



# **Dialogic® NaturalAccess™ Video Access Toolkit Video Mail Application Demonstration Program Manual**

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## Revision history

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Refer to [www.dialogic.com](http://www.dialogic.com) for product updates and for information about support policies, warranty information, and service offerings.

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# 1. Introduction

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The *Dialogic® NaturalAccess™ Video Access Toolkit 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 NaturalAccess. It assumes that you are familiar with telephony concepts, NaturalAccess, and the C programming language. If you are not familiar with NaturalAccess, read the *Dialogic® NaturalAccess™ Video Access Toolkit Video Access Overview Manual* to learn about the NaturalAccess features that relate to Video Access before reviewing this manual.

## Terminology

**Note:** The product to which this document pertains is part of the NMS Communications Platforms business that was sold by NMS Communications Corporation (“NMS”) to Dialogic Corporation (“Dialogic”) on December 8, 2008. Accordingly, certain terminology relating to the product has been changed. Below is a table indicating both terminology that was formerly associated with the product, as well as the new terminology by which the product is now known. This document is being published during a transition period; therefore, it may be that some of the former terminology will appear within the document, in which case the former terminology should be equated to the new terminology, and vice versa.

Former terminology	Dialogic terminology
CG 6060 Board	Dialogic® CG 6060 PCI Media Board
CG 6060C Board	Dialogic® CG 6060C CompactPCI Media Board
CG 6565 Board	Dialogic® CG 6565 PCI Media Board
CG 6565C Board	Dialogic® CG 6565C CompactPCI Media Board
CG 6565e Board	Dialogic® CG 6565E PCI Express Media Board
CX 2000 Board	Dialogic® CX 2000 PCI Station Interface Board
CX 2000C Board	Dialogic® CX 2000C CompactPCI Station Interface Board
AG 2000 Board	Dialogic® AG 2000 PCI Media Board
AG 2000C Board	Dialogic® AG 2000C CompactPCI Media Board
AG 2000-BRI Board	Dialogic® AG 2000-BRI Media Board
NMS OAM Service	Dialogic® NaturalAccess™ OAM API
NMS OAM System	Dialogic® NaturalAccess™ OAM System
NMS SNMP	Dialogic® NaturalAccess™ SNMP API
Natural Access	Dialogic® NaturalAccess™ Software
Natural Access Service	Dialogic® NaturalAccess™ Service
Fusion	Dialogic® NaturalAccess™ Fusion™ VoIP API
ADI Service	Dialogic® NaturalAccess™ Alliance Device Interface API

Former terminology	Dialogic terminology
CDI Service	Dialogic® NaturalAccess™ CX Device Interface API
Digital Trunk Monitor Service	Dialogic® NaturalAccess™ Digital Trunk Monitoring API
MSPP Service	Dialogic® NaturalAccess™ Media Stream Protocol Processing API
Natural Call Control Service	Dialogic® NaturalAccess™ NaturalCallControl™ API
NMS GR303 and V5 Libraries	Dialogic® NaturalAccess™ GR303 and V5 Libraries
Point-to-Point Switching Service	Dialogic® NaturalAccess™ Point-to-Point Switching API
Switching Service	Dialogic® NaturalAccess™ Switching Interface API
Voice Message Service	Dialogic® NaturalAccess™ Voice Control Element API
NMS CAS for Natural Call Control	Dialogic® NaturalAccess™ CAS API
NMS ISDN	Dialogic® NaturalAccess™ ISDN API
NMS ISDN for Natural Call Control	Dialogic® NaturalAccess™ ISDN API
NMS ISDN Messaging API	Dialogic® NaturalAccess™ ISDN Messaging API
NMS ISDN Supplementary Services	Dialogic® NaturalAccess™ ISDN API Supplementary Services
NMS ISDN Management API	Dialogic® NaturalAccess™ ISDN Management API
NaturalConference Service	Dialogic® NaturalAccess™ NaturalConference™ API
NaturalFax	Dialogic® NaturalAccess™ NaturalFax™ API
SAI Service	Dialogic® NaturalAccess™ Universal Speech Access API
NMS SIP for Natural Call Control	Dialogic® NaturalAccess™ SIP API
NMS RJ-45 interface	Dialogic® MD1 RJ-45 interface
NMS RJ-21 interface	Dialogic® MD1 RJ-21 interface

<b>Former terminology</b>	<b>Dialogic terminology</b>
NMS Mini RJ-21 interface	Dialogic® MD1 Mini RJ-21 interface
NMS Mini RJ-21 to NMS RJ-21 cable	Dialogic® MD1 Mini RJ-21 to MD1 RJ-21 cable
NMS RJ-45 to two 75 ohm BNC splitter cable	Dialogic® MD1 RJ-45 to two 75 ohm BNC splitter cable
NMS signal entry panel	Dialogic® Signal Entry Panel
Video Access Utilities	Dialogic® NaturalAccess™ Video Access Toolkit Utilities
Video Mail Application Demonstration Program	Dialogic® NaturalAccess™ Video Access Toolkit Video Mail Application Demonstration Program
Video Messaging Server Interface	Dialogic® NaturalAccess™ Video Access Toolkit Video Messaging Server Interface
3G-324M Interface	Dialogic® NaturalAccess™ Video Access Toolkit 3G-324M Interface

## 2. Overview of vmsamp

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### vmsamp overview

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

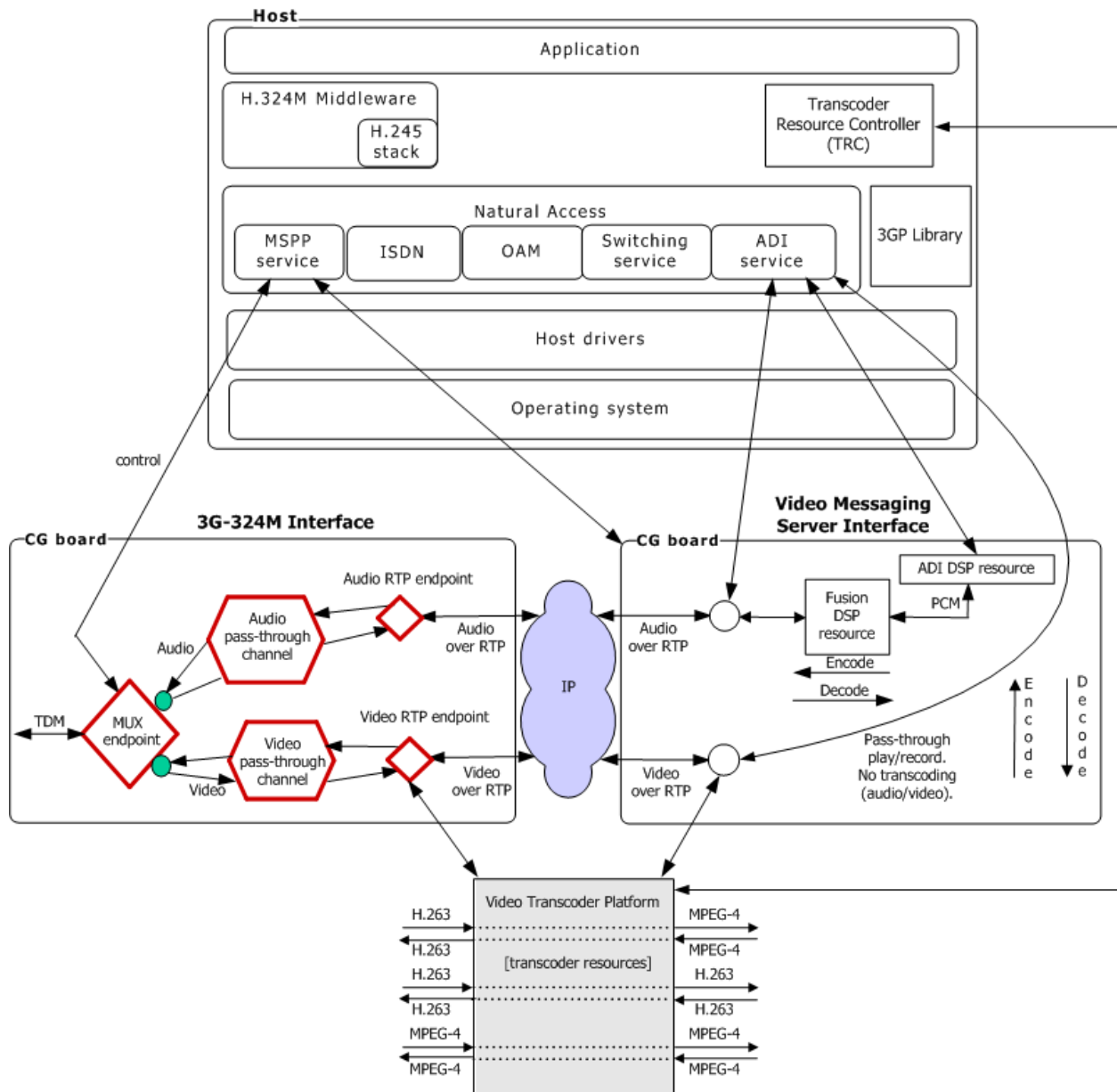
- Receive a call from a 3G terminal to the 3G-324M Interface or place a call to a 3G terminal.
- Record audio and video into a 3GP file using the Video Messaging Server Interface.
- Play 3GP files to the 3G terminal with or without video transcoding.
- Trombone two 3G-324M calls together.

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

## 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>Dialogic® NaturalAccess™ Video Access Toolkit 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>Dialogic® NaturalAccess™ Video Access Toolkit 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>Dialogic® NaturalAccess™ Video Access Toolkit 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>Dialogic® NaturalAccess™ Video Access Toolkit 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.

Manual	Description	Use this manual if...
<p><i>Dialogic®</i>  <i>NaturalAccess™</i>  <i>Video Access</i>  <i>Toolkit Utilities</i>  <i>Manual</i></p>	<p>How to use the Video Access utilities that are available for manipulating 3GP files and monitoring 3G-324M calls.</p>	<p>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.</p> <p>The utilities documented here can also be used to manipulate content created outside of Video Access.</p>

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

## 3. Configuring vmsamp

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### 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](#).

#### 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](#).

## 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

Parameter	Type	Default	Description
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.
outOfBandDCImode	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.

Parameter	Type	Default	Description
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).

Parameter	Type	Default	Description
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.
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.

Parameter	Type	Default	Description
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.

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

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.</p> <p>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.</p> <p>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	<p>Audio record format:</p> <p>0: NO_AUDIO</p> <p>1: N_AMR (passthrough or native AMR)</p> <p>2: N_G723 (passthrough or native G.723)</p> <p>3: F_G711U</p> <p>4: F_G711A</p> <p>5: F_G723_5</p> <p>6: F_G723_6</p> <p>7: F_GSM_FR</p> <p>8: F_AMR_475</p> <p>9: F_AMR_515</p> <p>10: F_AMR_590</p> <p>11: F_AMR_670</p> <p>12: F_AMR_740</p> <p>13: F_AMR_795</p> <p>14: F_AMR_102</p> <p>15: F_AMR_122</p>
Audio_RTP_Format	enum	11 (AMR_122)	<p>Audio RTP format (only applies if audio transcoding is performed):</p> <p>0: NO_AUDIO_RTP</p> <p>1: G723_5</p> <p>2: G723_6</p> <p>3: G711</p> <p>4: AMR_475</p> <p>5: AMR_515</p> <p>6: AMR_590</p> <p>7: AMR_670</p> <p>8: AMR_740</p> <p>9: AMR_795</p> <p>10: AMR_102</p> <p>11: AMR_122</p>

Parameter	Type	Default	Description
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 <b>mmCloseFile</b> is invoked before writing 3GP data to disk: 0: 3GP library waits until <b>mmCloseFile</b> is invoked before writing 3GP data to disk. 1: 3GP library writes data when <b>mmWriteStream</b> 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 <code>outputSDPFileName</code> . 0: Disables retrieval/storage of SDP information. 1: Enables retrieval/storage of SDP information.

Parameter	Type	Default	Description
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 <b>mmSeekToTime</b> 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.
skewCorrection	int	0	Enables the 3GP library to perform skew correction for an audio stream during a record operation.  The application sets the skew correction time by using <b>mmSetSkewCorrection</b> . The 3GP library adjusts audio skew with respect to video by inserting or removing audio frames.  skewCorrection is in ms. The range is -4095 to 4095 ms.  0: Disables skew correction.  A positive value indicates that audio leads video. A negative value indicates that audio lags video.

Parameter	Type	Default	Description
useHintTracks	int	0	Enables or disables the 3GP library from using hint tracks for each track in the 3GP file. 0: Disables the use of hint tracks. 1: Enables the use of hint tracks.
writeSyncPoints	int	0	Enables or disables the 3GP library from creating sync points for random access support. 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.

## 4. Using vmsamp

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### 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.

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 border="1"><thead><tr><th>Option</th><th>Description</th></tr></thead><tbody><tr><td><code>-b <i>boardnum</i></code></td><td>Board number. Default: 0</td></tr><tr><td><code>-l <i>trace level</i></code></td><td>Application trace level. Default: 0xF Refer to <i>vm_trace.h</i> for possible values.</td></tr></tbody></table> <p>If you start <i>vmsamp</i> without any command line parameters, the default parameters apply.</p>	Option	Description	<code>-b <i>boardnum</i></code>	Board number. Default: 0	<code>-l <i>trace level</i></code>	Application trace level. Default: 0xF Refer to <i>vm_trace.h</i> for possible values.
Option	Description						
<code>-b <i>boardnum</i></code>	Board number. Default: 0						
<code>-l <i>trace level</i></code>	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	Start audio and video recording on the Video Messaging Server Interface: <ul style="list-style-type: none"><li>• Enter <b>z</b> for play and record commands.</li><li>• Enter <b>r</b> to start audio and video recording.</li></ul>						

Step	Description
6	Stop audio recording on the Video Messaging Server Interface: <ul style="list-style-type: none"> <li>• Enter <b>z</b> for play and record commands.</li> <li>• Enter <b>u</b> to stop audio recording. Video recording automatically stops at the same time.</li> </ul>
7	Play the recorded 3GP file from the Video Messaging Server Interface: <ul style="list-style-type: none"> <li>• Enter <b>z</b> for play and record commands.</li> <li>• Enter <b>s</b> to play a recorded file.</li> </ul>
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 <b>mspSendQuery</b> 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 <b>mspEnableEndpoint</b> . f: Calls <b>mspDisableEndpoint</b> . g: Calls <b>mspSendCommand</b> to change the payload ID map (MSP_CMD RTPFDX_MAP). n: Calls <b>mspSendCommand</b> to enable receive skew calculations (MSP_CMD RTPFDX_CALC_SKEW_OFFSET). o: Calls <b>mspSendCommand</b> 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 <b>mspSendCommand</b> for the AMR bypass channel to change the jitter depth (MSP_CMD_JITTER_CHG_DEPTH).
v	AMR bypass channel query	Calls <b>mspSendQuery</b> 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 <b>h324VideoFastUpdate</b> . d: Calls <b>h324RoundTripDelay</b> . e: Calls <b>h324EndSession</b> . f: Calls <b>h324UserIndication</b> (alphanumeric, 128 bytes). g: Calls <b>h324UserIndication</b> (alphanumeric, 512 bytes). h: Calls <b>h324UserIndication</b> (non-standard, 128 bytes). i: Calls <b>h324UserIndication</b> (non-standard, 512 bytes). j: Calls <b>h324UserIndication</b> (signal). l: Calls <b>h324CloseChannel</b> . m: Calls <b>h324LineErrorReporting</b> . n: Calls <b>h324_h223SkewIndication</b> . o: Calls <b>h324VendorIDIIndication</b> . p: Calls <b>h324VideoTemporalSpatialTradeoff</b> .

## 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 <b>nccPlaceCall</b> . d: Calls <b>nccDisconnectCall</b> . e: Calls <b>h324CloseChannel</b> , <b>h324EndSession</b> , and <b>nccDisconnectCall</b> .

## 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 <b>adiRecordMMToMemory</b> or <b>adiRecordMMAsync</b> and <b>adiPlayMMFromMemory</b> or <b>adiPlayMMAsync</b> to start simultaneous play/record functionality.</p> <p>l: Calls <b>adiPlayMMFromMemory</b> or <b>adiPlayMMAsync</b> for audio and video multiple times based on the specified list of files.</p> <p>m: Calls <b>adiStopPlaying</b> to stop playing audio and video in simultaneous play/record mode.</p> <p>n: Calls <b>adiStopRecording</b> to stop recording audio and video simultaneous play/record mode.</p> <p>r: Calls <b>adiRecordMMToMemory</b> or <b>adiRecordMMAsync</b> to record audio and video using the default file names.</p> <p>s: Calls <b>adiPlayMMFromMemory</b> for playing audio and video using the default file names.</p> <p>t: Calls <b>adiPlayMMFromMemory</b> or <b>adiPlayMMAsync</b> for audio and video using the specified file name.</p> <p>u: If application is recording audio, calls <b>adiStopRecording</b> to stop recording audio. If application is playing audio, calls <b>adiStopPlaying</b> to stop playing audio.</p> <p>v: If application is recording video, calls <b>adiStopRecording</b> to stop recording video. If application is playing video, calls <b>adiStopPlaying</b> 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 <b>trcGetStatus</b> . d: Calls <b>trcResetVTP</b> . e: Calls <b>trcStartVideoChannel</b> . f: Calls <b>trcStopVideoChannel</b> .

## 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).

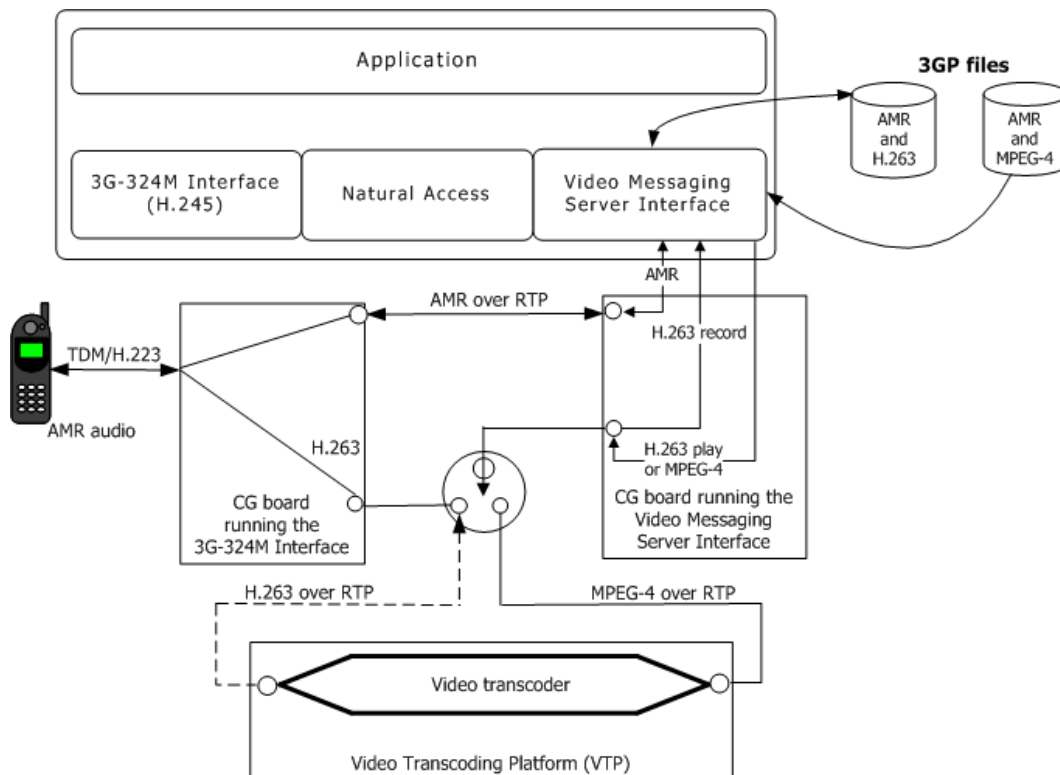
## Video transcoding

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:

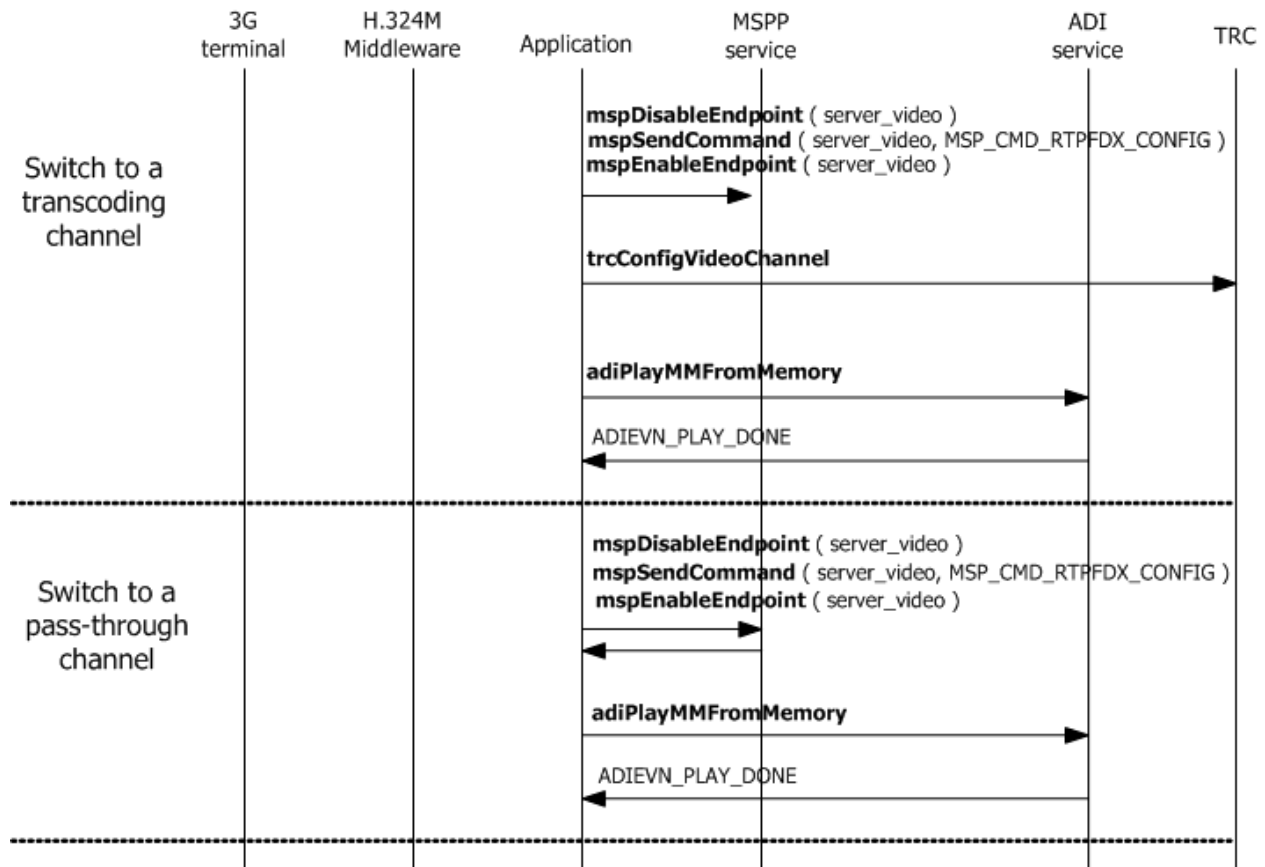
- Through the video transcoder platform, which then relays the packets to the 3G-324M Interface, or
- Directly to the 3G-324M Interface (internally on the CG board).

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.

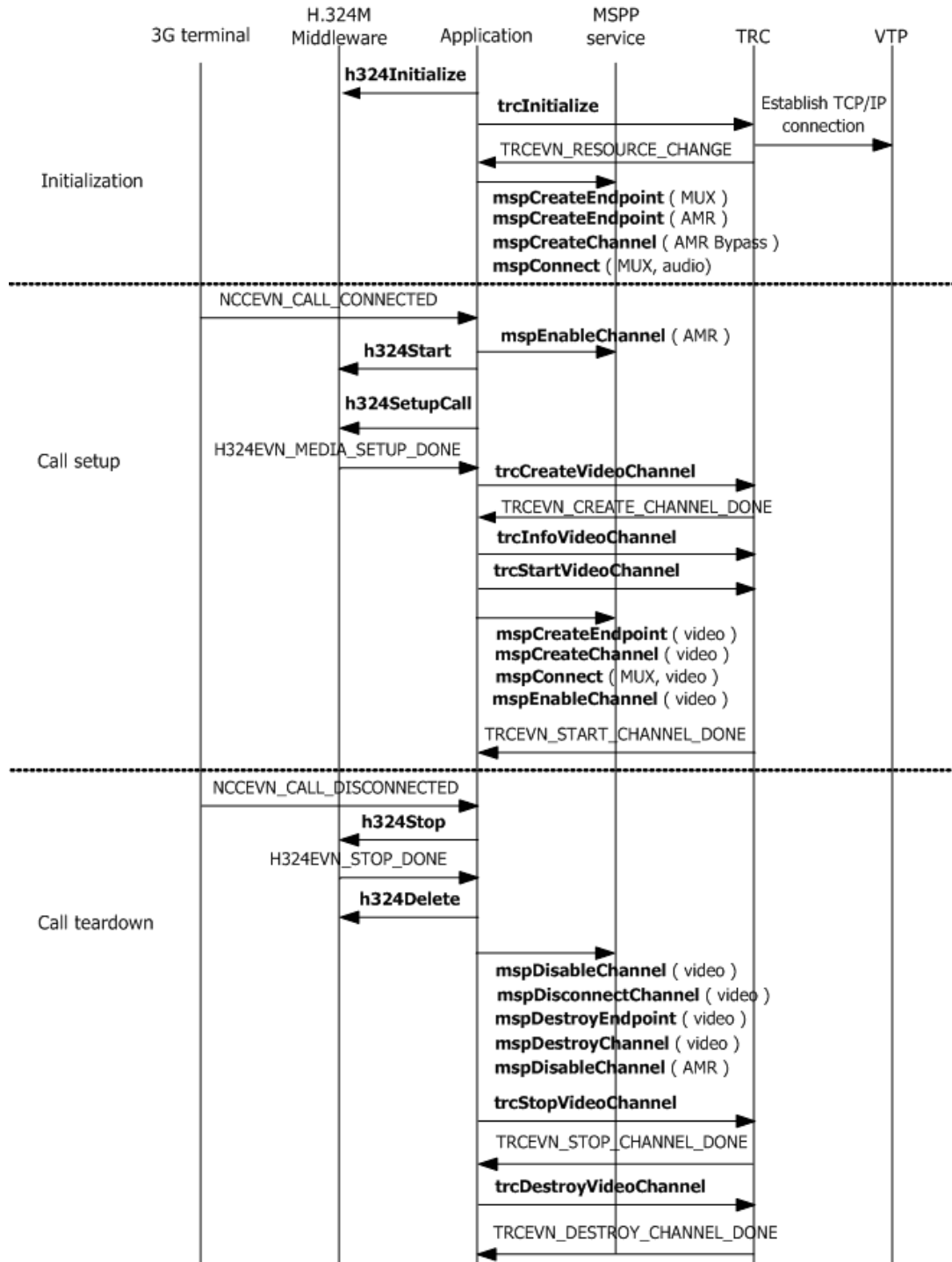
The application also requires an I-frame to be generated by the video transcoder whenever it switches the server to play though the transcoder:



## 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:

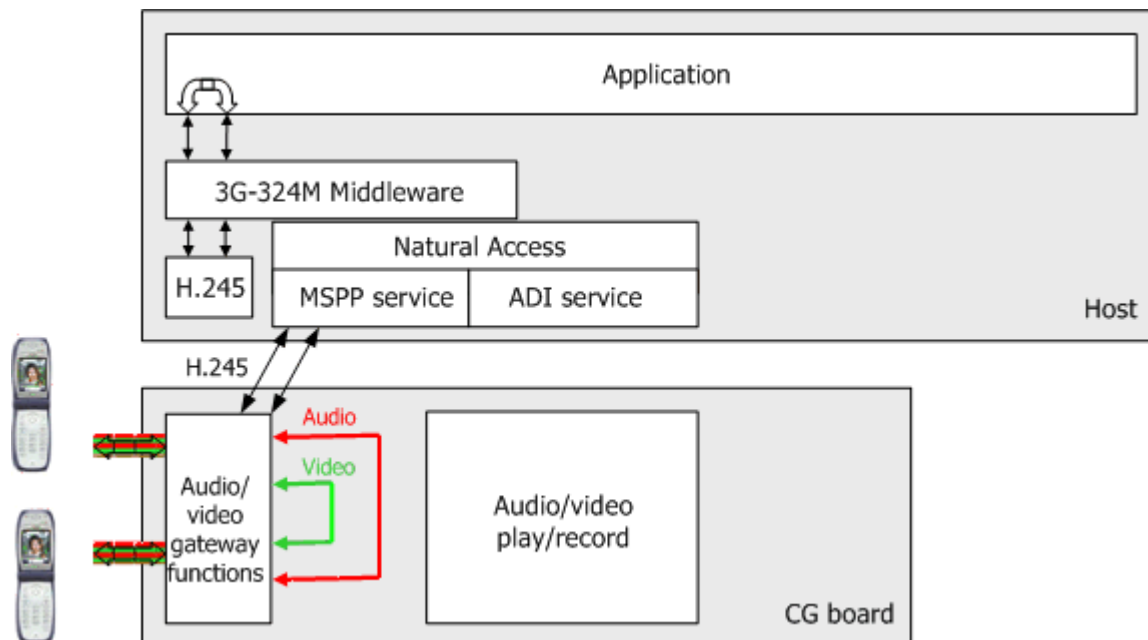
Characteristic	Description
Video	H.263 baseline level 10
Audio	AMR
General	Bit rates consistent with the 3G-324M Interface specification

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:



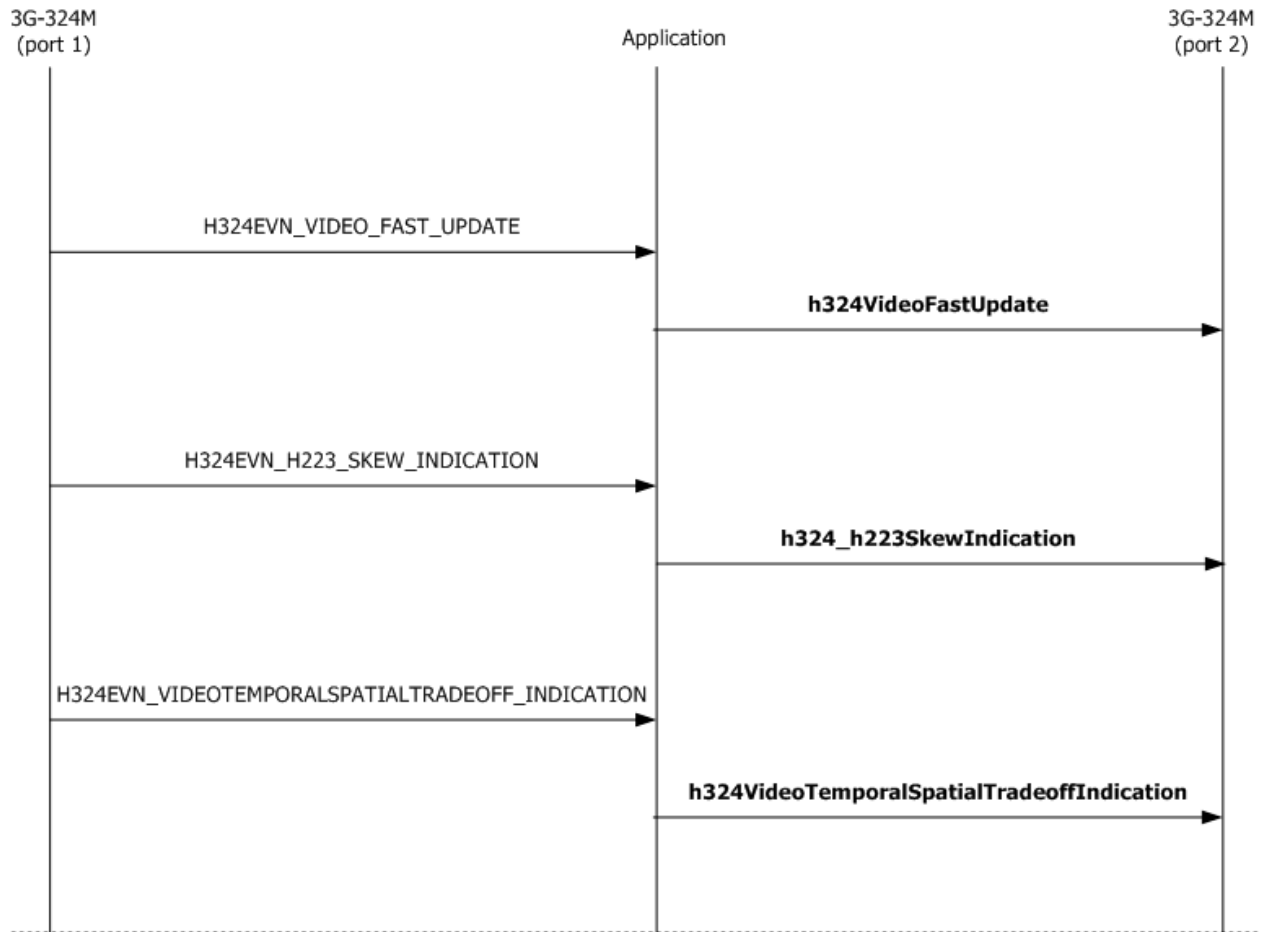
### H.245 relay

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.

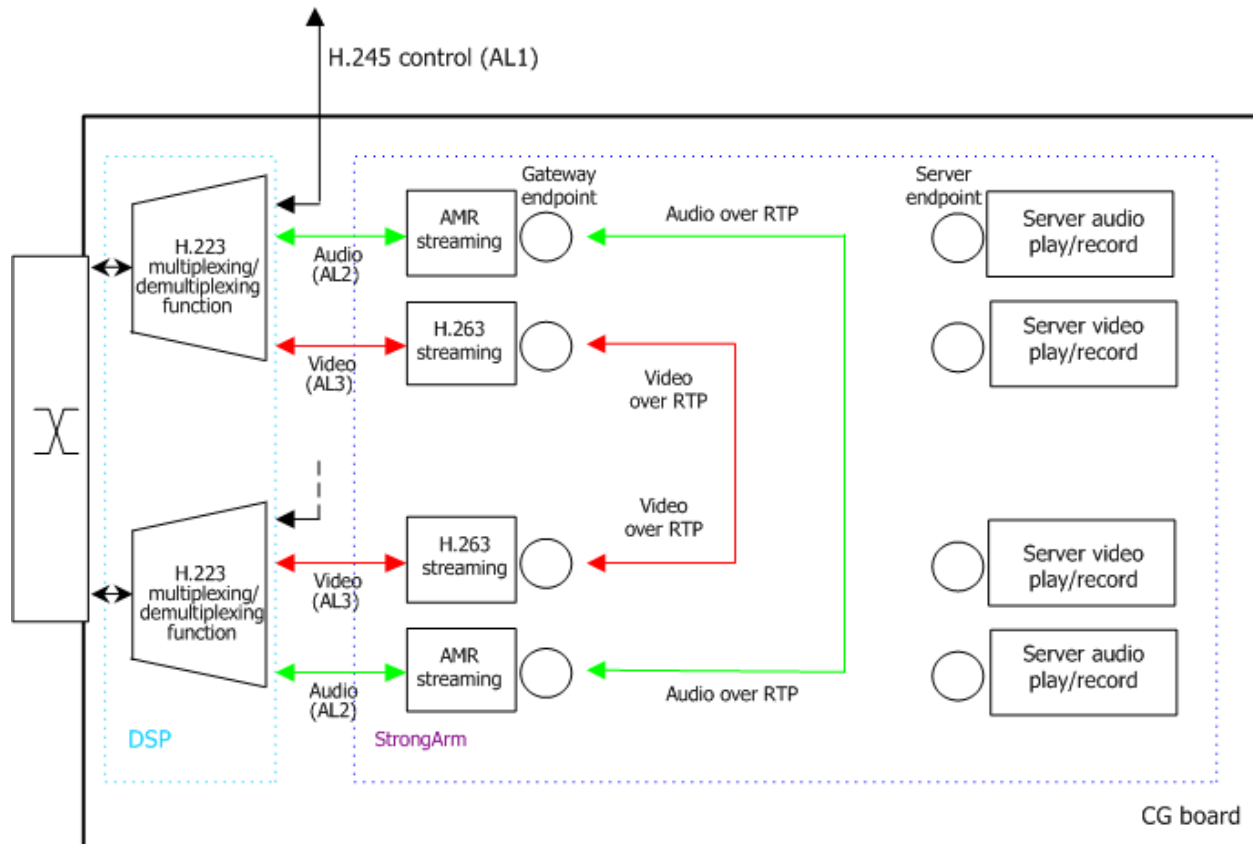
The following illustration shows the API call flow of a typical H.245 message that can be relayed by the application during the tromboning state:



### 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:

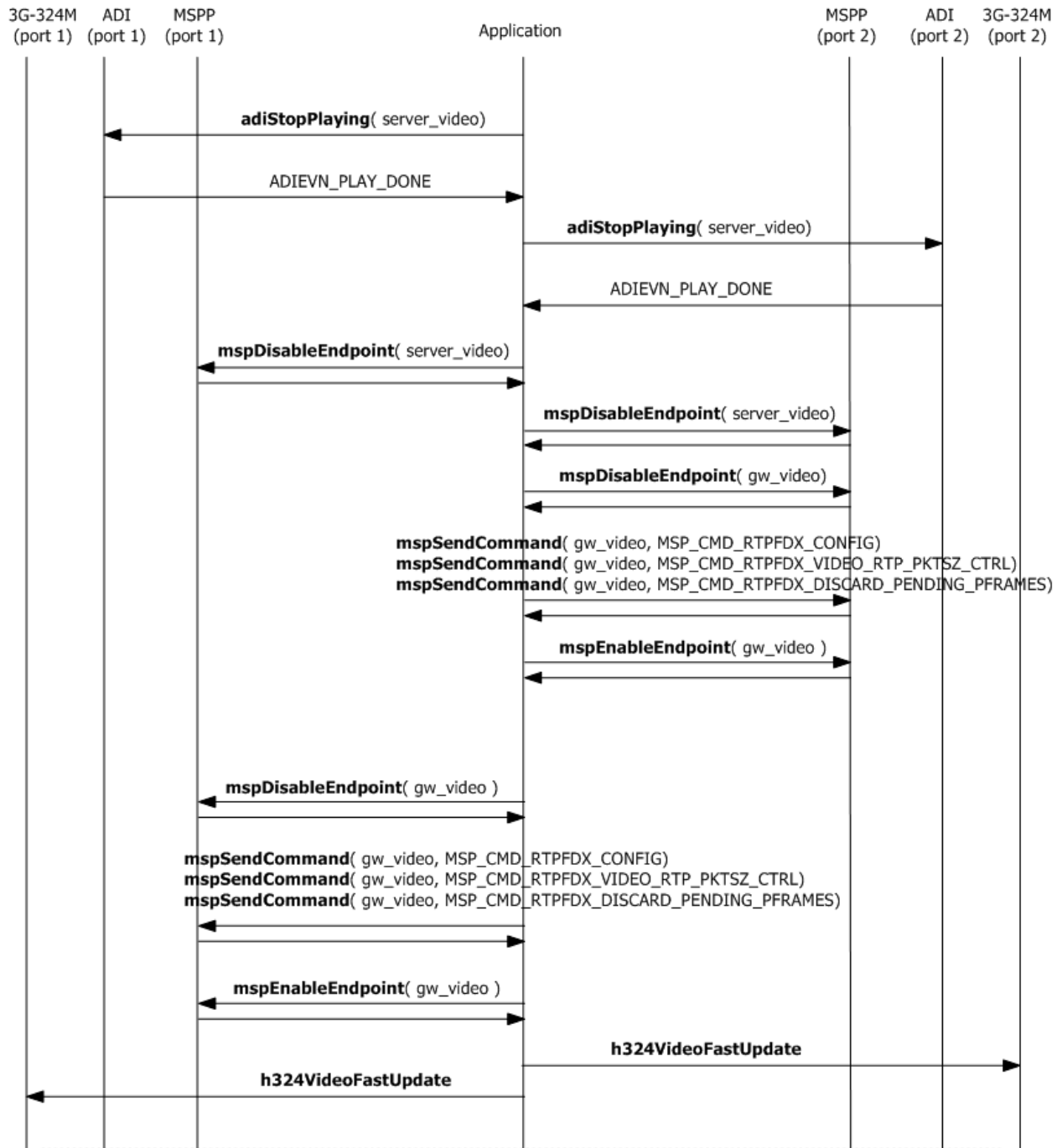
Step	Action
1	Stop the play video function on both server ports and wait for completion (ADIEVN_PLAY_DONE).
2	Disable both server video endpoints to prevent conflicts with gateway endpoints that have the same destination address.
3	Disable the gateway video RTP endpoint of the two tromboned calls.
4	Set the desired RTPAggregationThreshold and MaxRTTPacketSize 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:

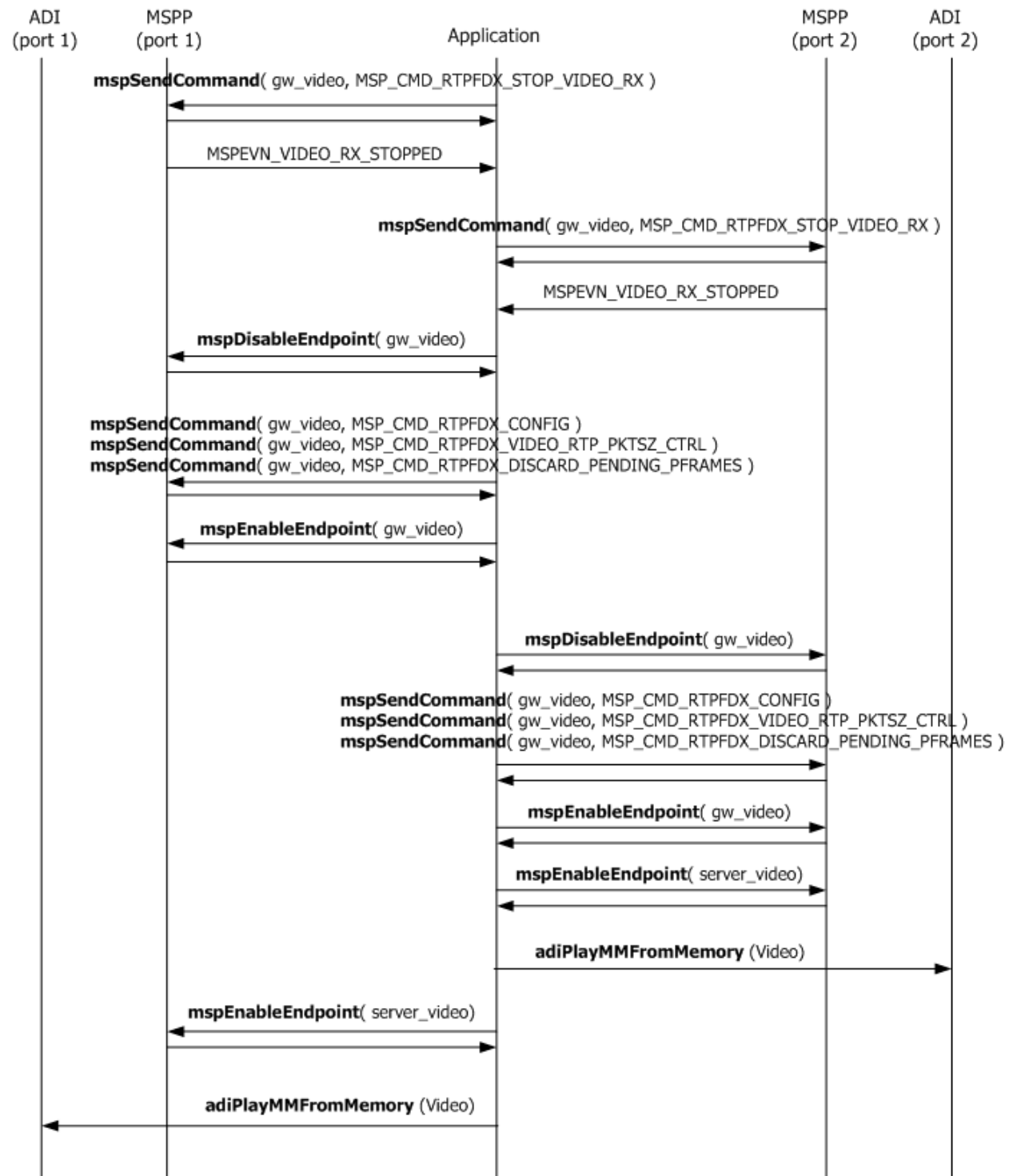
Step	Action
1	Request that both video gateway RTP endpoints stop sending video towards the TDM interface after the end of current frame is detected. The application must receive the corresponding unsolicited event (MSPPEVN_VIDEO_RX_STOPPED) from both endpoints before proceeding.
2	Disable the gateway 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.

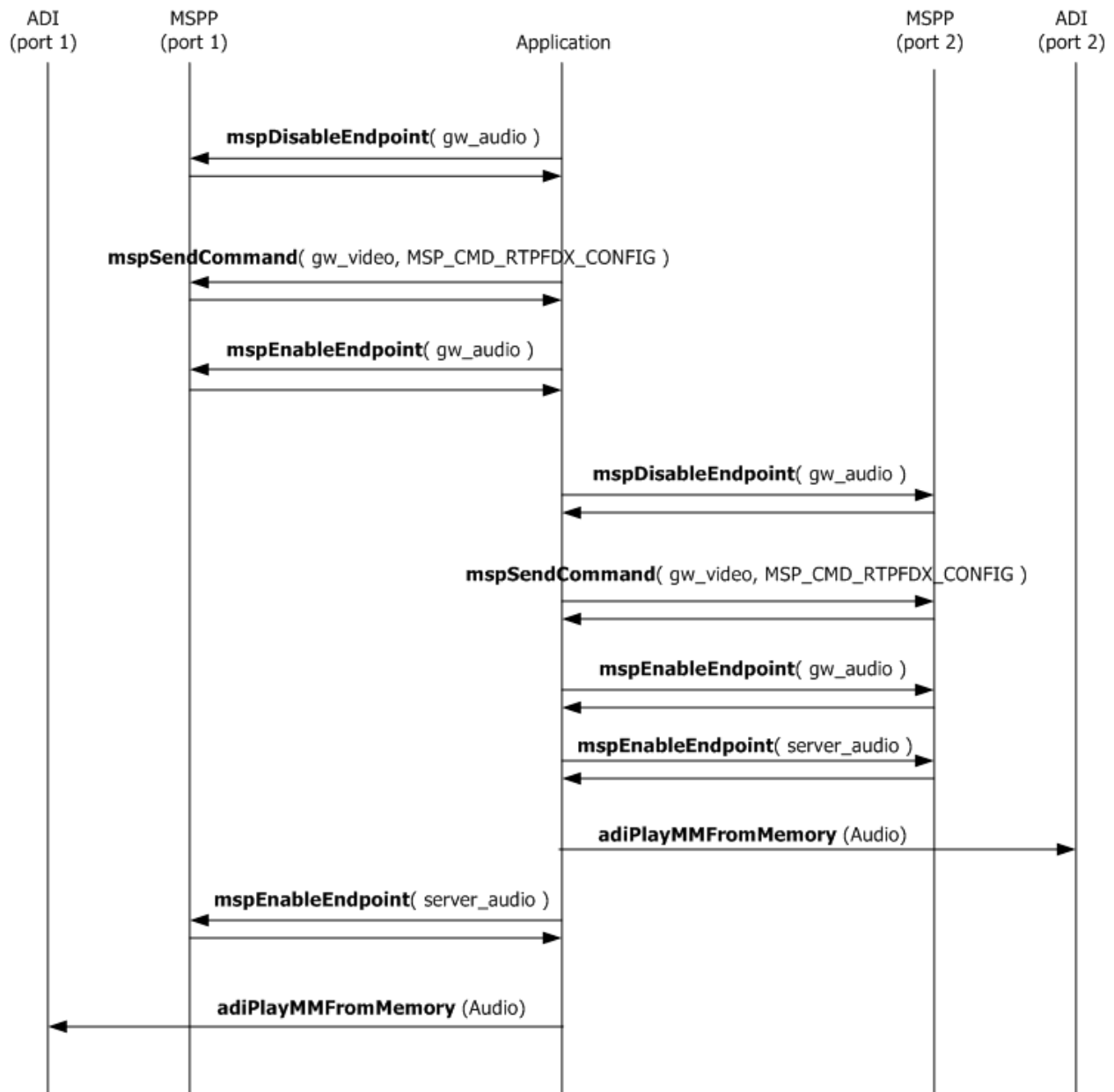
### Switching back video endpoints

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:





## 5. Index

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