromboned ports, the application must use the 3G-324M events and functions.

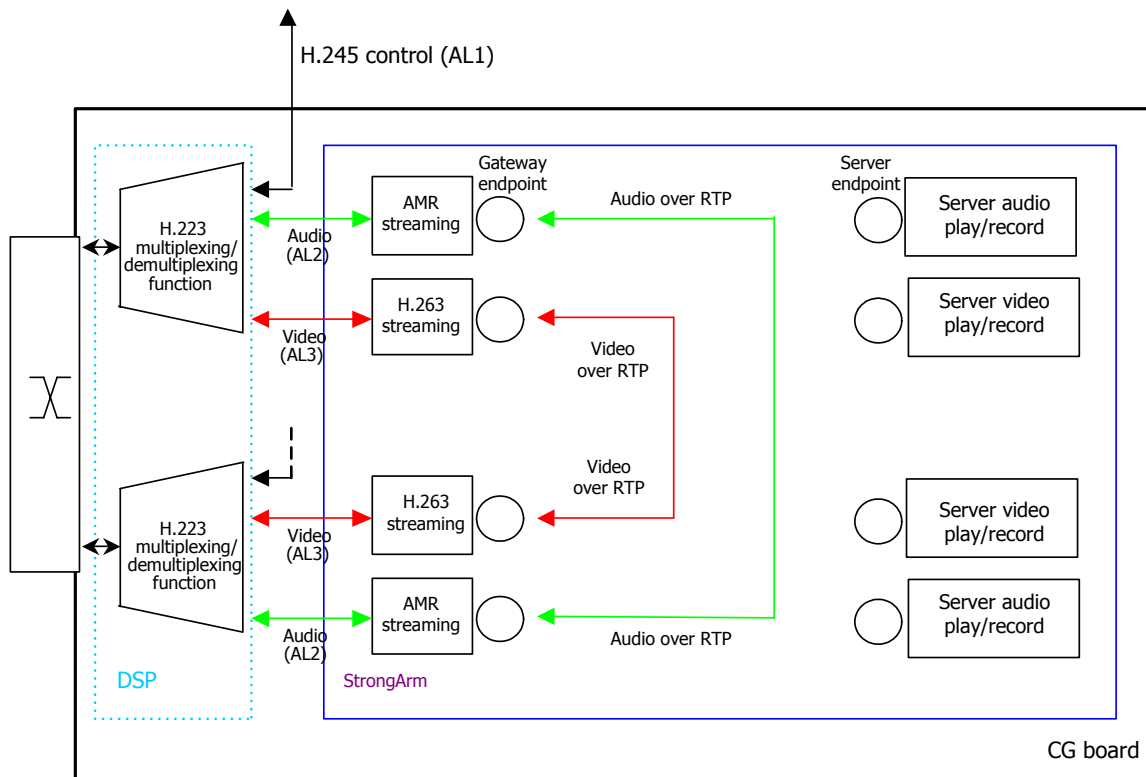
The following illustration shows the API call flow of a typical H.245 message that can be relayed by the application during the tromboning state:



RTP endpoint control

The application is responsible for performing switching at the RTP level required to bridge together audio and video streams from both ports. The application must control RTP endpoints to select destination, optimize video RTP packetization, and ensure a smooth transition when changing video source.

The following illustration shows the different board components and media flows involved in the switching process:



The MSPP service can reconfigure the gateway RTP endpoint destination at any point during the call, which provides media switching capability between the messaging state and the tromboning state. For more information, refer to the *MSPP Service Developer's Reference Manual*.

To trombone two calls together follow these steps:

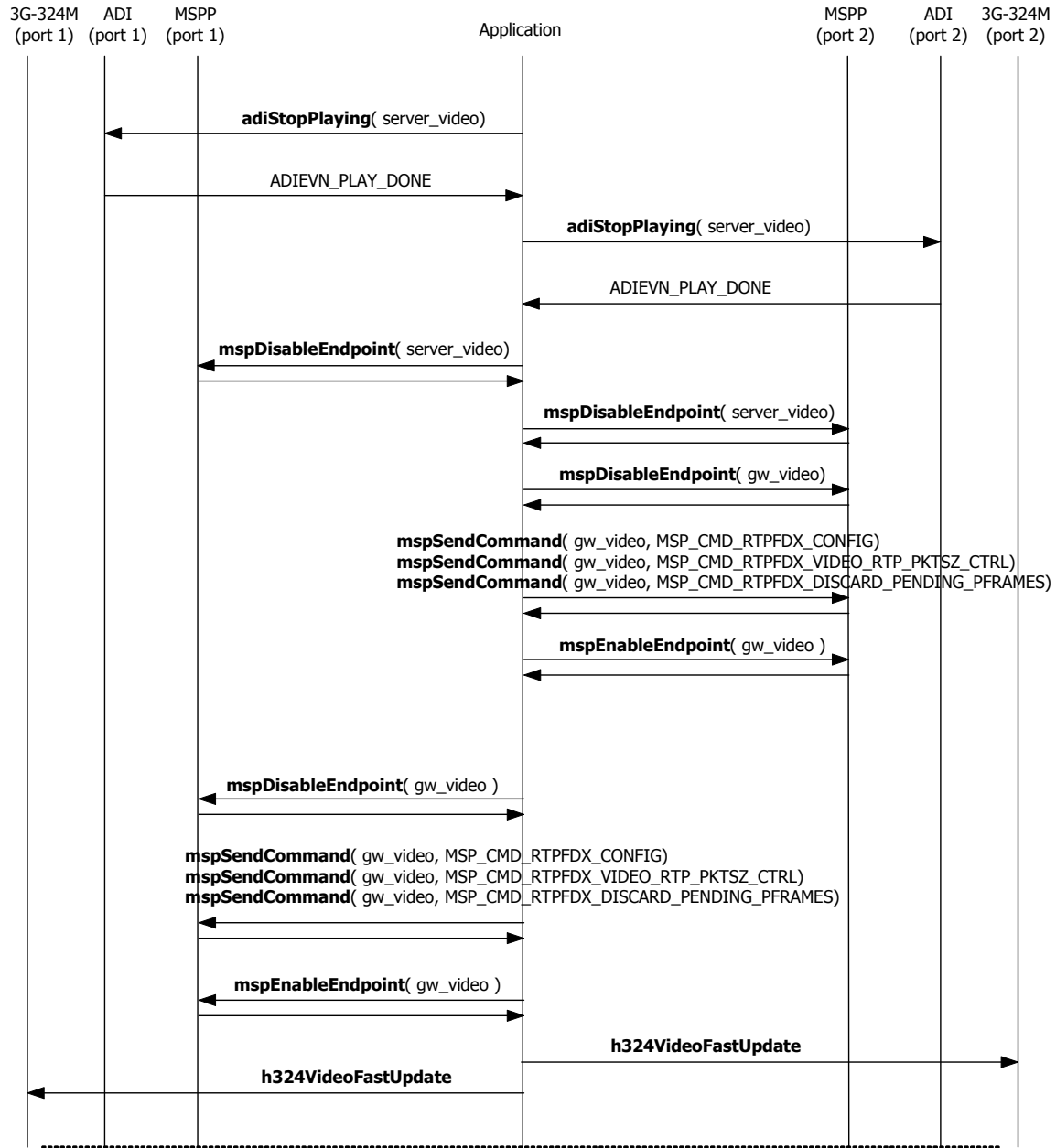
Step	Action
1	Stop the play video function on both server ports and wait for completion (ADIEVN_PLAY_DONE).
2	Disable both server video endpoints to prevent conflicts with gateway endpoints that have the same destination address.
3	Disable the gateway video RTP endpoint of the two tromboned calls.
4	Set the desired RTPAggregationThreshold and MaxRTPPacketSize on each gateway endpoint.
5	Set the gateway endpoint to drop video until an I-frame is detected.
6	Change the destination IP address and UDP port numbers of both gateway video RTP endpoints to match the tromboned partner gateway video endpoint settings.
7	Enable the gateway video RTP endpoints of the two calls to be tromboned together.
8	Send a FastVideoUpdate to both terminals.

The application must also switch audio RTP endpoint destinations, as follows:

Step	Action
1	Disable both server audio endpoints to prevent conflicts with gateway endpoints that have the same destination address.
2	Disable the gateway audio RTP endpoint of the two calls to be tromboned together.
3	Change the destination IP address and UDP port numbers of both gateway audio RTP endpoints to match the tromboned partner gateway audio endpoint settings.
4	Enable the gateway audio RTP endpoint of the two calls to be tromboned together.

Bridging video media streams

The following illustration shows the API call flow required to bridge together video media streams of two different 3G-324M ports:



Bridging audio media streams

The following illustration shows the API call flow required to bridge together audio media streams of two different 3G-324M ports:



Switching back tromboned calls

To switch the two tromboned calls back to a messaging configuration, follow these steps:

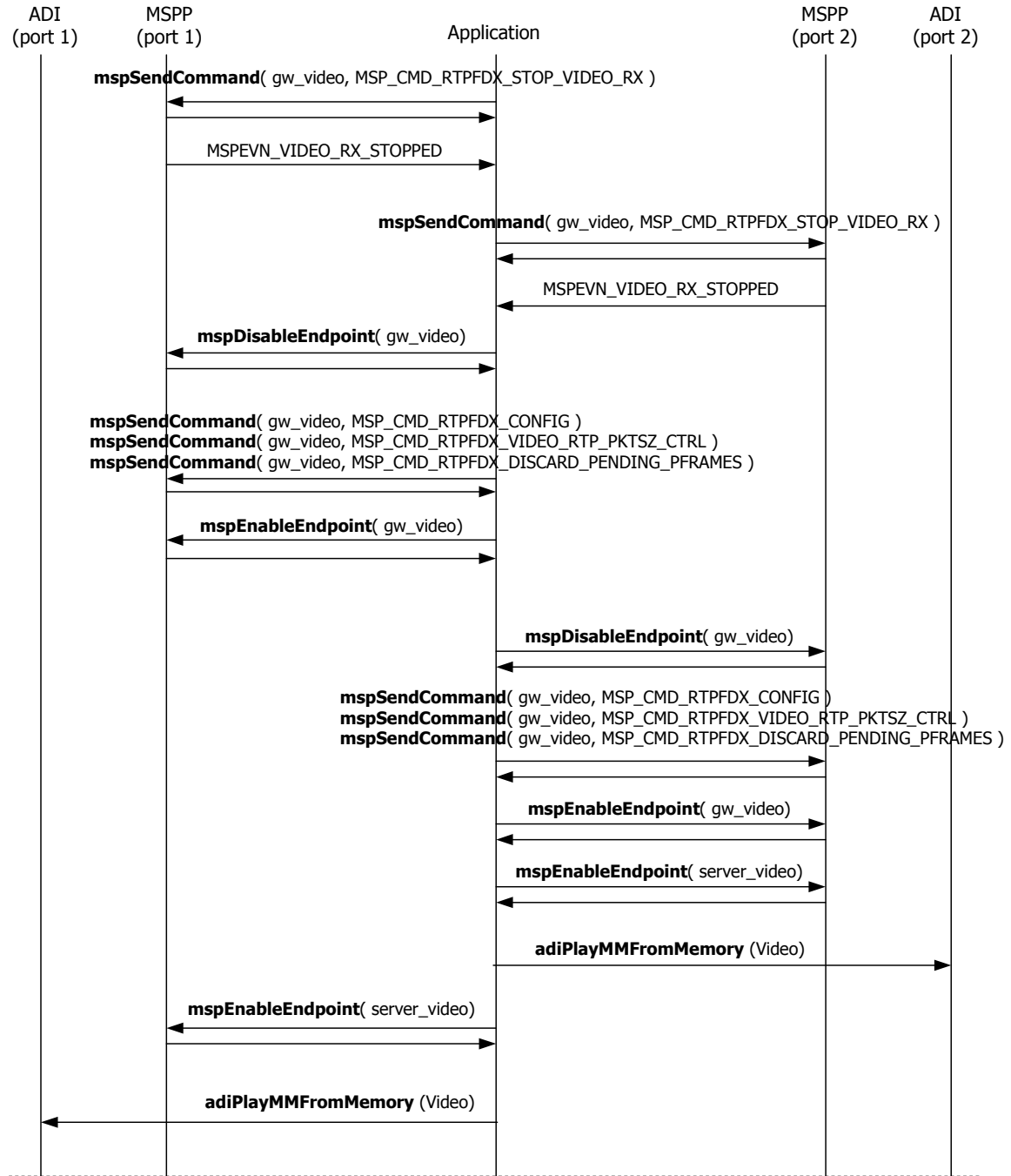
Step	Action
1	Request that both video gateway RTP endpoints stop sending video towards the TDM interface after the end of current frame is detected. The application must receive the corresponding unsolicited event (MSPPEVN_VIDEO_RX_STOPPED) from both endpoints before proceeding.
2	Disable the gateway Video RTP endpoint of the two tromboned calls.
3	Set the desired RTPAggregationThreshold and MaxRTPPacketSize on each gateway endpoint. Default values must be used for both parameters.
4	Change each tromboned gateway video RTP endpoint's IP destination address and UDP destination port number to match server endpoint settings.
5	Enable the gateway video RTP endpoint of the two calls.
6	Enable both server video endpoints.

To switch the audio RTP endpoints back to a messaging configuration, follow these steps:

Step	Action
1	Disable the gateway audio RTP endpoint of the two tromboned calls.
2	Change each tromboned gateway audio RTP endpoint's IP destination address and UDP destination port number to match server endpoint settings.
3	Enable the gateway audio RTP endpoint of the two calls.
4	Enable both server audio endpoints.

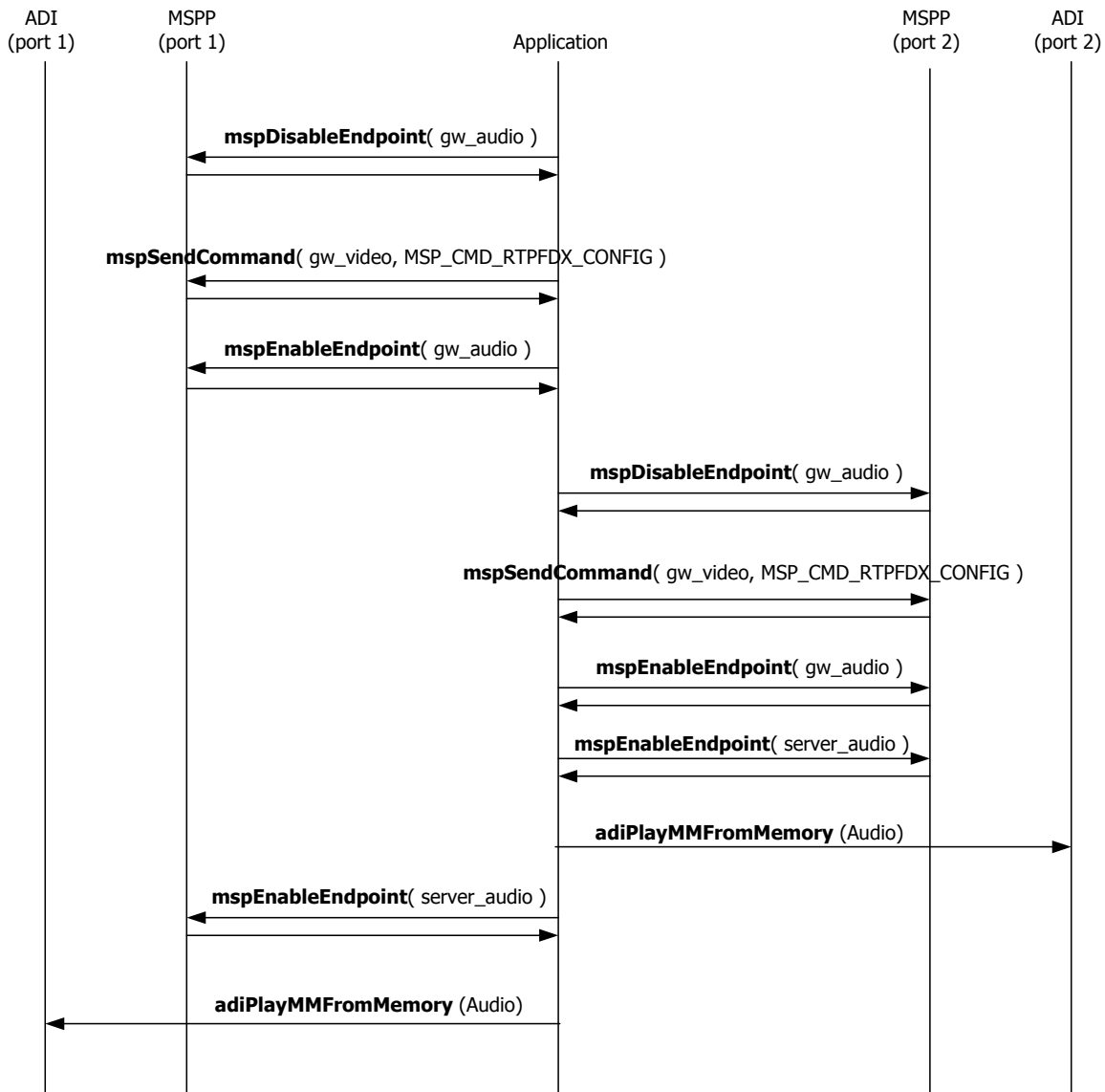
Switching back video endpoints

The following illustration shows the API call flow required to switch back video endpoints to the messaging state:



Switching back audio endpoints

The following illustration shows the API call flow required to switch back audio endpoints to the messaging state:



Index

3

3G-324M Interface 7, 8, 12, 16
3GP parameters 23

A

architecture 8

C

call control 16, 28, 30
codec parameters 16, 22
commands 26
configuration files 11, 12, 18

D

document set 9

E

endpoint commands and queries 27
environment 8
error parameters 13

G

gateway configuration file 11, 12, 25
gateway parameters 12
gw.cfg 11, 12, 25

H

H.245 relay 30
H.324M Middleware 8, 28

I

IP interface parameters 13, 18

J

jitter buffer 29

M

menu commands 26
MSPP commands and queries 27

N

Natural Access 8

O

overview 7

P

parameters 12
 3G-324M 16
 3GP 23
 audio and video codec 16, 22
 call setup 16
 error 13
 IP interface 13, 18
 play and record 19
 silence detection 20
 system 13, 19
 TDM interface 15
 Video Messaging Server Interface 18
 video transcoder 15
play 19, 29

R

random access 30
record 19, 29

S

server configuration file 11, 18, 25
server parameters 18
silence detection parameters 20
srv.cfg 11, 18, 25
system parameters 13, 19

T

TDM interface parameters 15
transcoding 7, 15, 30, 31
tromboning 7, 30, 34

V

Video Access 9
video codec parameters 16, 22

Video Messaging Server Interface 7, 8,
18

video transcoder parameters 15

video transcoding 7, 30, 31

video tromboning 7, 30, 34