



Dialogic[®] 3G-324M Multimedia Gateway

Demo Guide

December 2008

Copyright and Legal Notice

Copyright © 2008 Dialogic Corporation. All Rights Reserved. You may not reproduce this document in whole or in part without permission in writing from Dialogic Corporation at the address provided below.

All contents of this document are furnished for informational use only and are subject to change without notice and do not represent a commitment on the part of Dialogic Corporation or its subsidiaries ("Dialogic"). Reasonable effort is made to ensure the accuracy of the information contained in the document. However, Dialogic does not warrant the accuracy of this information and cannot accept responsibility for errors, inaccuracies or omissions that may be contained in this document.

INFORMATION IN THIS DOCUMENT IS PROVIDED IN CONNECTION WITH DIALOGIC® PRODUCTS. NO LICENSE, EXPRESS OR IMPLIED, BY ESTOPPEL OR OTHERWISE, TO ANY INTELLECTUAL PROPERTY RIGHTS IS GRANTED BY THIS DOCUMENT. EXCEPT AS PROVIDED IN A SIGNED AGREEMENT BETWEEN YOU AND DIALOGIC, DIALOGIC ASSUMES NO LIABILITY WHATSOEVER, AND DIALOGIC DISCLAIMS ANY EXPRESS OR IMPLIED WARRANTY, RELATING TO SALE AND/OR USE OF DIALOGIC PRODUCTS INCLUDING LIABILITY OR WARRANTIES RELATING TO FITNESS FOR A PARTICULAR PURPOSE, MERCHANTABILITY, OR INFRINGEMENT OF ANY INTELLECTUAL PROPERTY RIGHT OF A THIRD PARTY.

Dialogic products are not intended for use in medical, life saving, life sustaining, critical control or safety systems, or in nuclear facility applications.

Due to differing national regulations and approval requirements, certain Dialogic products may be suitable for use only in specific countries, and thus may not function properly in other countries. You are responsible for ensuring that your use of such products occurs only in the countries where such use is suitable. For information on specific products, contact Dialogic Corporation at the address indicated below or on the web at www.dialogic.com.

It is possible that the use or implementation of any one of the concepts, applications, or ideas described in this document, in marketing collateral produced by or on web pages maintained by Dialogic may infringe one or more patents or other intellectual property rights owned by third parties. Dialogic does not provide any intellectual property licenses with the sale of Dialogic products other than a license to use such product in accordance with intellectual property owned or validly licensed by Dialogic and no such licenses are provided except pursuant to a signed agreement with Dialogic. More detailed information about such intellectual property is available from Dialogic's legal department at 9800 Cavendish Blvd., 5th Floor, Montreal, Quebec, Canada H4M 2V9. **Dialogic encourages all users of its products to procure all necessary intellectual property licenses required to implement any concepts or applications and does not condone or encourage any intellectual property infringement and disclaims any responsibility related thereto. These intellectual property licenses may differ from country to country and it is the responsibility of those who develop the concepts or applications to be aware of and comply with different national license requirements.**

Any use case(s) shown and/or described herein represent one or more examples of the various ways, scenarios or environments in which Dialogic products can be used. Such use case(s) are non-limiting and do not represent recommendations of Dialogic as to whether or how to use Dialogic products.

Dialogic, Dialogic Pro, Brooktrout, Cantata, SnowShore, Eicon, Eicon Networks, Eiconcard, Diva, SIPcontrol, Diva ISDN, TruFax, Realblobs, Realcomm 100, NetAccess, Instant ISDN, TRXStream, Exnet, Exnet Connect, EXS, ExchangePlus VSE, Switchkit, N20, Powering The Service-Ready Network, Vantage, Making Innovation Thrive, Connecting People to Information, Connecting to Growth and Shiva, among others as well as related logos, are either registered trademarks or trademarks of Dialogic. Dialogic's trademarks may be used publicly only with permission from Dialogic. Such permission may only be granted by Dialogic's legal department at 9800 Cavendish Blvd., 5th Floor, Montreal, Quebec, Canada H4M 2V9. Any authorized use of Dialogic's trademarks will be subject to full respect of the trademark guidelines published by Dialogic from time to time and any use of Dialogic's trademarks requires proper acknowledgement.

The names of actual companies and products mentioned herein are the trademarks of their respective owners.

This document discusses one or more open source products, systems and/or releases. Dialogic is not responsible for your decision to use open source in connection with Dialogic products (including without limitation those referred to herein), nor is Dialogic responsible for any present or future effects such usage might have, including without limitation effects on your products, your business, or your intellectual property rights.

Using the AMR-NB resource in connection with one or more Dialogic products does not grant the right to practice the AMR-NB standard. To seek a patent license agreement to practice the standard, contact the VoiceAge Corporation at www.voiceage.com/licensing.php.

Publication Date: December 2008

Document Number: 05-2643-002

Contents

REVISION HISTORY	7
ABOUT THIS PUBLICATION	8
Purpose	8
Applicability	8
Intended Audience	8
How To Use This Publication	8
Related Information	9
1 DEMO DESCRIPTION	10
1.1 Overview	10
1.2 SIP Endpoints and the Gateway	11
1.3 3G-SIP Loopback Scenarios	13
1.4 3G-MM Scenarios	17
1.5 3G-RTSP Scenarios	18
1.6 3G Hairpinned Scenarios	19
1.7 Nb UP Scenarios	21
2 SYSTEM REQUIREMENTS	23
2.1 Dialogic® Multimedia Platform (MMP) for AdvancedTCA (ATCA) Hardware/Software Requirements	23
2.2 Dialogic® Host Media Processing Software 3.1LIN (HMP3.1) Hardware/Software Requirements	23
2.3 Dialogic® Multimedia Kit (MMK) for PCIe Hardware/Software Requirements	23
3 PREPARING TO RUN THE DEMO	25
3.1 PSTN Board Configuration	25
3.1.1 Clear Channel	25

3.1.2	ISDN	25
3.1.3	SS7	26
3.2	The 3G-324M Multimedia Gateway Configuration File	27
3.2.1	Configuration File Location	27
3.2.2	Sections in the Configuration File	27
3.3	Sample Configuration Files	39
4	COMPILING AND LINKING	42
4.1	Demo Compilation Options	42
4.1.1	Enabling Dialogic® 3G-324M API MONA Procedures Support...	42
4.1.2	Enabling RTSP Endpoint Support.....	43
4.2	Compiling the Demo.....	43
4.3	Modifying the Demo	44
5	RUNNING THE DEMO	45
5.1	Preparation.....	45
5.2	Run Procedure.....	45
5.2.1	Sample Configuration Scenario Examples	45
5.3	Exiting the Demo	47
5.4	Sample Output	47
5.5	Using the Demo	48
5.5.1	Connecting a Call.....	48
5.5.2	Disconnecting a Call.....	48
APPENDIX	49
	Using the Kapanga SIP Softphone with the Dialogic® 3G-324M Multimedia Gateway Demo.....	49
	Using SIPp with the Dialogic® 3G-324M Multimedia Gateway Demo .	50
	ISDN to ClearChannel Gateway	51
	UDP Socket Protocol.....	53

Figures

Figure 1. 3G Internal Loopback.....	13
Figure 2. 3G External Loopback.....	14
Figure 3. ISDN Gateway Loopback.....	15
Figure 4. SS7 Loopback.....	16
Figure 5. 3G to MM Connection	18
Figure 6. 3G to RTSP Connection.....	19
Figure 7. 3G Hairpinned Connection.....	20
Figure 8. 3G Hairpinned Connection with SIP Inbound/Outbound	21
Figure 9. Nb UP connection	22
Figure 10. ISDN to ClearChannel Gateway Configuration.....	51

Tables

Table 1. Sample Configuration Files	39
---	----

Revision History

This revision history summarizes the changes made in each published version of the document.

Document No.	Publication Date	Description of Revisions
05-2643-002	December 2008	Added RTSP Endpoint support and example RTSP configuration scenarios. Added MONA support and compilation instructions. Updated configuration section.
05-2643-001	June 2008	Initial version of document

About this Publication

This preface provides information about this document in the following sections:

- [Purpose](#)
- [Applicability](#)
- [Intended Audience](#)
- [How To Use This Publication](#)
- [Related Information](#)

Purpose

This guide describes how to run the Dialogic® 3G-324M Multimedia Gateway demo (also referred to as the 3G-324M Demo), for those who choose to do so. The guide describes the demo, lists its requirements, and provides details on how it works.

Applicability

This document version (05-2643-002) is published for Dialogic® Multimedia Software for AdvancedTCA (MMP Software for ATCA) Releases 1.1 and 2.0, Dialogic® Host Media Processing Software Release 3.1LIN (Dialogic® HMP Software), and Dialogic® Multimedia Kit Software Release 1.0 for PCIe (MMK for PCIe).

This document may also be applicable to later software releases (including service updates). Check the Release Guide for your software release to determine whether this document is supported.

Intended Audience

This publication is intended for the following customer types:

- System Integrators
- Independent Software Vendors (ISVs)
- Value Added Resellers (VARs)
- Original Equipment Manufacturers (OEMs)

How To Use This Publication

Refer to this publication after you have installed the Dialogic® software, which includes the 3G-324M Demo software. This publication assumes that you

understand computer telephony terms and concepts, and that you are familiar with and have prior experience with the operating system and the C programming language.

The information in this guide is organized as follows:

- Chapter 1, [Demo Description](#), provides a brief overview of the Dialogic® 3G-324M Multimedia Gateway demo.
- Chapter 2, [System Requirements](#), discusses the requirements for running the demo.
- Chapter 3, [Preparing to Run the Demo](#), lists the tasks to perform before running the demo.
- Chapter 4, [Compiling and Linking](#), describes the steps required to compile the source code for the demo.
- Chapter 5, [Running the Demo](#), describes the steps required to run and to stop the demo, and shows a sample output.

Related Information

For information on the Dialogic® 3G-324M software, see the Dialogic® 3G-324M API Programming Guide and Library Reference.

See the following for additional information:

- For Dialogic® product documentation, see <http://www.dialogic.com/manuals>
- For Dialogic technical support, see <http://www.dialogic.com/support/>
- For Dialogic® product information, see <http://www.dialogic.com>

1 Demo Description

This chapter provides a description of the Dialogic® 3G-324M Multimedia Gateway Demo. The following topics are covered:

- Overview
- SIP Endpoints and the Gateway
- 3G-SIP Loopback Scenarios
- 3G-MM Scenarios
- 3G-RTSP Scenarios
- 3G Hairpinned Scenarios
- Nb UP Scenarios

1.1 Overview

The Dialogic 3G-324M Multimedia Gateway Demo (also referred to as the 3G-324M Demo) is a 3G to SIP gateway application which, on one side, demonstrates 3G mobile connectivity to a remote 3G-324M endpoint using the Dialogic® 3G-324M API library. The other side of the 3G-324M gateway application bridges the call to a local SIP endpoint, multimedia (audio and video) endpoint, or RTSP (Real Time Streaming Protocol) endpoint. In addition, a hairpinned 3G-3G call may also be demonstrated, where an inbound 3G call is connected to an outbound 3G call and a mobile-to-mobile connection is established.

The 3G-324M Demo is fully bi-directional. This means that it will accept an incoming call from either the 3G side or SIP side, and attempt an outgoing call using the opposite technology. On successful completion of both call legs, audio and video are routed together until either leg is terminated.

Its bi-directionality makes it suitable for a range of different loopback tests. This means that it is possible to exercise the 3G-324M capabilities of a platform without a live 3G connection or 3G test utility. This can be important in verifying system operation and for learning the Dialogic 3G-324M API.

The 3G-324M Demo features the 3G-324M protocol library connectivity software. When used, the demo provides 3G audio and video connectivity to a remote 3G mobile network. All platforms use the Linux operating system. Note that all features described in this document are not supported on every platform. Please see the specific product Release Guide for more information about supported features.

The application is primarily based on the following Dialogic® APIs:

- Dialogic® 3G-324M API for 3G connectivity
- Dialogic® Global Call API for PSTN network interface connection

- Dialogic® Global Call API for Session Initiation Protocol (SIP) call control
- Dialogic® IP Media Library (IPML) API for Real-Time Transport Protocol (RTP) media manipulation
- Dialogic® Multimedia API for Play/Record of Multimedia files

The features of the 3G-324M Demo include:

- Using a Global Call PSTN network device for an E1 or T1 clear channel, SS7 or ISDN connection. Direct ISDN network connections are not supported on Dialogic MMP Software for ATCA.
- Establishing a H.223 data call session
- Multiplexing/de-multiplexing 3G-324M audio/video data
- Handling inbound/outbound SIP calls
- Bridging 3G calls to SIP calls
- Passing full-duplex audio/video data between a 3G mobile endpoint and a SIP endpoint
- Detecting and forwarding RFC 2833 DTMF digits between endpoints
- Connecting an inbound 3G call to a Multimedia endpoint to play or record an audio/video file on DTMF detection
- Connecting an inbound 3G call to an RTSP (Real-Time Streaming Protocol) server to request on-demand delivery of real-time multimedia data via DTMF detection (compile-time option)
- Connecting an inbound 3G call and hairpinning the call back out to an external 3G endpoint
- Demonstrating the use of the Nb User Plane (Nb UP) protocol for transporting the 3G data over IP
- Demonstrating the use of Media Oriented Negotiation Acceleration (MONA) procedures with Dialogic® 3G-324M API (compile-time option)
- Receiving a remote UDP socket indication of a PSTN call (optional)

1.2 SIP Endpoints and the Gateway

The audio and video codecs most commonly used with the 3G-324M protocol are the Adaptive Multi-Rate Narrow Band codec (AMR-NB) for audio and H.263 for video.

Note: Using the AMR-NB resource in connection with one or more Dialogic products does not grant the right to practice the AMR-NB standard. To seek a patent license agreement to practice the standard, contact the VoiceAge Corporation at www.voiceage.com/licensing.php.

The most common demo configuration, supported on all platforms, uses H.263 and AMR codecs “natively”, passing audio and video data without transcoding. Native H.263 video and AMR audio is supported by all Dialogic® 3G-324M enabled products. Video and audio transcode mode is supported on MMP Software for ATCA 2.0 and MMK for PCIe Advanced video platforms only. In the transcoding case, codecs supported on IP endpoints and multimedia files include MPEG-4 or H.263 for video and AMR-NB, G.711 mu-law, G.711 A-law, G.723 or G.729 for audio. However, since the default demo configuration supported on all platforms does not do any transcoding, the demo normally requires a SIP phone that supports the H.263 and AMR-NB codecs.

One such phone is the Kapanga video softphone, available from <http://www.kapanga.net/ip/home.cfm>.

It can be set up to use the default H.263 and AMR-NB codecs. Instructions for configuring the phone can be found in the Appendix, **Using the Kapanga SIP Softphone with the Dialogic® 3G-324M Multimedia Gateway Demo**.

A note on using the Kapanga phone: It is possible to limit the total bandwidth of audio/video sent by the phone so that it does not produce a video rate higher than 35 Kbps. This is necessary, as a single DS0 (64 Kbps) is the total bandwidth available for a 3G call. However, the Kapanga phone video bitrate limitation is an average bitrate setting. This may cause the video data rate to peak beyond the 64 Kbps limit of the 3G-324M data pipe. The issue may manifest as a video delay through the system or dropped video or audio data. In any event, media quality on the Dialogic HMP Software system, MMP Software for ATCA system, or MMK Software for PCIe system should not be judged by using the Kapanga phone alone, as it is not a true 3G-324M endpoint connection.

Another possibility for non-interactive testing is using the SIPp test utility. It can be found at <http://sipp.sourceforge.net/> and is an open-source project that may be used as a SIP endpoint either in the client or server mode. It is driven by an XML scenario script that can be set up to negotiate any audio or video codec, including AMR and H.263. Media is provided by a recorded packet capture (PCAP) file of an RTP stream of each of these two media types. Combining the XML script and media files results in a set of full SIP calls, with media streams. It is not possible to interactively view the video or hear the audio; however, the gateway itself operates the same as when interactive SIP endpoints are used. References to scripts and media files to drive SIPp can be found in the Appendix, **Using SIPp with the Dialogic® 3G-324M Multimedia Gateway Demo**.

Finally, the 3G-324M Gateway can be used in conjunction with the Dialogic® Multimedia demo to demonstrate video mail and video portal capabilities to a 3G mobile user. The Dialogic Multimedia demo can be used on the same system as the 3G-324M Gateway application, or on a separate system. The Dialogic Multimedia demo can be configured to use AMR-NB for audio. It demonstrates two use cases:

- 3G video mail/video blog – a 3G mobile device can record an audio/video clip and play it back at a later time.
- 3G video portal – a 3G mobile device can select video clips from a menu and stream to the 3G mobile device

1.3 3G-SIP Loopback Scenarios

In many environments, a 3G network connection or 3G simulator is not available for running the demo. In this case, several different loopback scenarios, where an outbound 3G channel is connected to an inbound channel, are possible. Note that not every possible scenario is shown here.

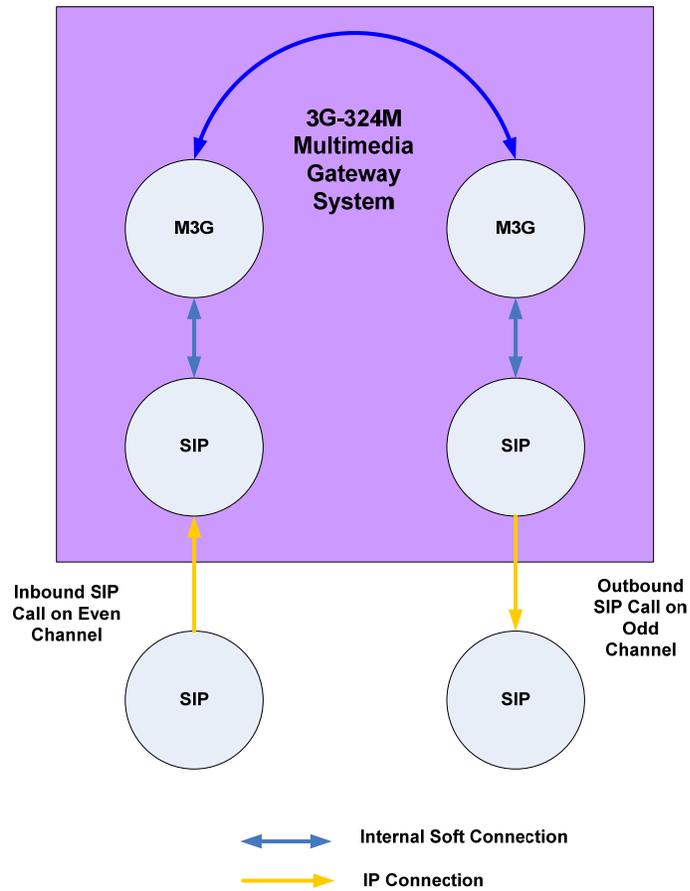


Figure 1. 3G Internal Loopback

In this scenario, two 3G devices (called M3G) created and controlled by Dialogic 3G-324M API library calls are connected via software to one another. DS0 channels on a PSTN board are not used. One 3G channel communicates directly with the other.

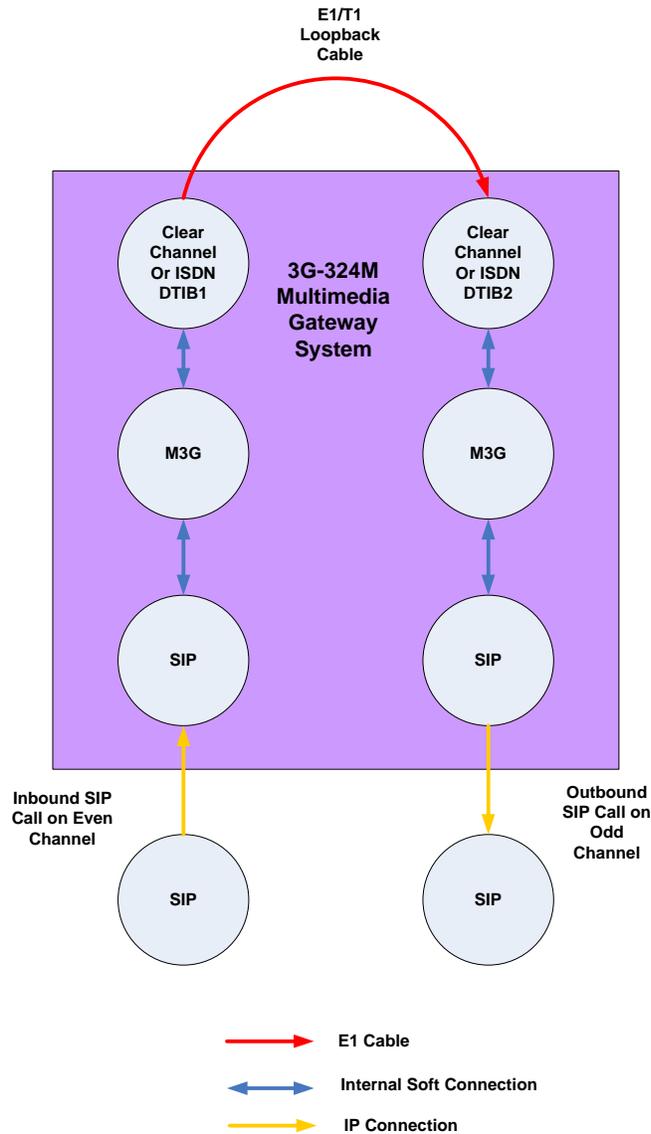


Figure 2. 3G External Loopback

In this scenario, two 3G devices (called M3G) created and controlled by Dialogic 3G-324M API library calls are connected to DS0 timeslots on two different E1 spans on a PSTN board. The two spans are physically connected by a crossover cable, allowing the 30 channels on one span to communicate with the corresponding channels on the other span.

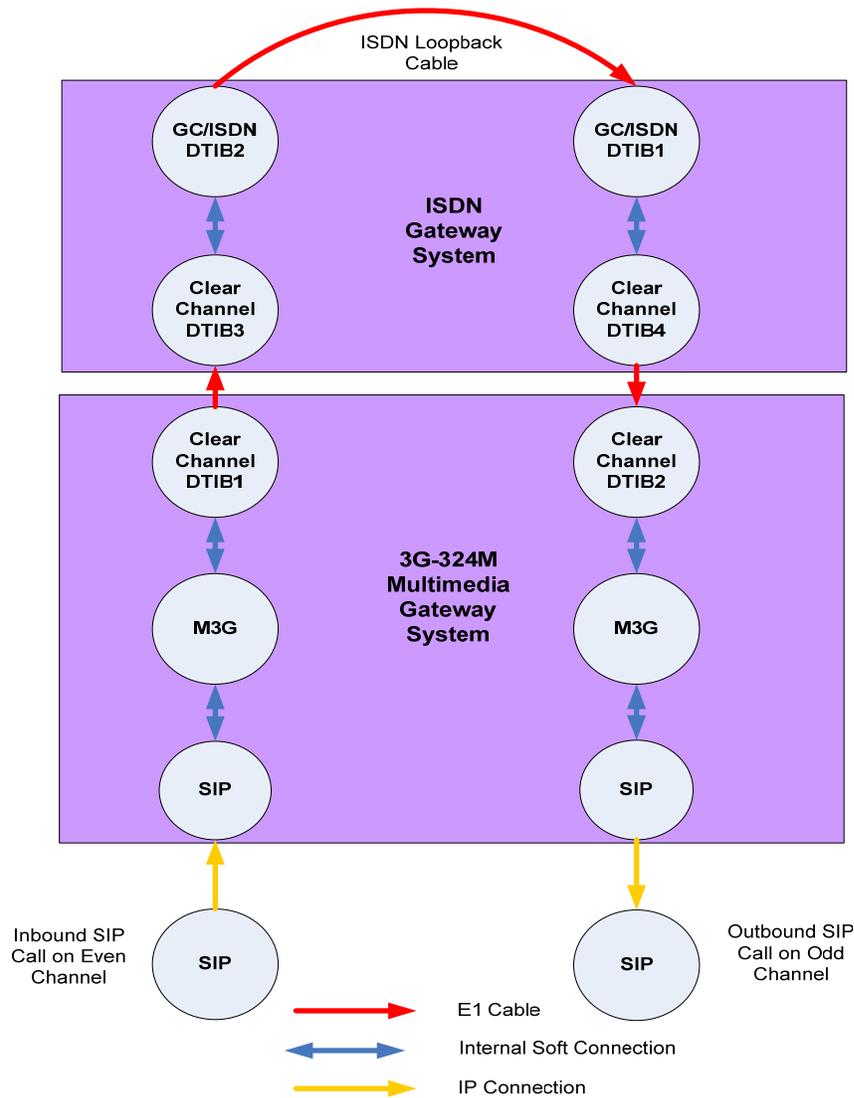


Figure 3. ISDN Gateway Loopback

As in 3G external loopback, two 3G devices (called M3G) created and controlled by Dialogic 3G-324M API library calls are connected to DS0 timeslots on two different E1 spans on a PSTN board. The two spans are physically connected to a second system. This system may act as a signaling gateway to a 3G network or, in this case, may loop back one channel to another and provide a 3G pathway independent of the system running the 3G-324M gateway. This separate ISDN signaling gateway is only needed on the Dialogic® Multimedia Platform for AdvancedTCA, which does not currently support ISDN signaling. Instructions for building an ISDN signaling gateway can be found in the Appendix, **ISDN to ClearChannel Gateway**.

These scenarios require different endpoint configuration, as will be covered in the section Endpoint Configuration.

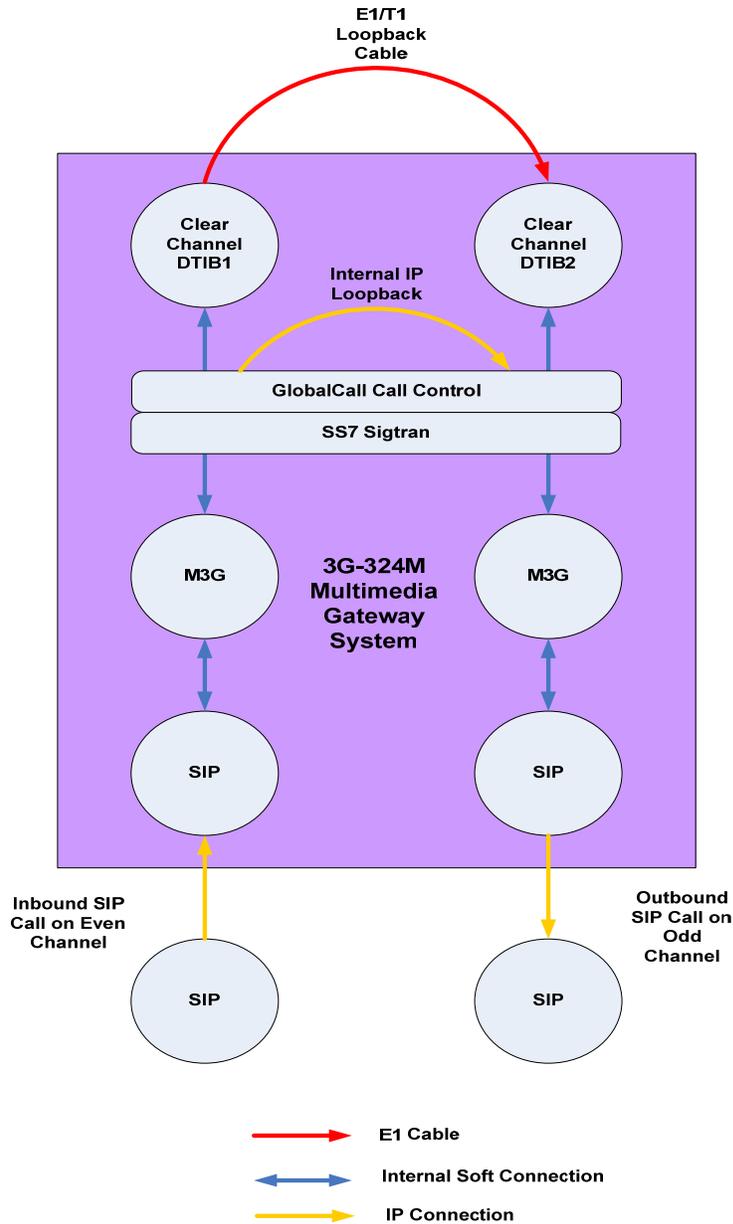


Figure 4. SS7 Loopback

In this scenario, SS7/Sigtran is used for signaling to set up the 3G call. This scenario is similar to the external loopback. Two 3G devices created and controlled by 3G-324M API library calls are connected to clear channel DS0 timeslots on two different E1 spans on a PSTN board. The two spans are physically connected by a crossover cable. The difference here is that the signaling for the outbound and inbound calls is done using SS7/Sigtran.

SS7 signaling between endpoints is carried via IP. Once the calls are connected, 3G negotiations over the clear channel connections can then take place.

This scenario requires a working SS7/Sigtran/Global Call installation on the 3G-324M Gateway system. Setting this up is beyond the scope of this document. Refer to the application note “Configuring an SS7/Sigtran Demo System on Host Media Processing 3.1LIN” for instructions on downloading the SS7 software and configuring the system.

1.4 3G-MM Scenarios

In addition to connecting a 3G to a SIP endpoint, it is also possible to pair a 3G endpoint (called M3G) with a multimedia endpoint (called MM). In this case, the 3G endpoint has a port connection to the multimedia endpoint, and a pair of Dialogic multimedia files are played or recorded when the call is terminated at the multimedia endpoint. The multimedia play/record operation and file name is selectable by DTMF digit from the 3G endpoint that are received as H.245 User Input Indication messages on the M3G interface. Using the configuration file, the user can specify a single operation per DTMF/UII <0-9> so that the system responds to digit events from the 3G endpoint. The supported operations are:

- **PLAY** – A multimedia file is played to the 3G endpoint when the specified DTMF/UII is selected.
- **RECORD** - A multimedia file is recorded when the specified DTMF/UII is selected. While recording, a second DTMF/UII stops the recording and immediately plays back the recorded file.

A call is disconnected when the 3G caller hangs up.

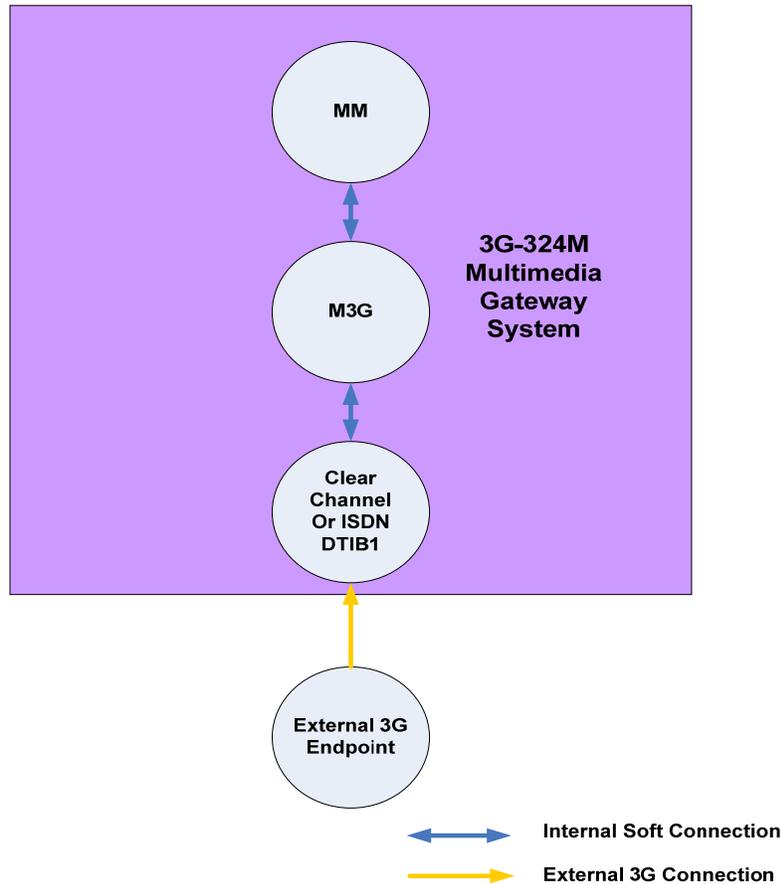


Figure 5. 3G to MM Connection

It is also possible to combine this scenario with an inbound SIP-3G endpoint pair using either an internal or external loopback. This allows a SIP phone to be used to demonstrate a 3G call working with a multimedia endpoint.

1.5 3G-RTSP Scenarios

Along with connecting a 3G endpoint to a SIP endpoint or to a Multimedia device for local play/record, it is also possible to pair a 3G endpoint (called M3G) with an RTSP endpoint. In this case, an RTSP endpoint is a Dialogic® IP Media device (ipm device) for RTP streaming with an associated RTSP client interface. The RTSP endpoint works as an RTSP client and establishes an RTSP session to request media (via rtsp URL) from an RTSP server. The rtspURL that is requested from the RTSP Server is selectable by DTMF digit. Using the configuration file, the user can specify a single rtsp URL per DTMF/UII <0-9> so that the system requests a different rtsp URL for each DTMF.

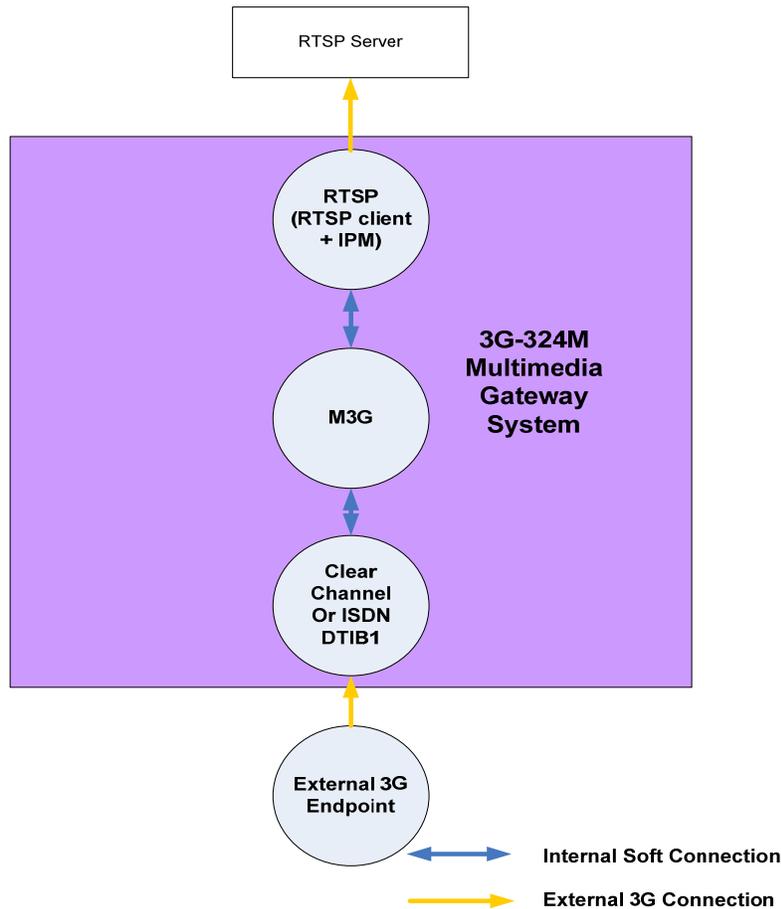


Figure 6. 3G to RTSP Connection

It is also possible to combine this scenario with an inbound SIP-3G endpoint pair using either an internal or external loopback. This allows a SIP phone to be used to demonstrate a 3G call working with a RTSP server.

1.6 3G Hairpinned Scenarios

A 3G endpoint (called M3G) may also be paired with another 3G endpoint to form a hairpinned 3G connection. Here, all A/V media streams are passed through the 3G endpoint pair. This simulates the end result of a 3G supervised call transfer scenario. For example, an inbound 3G call may already be held on the platform engaged in a multimedia play. The application then decides to hold the caller while an outbound 3G call is made. When the outbound call is ready, the inbound 3G is connected to the outbound 3G.

Although the gateway does not demonstrate the interconnection changes, it does show the result in this 3G hairpinned scenario. More simply, a 3G-3G pair is configured, and an inbound 3G call triggers an outbound 3G call. On completion, an A/V path is established between the two external 3G endpoints.

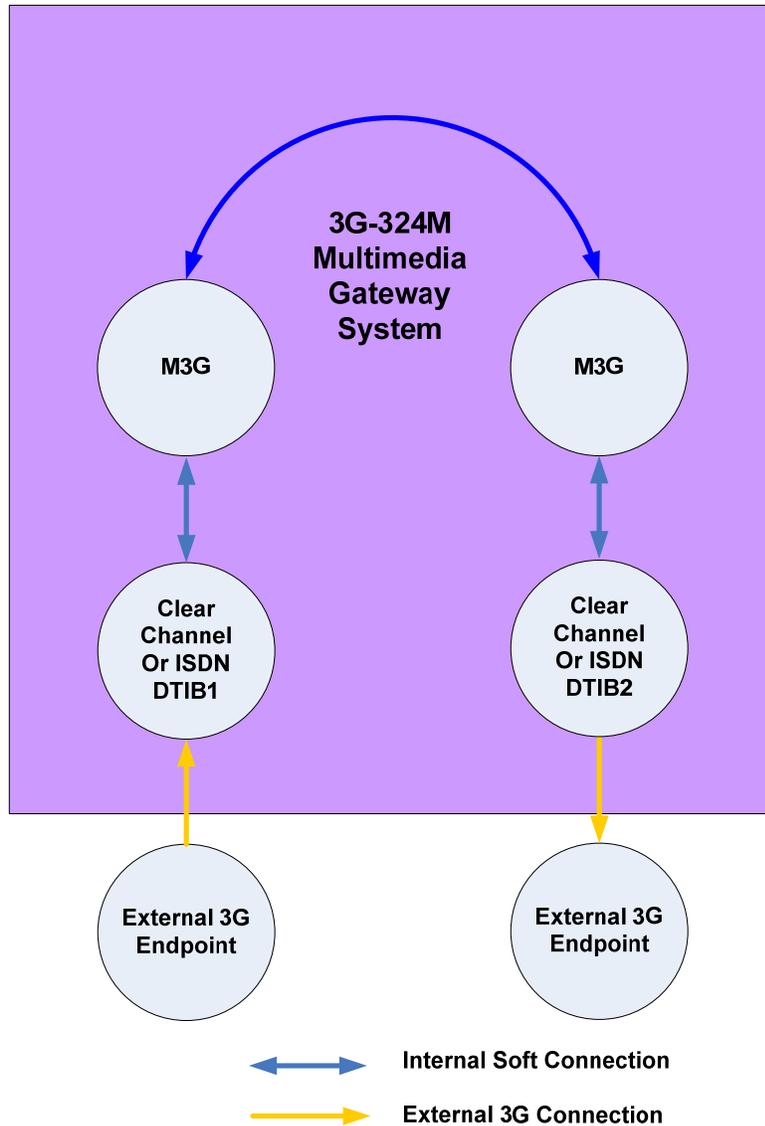


Figure 7. 3G Hairpinned Connection

It is also possible to combine this scenario with two SIP-3G endpoint pairs to demonstrate a complete call path without a 3G connection or simulator. Here, external loopback connections are needed. A SIP phone places a call to a SIP-3G pair. The 3G goes out through an external loopback to the 3G-3G hairpin. The outbound 3G call then goes again through the external loopback and into a second 3G-SIP pair. This results in an outbound SIP call to a second external SIP phone. The result is the two SIP phones are interconnected through the 3G-3G hairpin on the gateway.

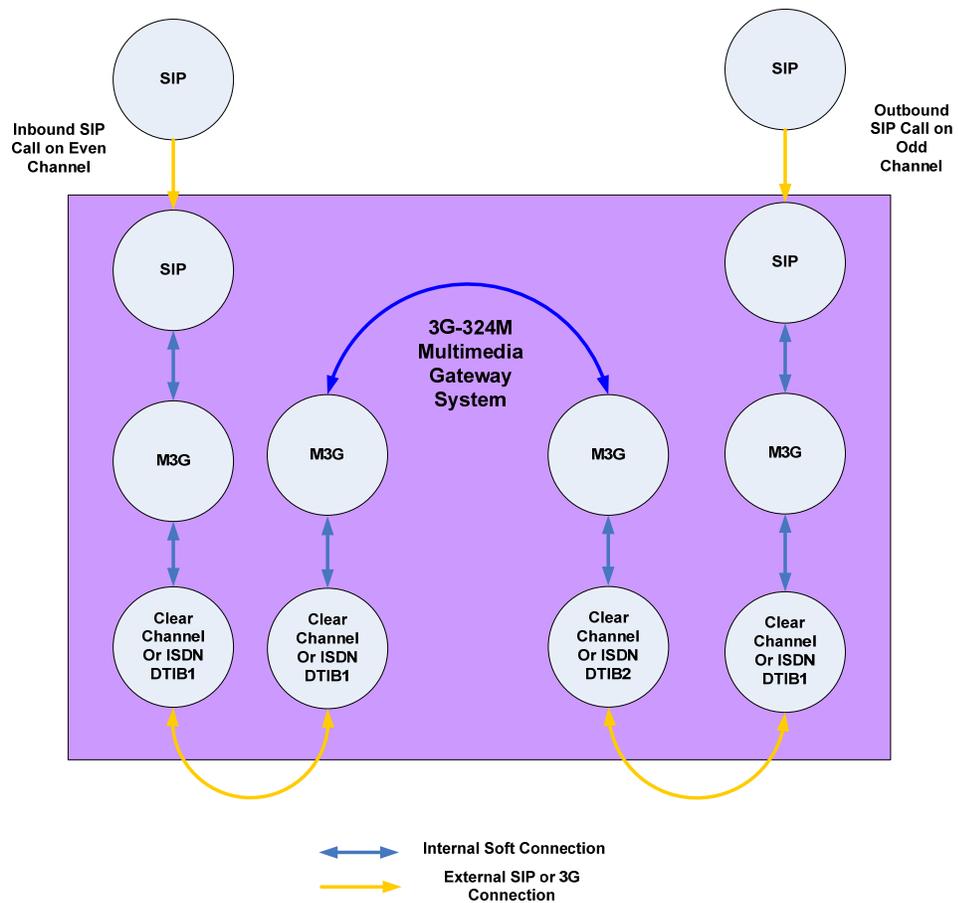


Figure 8. 3G Hairpinned Connection with SIP Inbound/Outbound

1.7 Nb UP Scenarios

The 3G-324M Demo may also be configured to use Nb UP as the aggregate transport. It converts the 3G data that normally travels over a TDM E1/T1 line to an RTP packet stream that may be sent over an IP connection. Most of the loopback scenarios described above can use an Nb UP connection. Here is an example of a pair of external SIP endpoints connected through the gateway using Nb UP as the 3G transport:

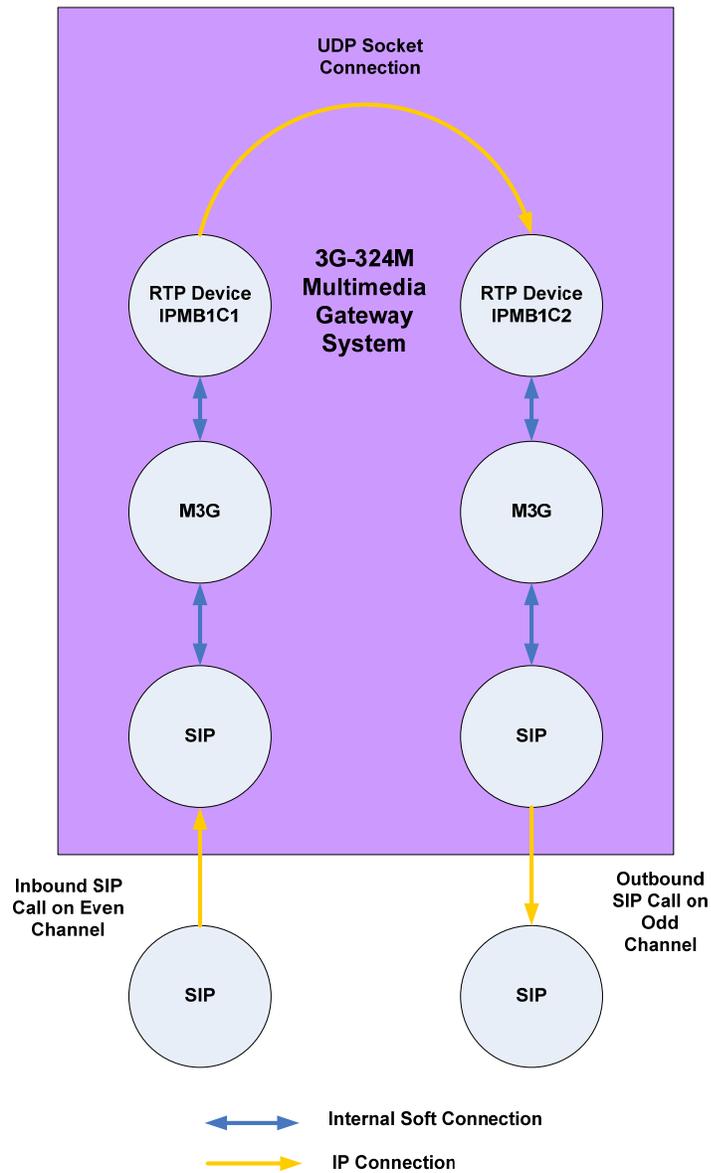


Figure 9. Nb UP connection

The UDP socket connection uses two different ports on the same system to pass RTP/Nb UP data. If two separate systems contained the 3G endpoints, some method of coordinating the Nb UP connection on either side would be needed. Bearer Independent Call Control (BICC) would likely be used. The gateway does not do this.

2 System Requirements

This chapter describes the hardware and software requirements that must be met before you can run the Dialogic® 3G-324M Multimedia Gateway Demo:

- [Dialogic® Multimedia Platform \(MMP\) for AdvancedTCA \(ATCA\) Hardware/Software Requirements](#)
- [Dialogic® Host Media Processing Software 3.1LIN \(HMP3.1\) Hardware/Software Requirements](#)
- [Dialogic® Multimedia Kit \(MMK\) for PCIe Hardware/Software Requirements](#)

2.1 Dialogic® Multimedia Platform (MMP) for AdvancedTCA (ATCA) Hardware/Software Requirements

For hardware requirements, refer to the Release Guide for the MMP for ATCA software version you are using. Required MMP hardware includes:

- Dialogic® Multimedia Platform Blade for AdvancedTCA with AMC
- Dialogic® Multimedia Platform IP+TDM (Rear Transition Module) RTM module
- Dialogic® Multimedia Platform RTM breakout box kit

2.2 Dialogic® Host Media Processing Software 3.1LIN (HMP3.1) Hardware/Software Requirements

If external loopback scenarios are used or connections to a 3G network or test tool are desired, a Dialogic® HMP Interface board (DNI board), which has a PSTN interface will be required. For a loopback scenario, at least two E1/T1 spans will be needed.

Dialogic HMP Software licenses for the 3G-324M Gateway, RTP, Dialogic® Multimedia, and Global Call SIP call control will also be needed, as required by the scenarios used.

2.3 Dialogic® Multimedia Kit (MMK) for PCIe Hardware/Software Requirements

Required hardware for the MMK for PCIe includes a Dialogic® Multimedia Accelerator Board for PCIe.

If external loopback scenarios are used or connections to a 3G network or test tool are desired, a DNI board will also be required. For a loopback scenario, at least two E1/T1 spans will be needed.

In addition to meeting the software license requirements in the release documentation, other possible requirements are:

- SIP – one or two SIP video endpoints as described in the previous section
- 3G – a 3G network connection with live 3G mobile endpoints or a simulated 3G endpoint to initiate a 3G call if one of the loopback modes are not used
- An ISDN/CC gateway (optional)
- An ISDN or SS7 signaling endpoint

3 Preparing to Run the Demo

This chapter describes what you need to do before running the Dialogic® 3G-324M Multimedia Gateway Demo, should you choose to do so.

- [PSTN Board Configuration](#)
- [The 3G-324M Multimedia Gateway Configuration File](#)
- [Sample Configuration Files](#)

3.1 PSTN Board Configuration

Certain scenarios use a TDM E1/T1 network interface. This must be configured correctly to carry 3G data. Internal loopback and scenarios that use Nb UP transport do not require a TDM network interface.

3.1.1 Clear Channel

Clear channel is used for external connections, either to an ISDN or SS7 signaling gateway or for an external loopback test scenario. No signaling is done on the board in this case. Instructions for configuring the interface on the Dialogic® Multimedia Platform for AdvancedTCA can be found in the Dialogic® Multimedia Blade for AdvancedTCA Technical Product Specification.

Configuring the interface on the Dialogic® HMP Interface board (DNI board) under Dialogic® Host Media Processing Software or Dialogic® Multimedia Kit for PCIe can be found in the Dialogic® System Configuration Guide.

3.1.2 ISDN

ISDN is available on Dialogic HMP Software and Dialogic Multimedia Kit (MMK) for PCIe only.

For a direct connection to an external ISDN switch or for testing loopback scenarios using ISDN, the E1/T1 interface must be configured correctly to match the ISDN protocol being used. That is beyond the scope of this document. Refer to the Dialogic System Configuration Guide.

If using an external loopback scenario with ISDN, remember that one E1/T1 span must operate in the default 'User mode' and the other must be configured to work in the 'Network mode'.

Data streams must also be configured on Dialogic HMP Software or MMK for PCIe to allow communication between the DNI board and the Dialogic HMP/MMK software. Again, refer to the Dialogic System Configuration Guide for instructions on how to do this. Consult the section “Transparent Data Routing for 3G-324M”.

3.1.3 SS7

When using SS7 signaling via Sigtran, all DS0 channels are configured as clear channel. See the Clear Channel section above. SS7 also requires a working SS7/Sigtran/Global Call installation on the 3G gateway system. Setting this up is beyond the scope of this document. Refer to the application note “Configuring an SS7/Sigtran Demo System on Host Media Processing 3.1LIN” for instructions on downloading the SS7 software and configuring the system.

3.2 The 3G-324M Multimedia Gateway Configuration File

This section discusses how to configure the demo for your system. It contains the following topics:

- [Configuration File Location](#)
- [Sections in the Configuration File *config.txt*](#)

The demo runs with the help of the configuration file named *config.txt*. Before running the 3G-324M Demo, the file must be set to reflect your endpoint configuration, system environment and 3G-324M settings. Each section in the configuration file is explained below.

3.2.1 Configuration File Location

This configuration file *config.txt* should reside in the Release directory of the demo.

In addition, a set of sample configuration files in the *sample_config_files* directory accompany the demo. Copy a file from this directory into *config.txt* as the starting point for your scenario. Alternatively, configuration files other than *config.txt* can be used by specifying them in the command line with the (-i) option. An example of this is ‘./m3g-sip_gateway -i <config_file_name>’

3.2.2 Sections in the Configuration File

Endpoint Configuration

This parameter section designates the endpoint configuration setup for the application. There are four types of endpoints:

- 3G endpoint – the devices used to connect to a remote 3G-324M device. They, in turn, can be paired with either a SIP, MM, RTSP or another 3G endpoint
- SIP endpoint – the devices used to connect to a remote SIP device
- MM endpoint – the multimedia devices that will terminate the 3G call
- RTSP endpoint – the device with associated RTSP client functionality that will connect to an RTSP server endpoint.

3G Endpoint

The 3G endpoint specifies the devices used in making a 3G connection to a 3G-324M device. It has the following attributes:

Endpoint Number – EP: <n> – A unique numerical endpoint number. All endpoints should be consecutively numbered

Endpoint type – m3g – The 3G endpoint type designates the 3G side of the call

Control Device – m3gB<x>T<y> - The 3G device that handles the 3G call for this endpoint

Stream Device – Five entries for stream devices are possible:

External clear channel - dtiB<x>T<y> - The E1/T1 data channel that carries the H.223 multiplexed data

ISDN - IdtiB<x>T<y> - The E1/T1 data channel that carries the H.223 multiplexed data. The span specified also carries ISDN signaling to an external ISDN switch. (Dialogic HMP Software and MMK only)

SS7 - SdtiB<x>T<y> - SS7/Sigtran is used to provide signaling for the E1/T1 data channel that carries the H.223 multiplexed data.

NBUP – ipmB1C<x> - The IP channel that carries the H.223 multiplex data

InternalLoopback - NONE – Used to designate internal loopback mode where no transport is needed for H.223 multiplex data

Local Peer - <n> - Specifies the number of the corresponding SIP, MM, RTSP or 3G endpoint peer for this 3G endpoint. Each 3G endpoint must have a corresponding local peer.

Loopback Peer - <n> - Specifies the corresponding 3G endpoint for 3G to 3G internal/external loopback mode. When loopback mode is not desired, a 0 should be entered. When using ISDN and SS7 stream devices, a loopback peer does not need to be specified, since the signaling for call setup/teardown is done by the respective protocol rather than directly by the gateway application.

Map ID - <text> – Specifies a unique text string to identify an endpoint for socket commands when an external ISDN signaling gateway is used. On an outbound call to an external ISDN gateway, the Map ID value will be sent via socket command to the external ISDN signaling gateway to uniquely identify this particular endpoint. On an inbound call from an external ISDN gateway, it is expected that the external gateway will send this Map ID to indicate that the call has arrived on this particular endpoint.

Audio/Video XCode - <(N)NATIVE or (T)TRANSCODE > - Specifies the Audio/Video Transcode mode between the endpoint EP(tx) and the endpoint's Local Peer(rx). (ie, EP→Local Peer). (N)NATIVE represents that the media will not be altered or modified in any way. (T)TRANSCODE represents that the media will be modified into an intermediate 'raw' format before being transcoded into the destination format. This field is equivalent to enabling or disabling the TRANSCODE_ON flag at the audio or video connection from the endpoint to the local peer device. The Audio/Video Xcode direction is in the internal media stream tx direction. If (T)TRANSCODE is specified the gateway will convert media from the Network specified codec to the codec specified at the local peer. Audio and video is specified separately for each endpoint.

Note: (N)NATIVE is supported on all platforms. (T)TRANSCODE is only supported on platforms that support transcoding between devices.

SIP Endpoint

The SIP endpoint specifies the devices used to make a SIP connection. Since this is a bidirectional gateway that is primarily used in a loopback mode, SIP endpoints are usually entered in pairs. Odd numbered devices (e.g. iptB1T1, iptB1T3) are used for outbound calls and even numbered devices (e.g. iptB1T2, iptB1T4) are used for inbound calls. Also, each SIP device must be paired with a corresponding 3G device.

A SIP endpoint has the following attributes:

Endpoint Number – EP:<n> – A unique numerical endpoint number. All endpoints should be consecutively numbered

Endpoint type – sip – The SIP endpoint type designates the SIP side of the call

Control Device – iptB<x>T<y> - Specifies the ipt device that handles the SIP call control for this endpoint

Stream Device – ipmB1T<y> - Specifies the ipm device used for RTP data traffic for this endpoint

Local Peer – <n> – Specifies the number of the corresponding 3G endpoint for this SIP endpoint. Each SIP endpoint must have a 3G endpoint

Loopback Peer – 0 – This parameter is not used for the SIP endpoint

Map ID – none – This parameter is not used for the SIP endpoint

Audio/Video XCode - <(N)NATIVE or (T)TRANSCODE > - Specifies the Audio/Video Transcode mode between the endpoint EP(tx) and the endpoint's Local Peer(rx). (ie, EP→Local Peer). (N)NATIVE represents that the media will not be altered or modified in any way. (T)TRANSCODE represents that the media will be modified into an intermediate 'raw' format before being transcoded into the destination format. This field is equivalent to enabling or disabling the TRANSCODE_ON flag at the audio or video connection from the endpoint to the local peer device. The Audio/Video Xcode direction is in the internal media stream tx direction. If (T)TRANSCODE is specified the gateway will convert media from the Network specified codec to the codec specified at the local peer. Audio and video is specified separately for each endpoint.

Note: (N)NATIVE is supported on all platforms. (T)TRANSCODE is only supported on platforms that support transcoding between devices.

Note: Many parameter configuration settings beyond this endpoint setting have an "Endpoints" column. The value in this column refers back to the endpoint number defined here. If there are, for example, four 3G endpoints, there must be an entry for each of them in further 3G parameter configuration.

Below is an example of endpoint definitions that may be used for internal loopback, external loopback and an external connection to an ISDN gateway. (Only the external loopback is active – the others are commented out.)

```
#####
# Endpoint Control Stream Local Loopback Map ID Audio Video
# Num Type Device Device Peer Peer Xcode Xcode
#####
# Internal loopback
```

```

#EP: 1 m3g m3gB1T1 NONE 3 2 2 N N
#EP: 2 m3g m3gB1T2 NONE 4 1 1 N N
#
# Clear channel external loopback connection from span 1 to span 2
EP: 1 m3g m3gB1T1 dtiB1T1 3 2 2 N N
EP: 2 m3g m3gB1T2 dtiB2T1 4 1 1 N N
#
# ISDN external loopback connection from span 1 to span 2
#EP: 1 m3g m3gB1T1 IdtiB1T1 3 0 2 N N
#EP: 2 m3g m3gB1T2 IdtiB2T1 4 0 1 N N
#
# NBUP loopback connection from span 1 to span 2
#EP: 1 m3g m3gB1T1 ipmB1C1 3 2 2 N N
#EP: 2 m3g m3gB1T2 ipmB1C2 4 1 1 N N
#
# External connection to ISDN Gateway
#EP: 1 m3g m3gB1T1 dtiB1T1 3 2 :N_DTIB1T1:P_ISDN N N
#EP: 2 m3g m3gB1T2 dtiB2T1 4 1 :N_DTIB2T1:P_ISDN N N
#
# SIP endpoints the same in either case
EP: 3 sip iptB1T1 ipmB1C3 1 0 none N N
EP: 4 sip iptB1T2 ipmB1C4 2 0 none N N
#####

```

Multimedia Endpoint

The multimedia endpoint specifies the devices used to terminate 3G calls. It has the following attributes:

Endpoint Number – EP: <n> – A unique numerical endpoint number. All endpoints should be consecutively numbered

Endpoint type – vmm – Multimedia device

Control Device – none

Stream Device – mmB<x>C<y> – Specifies the media device

Local Peer – <n> – Specifies the number of the corresponding 3G endpoint for this multimedia endpoint. Each multimedia endpoint must have a corresponding 3G endpoint

Loopback Peer – <n> – always “0”

Map ID – <text> – always “none”

Audio/Video XCode - <(N)NATIVE or (T)TRANSCODE > - Specifies the Audio/Video Transcode mode between the endpoint EP(tx) and the endpoint’s Local Peer(rx). (ie, EP→Local Peer). (N)NATIVE represents that the media will not be altered or modified in any way. (T)TRANSCODE represents that the media will be modified into an intermediate ‘raw’ format before being transcoded into the destination format. This field is equivalent to enabling or disabling the TRANSCODE_ON flag at the audio or video connection from the endpoint to the local peer device. The Audio/Video Xcode direction is in the internal media stream tx direction. If (T)TRANSCODE is specified the gateway will convert media from the specified codec in the multimedia file to the codec specified at the local peer. Audio and video is specified separately for each endpoint.

Note: (N)NATIVE is supported on all platforms. (T)TRANSCODE is only supported on platforms that support transcoding between devices.

Below is an example of a single 3G device terminated at a multimedia device:

```
#####  
# Endpoint Control Stream Local Loopback Map ID Audio Video  
# Num Type Device Device Peer Peer Xcode Xcode  
#####  
#  
# Configuration for a single inbound 3G terminating at an MM  
#  
# 3G endpoint associated with an MM endpoint  
EP: 1 m3g m3gB1T1 dtiB1T1 2 0 none N N  
#  
# Single Multimedia endpoint  
EP: 2 vmm none mmB1C1 1 0 none N N  
#  
#####
```

Logging

This configuration section specifies whether application event and error logging will be enabled (ON) or disabled (OFF). If logging is enabled, log output will be collected in the file, *logfile.txt*.

```
#####  
# Logging (ON/OFF):  
#####  
LOG: ON  
#####
```

DTMF Type

A switch (on/off) may be set so that the gateway will either use a SIP INFO message (on) or use RFC 2833 (off) for DTMF signaling with an external SIP device.

```
#####  
# SIP INFO DTMF (ON/OFF):  
#####  
SIP_INFO_DTMF: OFF  
#####
```

Board Parameter Settings

This parameter configuration section allows the user to set application specified 3G board parameters. The format to set a board parameter is *PRM:<Parameter Name> <value>*

```
#####  
# Board Parameters:  
# Parameters M3G_E_PRM_XXXXXXXXXXXX  
#####  
# Activate UII-RFC2833 DTMF translation  
PRM: DIGIT_TO_MEDIA 1 # M3G_E_PRM_RELAY_DIGIT_TO_MEDIA_DEV  
PRM: DIGIT_TO_H245 1 # M3G_E_PRM_RELAY_DIGIT_TO_H245UII  
PRM: FASTUPDATE_TO_MEDIA 1 #M3G_E_PRM_RELAY_FASTUPDATE_TO_MEDIA_DEV  
PRM: FASTUPDATE_TO_H245 1 # M3G_E_PRM_RELAY_FASTUPDATE_TO_H245  
# PRM: SKEWINDICATION # M3G_E_PRM_SKEWINDICATION
```

```

# PRM: AUDIOVISUALSYNC          # M3G_E_PRM_AUDIOVISUALSYNC
# PRM: RETRANSMIT_ON_IDLE 1     # M3G_E_PRM_RETRANSMIT_ON_IDLE
PRM: H245_TERMINAL_TYPE 127    # M3G_E_PRM_H245_TERMINAL_TYPE
# PRM: MAX_CCSSLR_SEGMENT 255   # M3G_E_PRM_MAX_CCSSLR_SEGMENT
#### AMR payload formats: 1:=BW efficient 2:=octet aligned
# PRM: AMR_PAYLOAD_FORMAT 1     # M3G_E_PRM_AMR_PAYLOAD_FORMAT
PRM: PRM_VIDEO_BIT_RATE 42000  # M3G_E_PRM_VIDEO_BIT_RATE
#### Video frame rates: 0:=6fps 1:=10fps 2:=15fps
# PRM: PRM_VIDEO_FRAME_RATE 0   # M3G_E_PRM_VIDEO_FRAME_RATE
# PRM: PRM_EARLY_MES 1         # M3G_E_PRM_EARLY_MES
# PRM: PRM_AUTO_VFU_PERIOD 5    # M3G_E_PRM_AUTO_VFU_PERIOD
# PRM: PRM_H223_SYNC_TIMER 10000 # M3G_E_PRM_H223_SYNC_TIMER
#####

```

Event Notification

This parameter configuration section allows the user to specify the maskable 3G-324M library events reported to the application. Parameters in this section overwrite the default values.

The parameter format to enable the event is *ENEV*:<Event Name> and to disable the event is *DSEV*:<Event Name>

```

#####
# Event Notification:
# ENEV:=Enable DSEV:=Disable
#####
# ENEV: H245_UII_EVT          # H245 User Input Indication (UII) Digit received event.
# ENEV: FASTUPDATE_EVT       # H245 Fast Video Update (FVU) received event.
# ENEV: TEMP_SPAT_TRDFF_EVT  # H245 Temporal Spatial Tradeoff received event
# ENEV: VIDEO_FREEZE_EVT     # H245 Video Freeze received event
# ENEV: SYNC_GOB_EVT         # H245 Video Synch every GOB received event*
# ENEV: SKEW_INDICATION_EVT  # H245 Skew Indication received event
# ENEV: VERBOSE_MSD_EVT      # Master/Slave Determination event reporting
# ENEV: MES_EVTS_EVT         # MES event reporting
# ENEV: MONA_PREF_MSG_EVT    # MONA Preference message received event
# ENEV: REMOTE_VENDORID_EVT  # Remote Vendor ID received event
# ENEV: CALL_STATISTICS_EVT  # Call Statistics reporting
#Note: Events designated with * are disabled by default
#####

```

H.223 Session

This parameter configuration section allows the user to specify parameters related to the physical multiplex layer or H.223 session for each endpoint or group of endpoints. The parameter format for configuring the H.223 session is *HSES*:<endpoint/endpoint range> <Mux Level> <Max AL2 size> <WSNRP(T/F)> <Multiple Msgs(T/F)>

```

#####
# H223 Session:
# Default H223MuxLevel: L0 - Level0
#           L1 - Level1
#           L1D - Level1 Double Flags
#           L2 - Level2
#           L2H - Level2 Optional Header
#           L3 - Level3
#           L3H - Level3 Optional Header
# Endpoint Mux MaxAL2 WSNRP MultipleMsgs
#           Lvl Size (T/F) Per PDU (T/F)
#####

```

```
HSES: 1-2 L2 1024 T T
#####
```

H.223 Capabilities

This parameter configuration section allows the user to specify the H.223 capabilities for the endpoint or group of endpoints.

```
#####
# H223 Capabilities:
# Options:
# AL: AudioAL1 0x01 Bitrate: 320 or 640 (x 100bps)
# AudioAL2 0x02
# AudioAL3 0x04 MobileOpXmtCap: none: 0x00
# VideoAL1 0x08 mode change: 0x01
# VideoAL2 0x10 AnnexA: 0x02
# VideoAL3 0x20 AnnexADblFlag 0x04
# AnnexB 0x08
# MaxAL2Size:0-64k AnnexBwHeader 0x10
# MaxAL3Size:0-64k
# Endpoint AL AL2 AL3 AnxA Dbl AnxB AnxB RS MblOp Bit
# (Hex) Size Size Frame Flag Frame Hdr Code Xmt Rate
#####
HCAP: 1-2 12 512 512 T T T T T T 600
#####
```

Terminal Capabilities

This parameter configuration section allows the user to specify the Terminal Capabilities for each endpoint or group of endpoints. Terminal Capabilities are supported Audio/Video codecs designated by the endpoint for each 3G call. Possible values for terminal capabilities include AMR and G.723 for audio codecs and H.263 and MPEG-4 for video codec.

```
#####
# Terminal Capabilities:
#
# Endpoint Capabilities (amr g723 h263 mpeg4)
#####
MCAP: 1-2 amr h263
# MCAP: 1-2 amr mpeg4
#####
```

OpenLogicalChannel H.223 Parameters

This parameter configuration section allows the user to specify the OpenLogicalChannel Parameters for the H.223 physical multiplexing layer.

```
#####
# OpenLogicalChannel H223 Parameters:
#
# ALTypes: AL1_FRM
# AL1
# AL2_SEQ
# AL2
# AL3
#
# Endpoint Audio Video Segmentable Segmentable AL3 Control AL3 Send
# AL AL Audio Video Field Size Bufr Size
#####
```

```
HOLC: 1-2 AL2 AL2 F T 1 1024
#####
```

3G Tracing Level

This parameter configuration section allows the user to specify the level of runtime tracing and trace filename on a per channel or board basis when the 3G-324M gateway is run.

```
#####
# 3G Tracing Level:
# 0x0000 - disable
# 0x0001 - decoded H.245 messages
# 0x0002 - raw H.223 multiplex bitstreams
# 0x0004 - raw audio bitstreams
# 0x0008 - raw audio bitstreams
# 0x0010 - internal 3G module debug tracing
# 0x0020 - 3G session statistics
#
# Endpoints Level Filename
#####
ETRC: 1-2 31 ../log/ep_3g_trace.txt
#####
# Board Level Filename
#####
#BTRC: 3F bd_3g_trace.txt
#####
```

MPEG-4 DCI

This parameter configuration section allows the user to specify the MPEG-4 Decoder Configuration Information that is specified to the remote 3G endpoint.

```
#####
# MPEG-4 Decoder Configuration Information (Tx to 3G peer):
# Mode: 1 - in-band DCI
# 2 - out-of-band DCI via H.245 OLC
# 3 - both inband and out-of-band
#
# Endpoints Mode DCI
#####
#DCI: 1-2 3 000001b008000001b509000001000000012000845d4c282c2090a28f
#####
```

Vendor Identification

This parameter configuration section allows the user to specify the Vendor Information that is sent to the remote 3G endpoint in the VendorId message. The vendor ID can be specified through the object identifier or H.221 non-standard identifier.

```
#####
# Vendor Identification:
# Type: OID - object identifier
# H221 - H.221 NonStandard identifier
#####
### OID OID Product Version
```

```

###          Number  Number
VI: OID 4469616c6f676963 4d4d50 322e30
#### H221 T.35 T.35 manufacturer Product Version
#### country ext. code Number Number
#VI: H221 b5 00 00 4d4d50 322e30
#####

```

SIP Endpoint Parameters

This parameter configuration section allows the user to specify the IP addresses for outbound SIP calls. Note that all SIP endpoints use the same outbound address – that is, they are not individually settable.

Local SIP Addr – is the originating SIP address for the 3G-324M gateway application in the form <phone number>@<local ip address>:<sip udp port> or <local ip address> (where local default SIP port=5062)

Dest SIP Addr – is the destination SIP address for outgoing SIP call in the form <phone number>@<ip address>:<sip udp port>

```

#####
# SIP Endpoint Parameters
#
# Defines the SIP Local and Remote IP addresses -
# Stick with full IP address of the system (not local loopback)
# For MMP, that would be eth6
#
# Although there may be multiple SIP endpoints, they all share the same
# IP and SIP Local (origin) and Remote (destination) addresses.
# Local "Origin" is the 3g-SIP gateway address/port
# Remote "Destination" is the MMDemo or other SIP endpoint address/port.
# Can be same or different system.
#
# Local SIP Addr = <Phone#>@<Local IP Address>:<SIP Port> or
# <Local IP Address> (where local default SIP port=5062)
# Remote SIP Addr = <Phone#>@<Remote IP Address>:<SIP Port>
#
# Local SIP Addr Remote SIP Addr
#####
SIPINFO: 146.152.126.156 1000@146.152.126.156:5060
#####

```

SIP Coders

This parameter configuration section allows the user to specify the SIP Audio and SIP Video coders used for outbound SIP calls. SIP inbound calls are defined by the SIP sdp presented in the INVITE message.

Note: When SIP endpoints are set to N(NATIVE) for Audio XCode or Video XCode in the Endpoint (EP) line, the coders specified must match the 3G Terminal Capabilities coders specified in the (MCAP) line. When SIP endpoints are set to T(TRANSCODE) for Audio XCode or Video XCode in the Endpoint (EP) line, the 3G-324M gateway will transcode from media format specified for the SIP endpoint to the media format specified for the 3G endpoint..

```

#####
# SIP Coders

```

```

#
# SIPVID: h263 mpeg4
# SIPAUD: amr g711u g711a g723 g729
#
# This features is only supported on releases that include video transcoding
# (ie, MMP2.0, MMK...)
# Note: Used for sip outbound calls, sip inbound calls defined by SIP sdg
#   To enable transcoding: Audio Xcode and Video Xcode are set in EP line
#####
SIPVID: h263
SIPAUD: amr
#####

```

Command Information

This set of parameters is only used when an external ISDN (or SS7) signaling gateway is used to front-end calls into the 3G-324M gateway. Connect and disconnect commands are sent via a UDP socket to the address specified using the local ports given here.

If no gateway is used, but rather just an external loopback connection between two spans on the SIP-3G gateway system, specify the IP address of the gateway system.

If the internal loopback mode is used, this parameter will be ignored.

```

#####
# IP Address and Port Info for Connect/Disconnect Commands
#
#           Remote System IP Addr  Local Rcv Port  Local Send Port
#####
# Next line for local external loopback
#CMDINFO:   146.152.99.121         40121          40121
#
# Next line for ISDN gateway
#CMDINFO:   146.152.99.151         40121          40122
#####

```

Multimedia Endpoint Information

This set of parameters is used when a multimedia (MM) endpoint is paired with a 3G endpoint. It determines the function of the multimedia device when DTMF/UII <0-9> is received. The operation is either PLAY or RECORD. The MMINFO line allows the user to specify the Audio coder used in the audio file and Video Coder, Video Resolution, Video Frames per Sec, and Video Bit Rate of the the video file. Audio coder formats supported include amr, pcm, g711u, g711a, g723 and g729. Video Coder formats supported include h263 and mpeg4.

Only one operation per DTMF is allowed. The A/V file pairs need to be specified for the desired operation. The other two file names are ignored.

Note: When MM endpoints are set to N(NATIVE) for Audio XCode or Video XCode in the Endpoint (EP) line, the coders specified must match the 3G Terminal Capabilities coders specified in the (MCAP) line. When MM endpoints are set to T(TRANSCODE) for Audio XCode or Video XCode in the Endpoint (EP) line, the 3G-324M gateway will transcode from media format specified in MMINFO to the media format specified in the 3G endpoint.

```
#####
# Multimedia Endpoints
#
# Op - Operation (PLAY/RECORD) - what it does
# Digit - DTMF <0-9> which controls the Play/Record operation
#   - digit 0 must exist for default operation
# Aud - Audio codec in file (amr, pcm, g723...)
# Vid - Video codec in file (h263, mpeg4)
# VRes - Video Resolution in file (qcif, cif, sqcif)
# VFR - Video Frame Rate in fps (6, 7(=7.5fps), 10, 15, 25, 30)
# VBR - Video Bit Rate in kbps (ex. 40=40kbps, 256=256kbps)
# File Names - separate audio/video with unique names for play and record
#
# Digit Op  Aud Vid  VRes VFR VBR Audio Play File  Video Play File  Audio Rec File Video Rec File
#####
MMINFO: 0 PLAY  amr h263 qcif 10 40 main_menu_amr.aud main_menu_amr.vid recording.aud recording.vid
#####
```

RTSP Endpoint Information

This set of parameters is used when an RTSP endpoint is paired with a 3G endpoint. It determines the RTSP URL to request from the RTSP server when the DTMF/UII <0-9> is received. Upon receiving the digit, the RTSP endpoint will establish a session with the RTSP server, request the specified file and wait for RTP data. The RTSP server address and 3g media file can be different for each DTMF digit.

Note: When RTSP endpoints are set to N(NATIVE) for Audio XCode or Video XCode in the Endpoint (EP) line, the coder contained in the file on the RTSP server must match the 3G Terminal Capabilities coders specified in the (MCAP) line. When RTSP endpoints are set to T(TRANSCODE) for Audio XCode or Video XCode in the Endpoint (EP) line, the 3G-324M gateway will transcode from media format in the specified file to the media format specified for the 3G endpoint.

```
#####
# RTSP Endpoints
#
# Digit - Specifies the DTMF <0-9> used to command rtsp PLAY
# RTSP URL File - RTSP URL in the form rtsp://<RTSP Server Address>/<3gp filename>
#
# Note: When using 'Native' Xcode, coder in .3gp file must match m3g MCAP coder
#
#   Digit RTSP File URL
#####
RTSPINFO: 0 rtsp://146.152.130.190/media0.3gp
#RTSPINFO: 1 rtsp://146.152.130.190/media1.3gp
#RTSPINFO: 2 rtsp://146.152.130.190/media2.3gp
#RTSPINFO: 3 rtsp://146.152.130.190/media3.3gp
#RTSPINFO: 4 rtsp://146.152.130.190/media4.3gp
#RTSPINFO: 5 rtsp://146.152.130.190/media5.3gp
#RTSPINFO: 6 rtsp://146.152.130.190/media6.3gp
#RTSPINFO: 7 rtsp://146.152.130.190/media7.3gp
#RTSPINFO: 8 rtsp://146.152.130.190/media8.3gp
#RTSPINFO: 9 rtsp://146.152.130.190/media9.3gp
#####
```

Nb UP Information

This set of parameters must be set when Nb UP is used for a transport on a 3G endpoint. Each pair of 3G endpoints using Nb UP must have one Send and one Receive direction for the protocol to connect. IP addresses must be the same, as Nb UP is only supported in a loopback mode for the demo. Nb UP ports must correspond to the port of the remote ipm device, which start at 40960 for ipmB1C1 and increment by 2 for each ipm channel (ie, ipmB1C1=40960, ipmB1C2=40962, ipmB1C3=40964,...).

```
#####  
#  
# Only Needed if NBUP is used.  
#  
# Make sure endpoints are defined before this entry, as it refers back to them  
#  
#      3G Endpoint Nbuf Dir  Remote Nbuf IP Addr  Remote Nbuf  
#                               Port  
#####  
#  
# Here, Nbuf connection is on the same system, on ports 40960 and 40962  
#  
NBUPINFO:   1      S      192.168.1.117      40962  
NBUPINFO:   2      R      192.168.1.117      40960  
#####
```

3.3 Sample Configuration Files

A number of configuration files for various scenarios are contained in the sample_config_files directory. Additional scenarios are possible as well, but these samples have been tested, and should require little or no changes for use.

The platforms listed in the table are: Dialogic® Host Media Processing Software Release 3.1LIN (HMP 3.1), Dialogic® Multimedia Software for AdvancedTCA (MMP) Releases 1.1 and 2.0, and Dialogic® Multimedia Kit Software Release 1.0 for PCIe (MMK).

Table 1. Sample Configuration Files

Sample config file	Connection	Description	Platforms	Notes:
config_3g-hairpin-cc-isdngw.txt	Isdngw→dti[1](cc)→m3g[1]→ Isdngw←dti[2](cc)←m3g[2]←	3G Hairpin – Inbound 3G call bridged to outbound 3G call [clearchannel calls using isdngw]	HMP3.1(cc) MMP1.1 MMP2.0 MMK(cc)	Requires Isdngw
config_3g-mm-cc-isdngw.txt	Isdngw→dti(cc)→m3g→mm	3G Multimedia - Inbound 3G call [over clearchannel using isdngw] to mm	HMP3.1(cc) MMP1.1 MMP2.0 MMK(cc)	Requires Isdngw
config_3g-mm-isdn.txt	dti(isdn)→m3g→mm	3G Multimedia - Inbound 3G call [using isdn protocol] to mm	HMP3.1(isdn) MMK(isdn)	*not supported on MMP
config_3g-sip-cc-isdngw.txt	Isdngw→dti(cc)→m3g→sip	3G Gateway – Inbound 3G call [over clearchannel using isdngw] to outbound SIP call.	HMP3.1(cc) MMP1.1 MMP2.0 MMK(cc)	Requires Isdngw
config_3g-sip-isdn.txt	dti(isdn)→m3g→sip	3G Gateway – Inbound 3G call [using isdn protocol] to outbound SIP call.	HMP3.1(isdn) MMK(isdn)	*not supported on MMP
config_sipin-3g-hairpin-sipout-ccexternal.txt	←dti[1](cc)←m3g[1]←sip[1] →dti[2](cc)→m3g[2]→ ←dti[3](cc)←m3g[3]← →dti[4](cc)→m3g[4]→sip[2]	3G Hairpin (loopback)– Inbound SIP call, external clearchannel loopback with 3G bridging, outbound SIP call	HMP3.1(cc) MMP1.1 **MMP2.0 MMK(cc)	Requires external cross-over cable **not supported with MMP2.0 advanced video licenses
config_sipin-3g-hairpin-sipout-isdnexternal.txt	←dti[1](isdn)←m3g[1]←sip[1] →dti[2](isdn)→m3g[2]→ ←dti[3](isdn)←m3g[3]← →dti[4](isdn)→m3g[4]→sip[2]	3G Hairpin (loopback)– Inbound SIP call, external isdn loopback with 3G bridging, outbound SIP call	HMP3.1(isdn) MMK(isdn)	*not supported on MMP
config_sipin-mm-ccexternal.txt	←dti[1](cc)←m3g[1]←sip →dti[2](cc)→m3g[2]→mm	3G Multimedia (loopback) - Inbound SIP call, 3G external clearchannel loopback, to mm	HMP3.1(cc) MMP1.1 **MMP2.0 MMK(cc)	Requires external cross-over cable **not supported with MMP2.0 advanced video licenses

Sample config file	Connection	Description	Platforms	Notes:
config_sipin-mm-internal.txt	←m3g[1]←sip →m3g[2]→mm	3G Multimedia (loopback) - Inbound SIP call, 3G internal loopback, to mm	HMP3.1 MMP1.1 MMP2.0 MMK	
config_sipin-mm-isdnexternal.txt	←dti[1](isdn)←m3g[1]←sip →dti[2](isdn)→m3g[2]→mm	3G Multimedia (loopback) - Inbound SIP call, 3G external isdn loopback, to mm	HMP3.1(isdn) MMK(isdn)	Requires external cross-over cable *not supported on MMP
config_sipin-mm-nbup.txt	←ipm-nbup[1]←m3g[1]←sip →ipm-nbup[2]→m3g[2]→mm	<u>3G Multimedia (loopback)</u> - Inbound SIP call, 3G NbUP loopback, to mm	HMP3.1 MMP1.1 MMP2.0 MMK	
config_sipinout-cc-isdngw.txt	Isdngw←dti[1](cc)←m3g[1]←sip [1] Isdngw→dti[2](cc)→m3g[2]→sip [2]	3G Gateway – Inbound SIP call to outbound 3G call. Inbound 3G call to outbound SIP call.	HMP3.1(cc) MMP1.1 **MMP2.0 MMK(cc)	Requires Isdngw
config_sipin-sipout-ccexternal.txt	←dti[1](cc)←m3g[1]←sip[1] →dti[2](cc)→m3g[2]→sip[2]	3G Gateway (loopback) – Inbound SIP call, 3G external clearchannel loopback, to outbound SIP call.	HMP3.1(cc) MMP1.1 **MMP2.0 MMK(cc)	Requires external cross-over cable **not supported with MMP2.0 advanced video licenses
config_sipin-sipout-internal.txt	← m3g[1]←sip[1] → m3g[2]→sip[2]	3G Gateway (loopback) – Inbound SIP call , 3G internal loopback, to outbound SIP call.	HMP3.1 MMP1.1 MMP2.0 MMK	
config_sipin-sipout-isdnexternal.txt	←dti[1](isdn)←m3g[1]←sip[1] →dti[2](isdn)→m3g[2]→sip[2]	3G Gateway (loopback) – Inbound SIP call , 3G external isdn loopback, to outbound SIP call.	HMP3.1(isdn) MMK(isdn)	Requires external cross-over cable *not supported on MMP
config_sipin-sipout-nbup.txt	←ipm-nbup[1]←m3g[1]←sip[1] →ipm-nbup[2]→m3g[2]→sip[2]	3G Gateway (loopback) – Inbound SIP call , NbUP 3G loopback, to outbound SIP call.	HMP3.1 MMP1.1 MMP2.0 MMK	
config_3g-rtsp-isdn.txt	dti(isdn)→m3g→rtsp	3G RTSP Streaming - Inbound 3G call [using isdn protocol] to rtsp	HMP3.1(isdn) MMK(isdn)	*not supported on MMP
config_sipin-rtsp-internal.txt	←m3g[1]←sip →m3g[2]→rtsp	3G RTSP Streaming (loopback) - Inbound SIP call, 3G internal loopback, to mm	HMP3.1 MMP1.1 MMP2.0 MMK	
config_sipin-rtsp-isdnexternal.txt	←dti[1](isdn)←m3g[1]←sip →dti[2](isdn)→m3g[2]→rtsp	3G RTSP Streaming (loopback) - Inbound SIP call, 3G external isdn loopback, to rtsp	HMP3.1(isdn) MMK(isdn)	Requires external cross-over cable *not supported on MMP
config_sipin-rtsp-ccexternal.txt	←dti[1](cc)←m3g[1]←sip →dti[2](cc)→m3g[2]→rtsp	3G RTSP Streaming (loopback) - Inbound SIP call, 3G external clearchannel loopback, to rtsp	HMP3.1(cc) MMP1.1 **MMP2.0 MMK(cc)	Requires external cross-over cable **not supported with MMP2.0 advanced video

Sample config file	Connection	Description	Platforms	Notes:
				licenses
config_sipin-rtsp-nbup.txt	←ipm-nbup[1]←m3g[1]←sip →ipm-nbup[2]→m3g[2]→rtsp	3G RTSP Streaming (loopback) _ Inbound SIP call, 3G NbUP loopback, to rtsp	HMP3.1 MMP1.1 MMP2.0 MMK	

4 Compiling and Linking

This section contains the following information:

- [Demo Compilation Options](#)
- [Compiling the Demo](#)
- [Modifying the Demo](#)

4.1 Demo Compilation Options

The 3G-324M Multimedia Gateway Demo application comes with the compilation parameters that will allow it to run on all Dialogic® 3G-324M enabled platforms. There are two features that can be enabled in the Makefile which add feature capability when the capabilities are supported on the particular platform. The features that are disabled by default, but can be enabled through the Makefile are:

1. Dialogic® 3G-324M API Media Oriented Negotiation Acceleration (MONA) procedures
2. RTSP Endpoint Support

4.1.1 Enabling Dialogic® 3G-324M API MONA Procedures Support

This chapter provides a description of the Dialogic® 3G-324M Multimedia Gateway Demo. The following topics are covered:

The 3G-324M Demo provides the possibility of using Dialogic® 3G-324M Media Oriented Negotiation Acceleration (MONA) procedures to establish a 3G-324M session. MONA procedures can significantly reduce the call setup time of a 3G-324M call if both endpoints support MONA. Dialogic® 3G-324M API is a MONA Class II terminal which supports MPC and ACP MONA procedures.

The default behavior of the 3G-324M Gateway demo is to function without MONA support. The Makefile includes a flag for allowing the demo to operate with MONA procedures enabled. This flag is set in Makefile by the USE_MONA variable.

To use MONA: uncomment the line, so that line is: 'USE_MONA=1'.

To disable MONA: comment out the line: '#USE_MONA=1'.

4.1.2 Enabling RTSP Endpoint Support

The 3G-324M Demo provides the possibility of connecting to an RTSP server to stream live content to a 3G handset. In order to enable the RTSP client capability, the demo needs to be compiled with the open-source RTSP stack from Live Networks Inc. Below is the procedure for installing the RTSP stack and enabling RTSP Endpoint support through the demo Makefile.

Installing RTSP Stack

Installation of the RTSP Stack can be done as follows:
Download Live Networks, Inc.'s stack source-code from
<http://www.LIVE555.com/liveMedia/public/>

```
Uncompress the downloaded archive into a directory. For example,  
mkdir /export/RTSP  
cd /export/RTSP  
tar -xvzf live.2008.07.25.tar.gz
```

```
Files should be uncompressed into /export/RTSP/live after step above  
cd live  
./genMakefiles linux  
make
```

Makefile changes

Once the RTSP Stack is installed on the system, update location of LIVE555 stack source code. The Makefile contains an environment variable for the location of LIVE555 stack. The variable is LIVE555. Assuming the stack was installed according to the instructions of the previous section, in /export/RTSP/live directory, modify the Makefile so that:

```
LIVE555 = /export/RTSP/live
```

The default behavior of the 3G-324M Gateway demo is to function without RTSP support. The Makefile includes a flag for allowing the demo to operate in RTSP mode. This flag is set in Makefile by the USE_RTSP variable.

To use RTSP: uncomment the line, so that line is: 'export USE_RTSP=1'.

To disable RTSP: comment out the line: '#export USE_RTSP=1'.

4.2 Compiling the Demo

To compile the source code for the Dialogic® 3G-324M Multimedia Gateway Demo application, do the following:

1. Go to this location: `${DIALOGIC_DIR}/demos/3GDemo/m3g-sip_gateway`
2. At the command prompt, type `make clean; make`

4.3 Modifying the Demo

If you have added or changed files, you can do the following to compile the project:

1. Put the files in `${DIALOGIC_DIR}/demos/3GDemo/m3g-sip-mm_gateway`
2. Modify the Makefile to take into account the new files generated by adding the filename.o to the OBJS variable
3. At the command prompt, type `make clean; make`

Successful compilation will result in an executable called `m3g-sip_gateway` at the following location: `${DIALOGIC_DIR}/demos/3GDemo/m3g-sip_gateway/Release`.

5 Running the Demo

This section describes how to start the Dialogic® 3G-324M Multimedia Gateway Demo and contains the following information:

- Preparation
- Run Run Procedure
- Exiting the Demo
- Sample Output

5.1 Preparation

Before running the demo, be sure to follow the instructions provided in [Chapter 3, Preparing to Run the Demo](#).

In a normal installation, the Dialogic® Multimedia Platform for AdvancedTCA, the Dialogic® Multimedia Kit for PCIe, or the Dialogic® Host Media Processing Software is automatically started during system boot-up.

5.2 Run Procedure

To start the 3G-324M Demo, you can do the following:

1. Launch a terminal window.
2. Change to the following directory:
`${DIALOGIC_DIR}/demos/3GDemo/m3g-sip_gateway/Release.`
3. Modify the configuration file, `config.txt`, to support the desired configuration scenario.
 - a. Alternatively, copy one of the sample configuration files from the `${DIALOGIC_DIR}/demos/3GDemo/m3g-sip_gateway/sample_configuration_files/` directory into the Release directory and modify it with the proper settings.
4. At the command prompt, type `./m3g-sip_gateway` or alternatively, `./m3g-sip_gateway -i <sample_config_file_name>`, to use a config file other than `config.txt`.

The 3G-324M gateway application will be ready for incoming SIP and 3G calls.

5.2.1 Sample Configuration Scenario Examples

This section gives a few examples of how the sample configuration files can be edited to create the scenarios discussed in this document. These examples show the minimum configuration necessary for the scenario. Other configuration options may be necessary depending on the target system, network and usage.

3G-SIP Gateway (ISDN) Scenario:

1. Start in the `${DIALOGIC_DIR}/demos/3GDemo/m3g-sip_gateway/Release` directory.

2. Copy the sample 3g-sip-isdn config file to Release directory.

```
cp ../sample_config_files/config_3g-sip-isdn.txt .
```

3. Edit the config file to change SIPINFO: to set the Local and Destination SIP Addresses.

```
SIPINFO: <System IP Address> 1000@<Dest. SIP Address>:5060
```

4. Run the scenario.

```
./m3g-sip_gateway -i config_3g-sip-isdn.txt
```

The 3G-324M gateway application will be ready for incoming SIP and 3G calls.

3G-MM Multimedia (Internal Loopback) Scenario:

1. Start in the `${DIALOGIC_DIR}/demos/3GDemo/m3g-sip_gateway/Release` directory.

2. Copy the sample 3g-sipin-mm config file to Release directory

```
cp ../sample_config_files/config_3g-sipin-mm-internal.txt .
```

3. Edit the config file to change SIPINFO: to set the Local and Destination SIP Addresses. Destination IP address must be set, but is not used in this scenario, so set it to `<System IP Address>`

```
SIPINFO: <System IP Address> 1000@<System IP Address>:5060
```

4. Set the Multimedia file to Play file (`manin_menu_amr`) for DTMF 0

```
MMINFO: 0 PLAY amr h263 qcif 10 40 main_menu_amr.aud main_menu_amr.vid recording.aud recording.vid
```

5. Run the scenario.

```
./m3g-sip_gateway -i config_3g-mm-internal.txt
```

The 3G-324M gateway application will be ready for incoming SIP call to port 5062. Upon accepting a SIP call, the demo application will make an outbound 3G call to itself over an internal connection to the 3G-MM pair.

3G-RTSP (ISDN) Scenario:

1. Start in the `${DIALOGIC_DIR}/demos/3GDemo/m3g-sip_gateway/Release` directory.

2. Copy the sample 3g-sip-isdn config file to Release directory

```
cp ../sample_config_files/config_3g-rtsp-isdn.txt .
```

3. Edit the config file to change SIPINFO: to set the Local and Destination SIP Addresses. Destination IP address must be set, but is not used in this scenario, so set it to <System IP Address>

```
SIPINFO: <System IP Address> 1000@<System IP Address>:5060
```

4. Set the RTSP Server URL of the Play file for DTMF 0

```
RTSPINFO: 0 rtsp://<RTSP Server Address>/media0.3gp
```

5. Run the scenario

```
./m3g-sip_gateway -i config_3g-rtsp-isdn.txt
```

The 3G-324M gateway application will be ready for incoming 3G calls.

5.3 Exiting the Demo

Stopping the demo is done through the command line. Enter a “Q” to stop the demo. This is best done when all devices and channels are in an idle or inactive state. If the demo fails to shut down and exit after 10-15 seconds, two Ctrl-C’s should be issued. This will force the 3G devices to reset and exit cleanly. If the demo still does not exit, four additional Ctrl-C’s should be issued. This will abort the demo process. If this is done, it may be necessary to reset the terminal window with a “reset” command and stop and start the Dialogic® services with “dlstop” and “dlstart”.

5.4 Sample Output

The example below shows the sample display output for a pair of 3G devices (endpoint 3G 1 & 2) configured to bridge to a pair of SIP devices (endpoint SIP 3 & 4). The 3G devices are both armed and ready to synchronize to H.223 data after indication of an available call.

One SIP device is idle and in a wait-for-call state, prepared to take an inbound SIP call. The second SIP device is in an idle state, ready to make an outbound call.

Two bridge devices (device BRG 1 & 2) display the state machine that coordinates its 3G device and SIP device.

```
=====
3G-324M - SIP GATEWAY DEMO
=====
STATE:  AUDIO:      VIDEO:      MISC.
TxLCN: RxLCN: CAP: TxLCN: RxLCN: CAP:  CMD/IND:
=====
M3G 1: H245_INACTIVE  0  0  none  0  0  none
```

```
M3G 2: H245_INACTIVE 0 0 none 0 0 none
SIP 3: IDLE          0 0 n/a 0 0 n/a
SIP 4: IDLE          0 0 n/a 0 0 n/a
BRG 1: IDLE          M3G:1 <-> SIP:3
BRG 2: IDLE          M3G:2 <-> SIP:4
```

PROMPT: (Q)uit

5.5 Using the Demo

This section describes how to use the 3G-324M Demo. First, start the demo as described in [section 5.2, Run Procedure](#).

5.5.1 Connecting a Call

Once the 3G-324M Demo is started, it is armed and ready to receive 3G calls on the 3G control devices connected to the DS0 timeslots specified by the DTI streaming devices. If SIP endpoints are used, they are also ready to receive calls via the system's Ethernet connection. Depending on the gateway's configuration, incoming calls will be directed to:

- An outbound 3G/SIP endpoint pair, resulting in an outbound SIP call
- A multimedia endpoint, resulting in a audio/video file play or record
- A 3G endpoint pair, resulting in an outbound 3G call

5.5.2 Disconnecting a Call

To stop the 3G-324M Demo during a call, simply hang up the 3G mobile device or SIP phone. Endpoints and bridged objects will return to inactive or idle states.

Appendix

Using the Kapanga SIP Softphone with the Dialogic® 3G-324M Multimedia Gateway Demo

The Kapanga SIP softphone has a configuration file, normally named default.cfg, that can be modified to work with the Dialogic 3G-324M Multimedia Gateway Demo. Notable differences beyond the default configuration are:

- Direct IP to IP (no SIP proxy)
- Video rate control on
- Video rate set to 30Kbps
- Display real time bandwidth and MOS
- QCIF resolution set
- Audio/Video codecs set to AMR/H.263
- SIP port of target system (3G-324M Gateway) = 5062

Using SIPp with the Dialogic® 3G-324M Multimedia Gateway Demo

SIPp may be downloaded from http://sourceforge.net/project/showfiles.php?group_id=104305%20. Follow the instructions for building the utility on Linux.

Two scripts and a set of media files may be downloaded from the Dialogic FTP site in furtherance of SIPp use. The Helicopters media files contain a short A/V clip in PCAP format.

The scripts are for configuring and starting SIPp. The “runuas” script invokes videouas.xml - SIPp as a user agent server:

- Waits for a SIP call
- Accepts a call
- Plays an audio and video stream from the two PCAP files specified
- Holds the call until disconnected from far end

The “runuac” script invokes videouac.xml - SIPp as a user agent client:

- Places an outbound SIP call
- On answer, plays an audio and video stream from two PCAP files
- Pauses while files play
- Hangs up

IP addresses must be changed in runuac and runuas to that of the 3G-324M gateway system. The scripts expect to be run in the same directory as the SIPp executable.

ISDN to ClearChannel Gateway

This section suggests one method to provide an ISDN to ClearChannel gateway needed for Dialogic® Multimedia Platform for AdvancedTCA (MMP) connection to ISDN.

The MMP provides a ClearChannel TDM connection (i.e., bearer channel connection) for “Media only” traffic through the RTM Module. Incoming call control indication from the 3G PSTN network is normally done with ISDN or ISUP (SS7) signaling and depends on the 3G PSTN network connection. This section describes one example method to create a TDM signaling entity for an ISDN connection by detailing an ISDN to ClearChannel gateway. This example is provided as a quick method to provide incoming call indication to the Dialogic® 3G-324M Multimedia Gateway application.

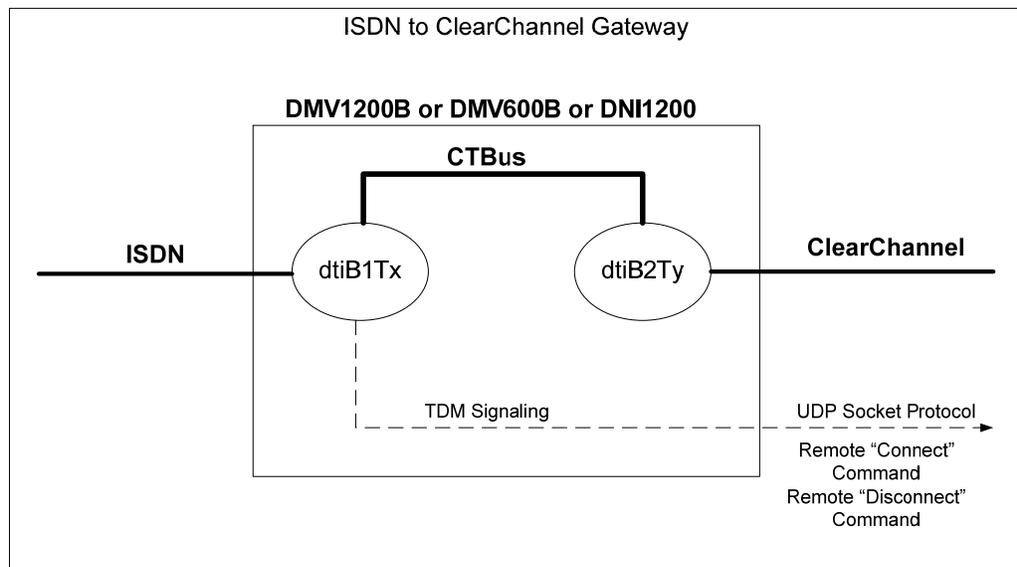


Figure 10. ISDN to ClearChannel Gateway Configuration

An ISDN to ClearChannel gateway handles ISDN protocol on one trunk and makes a “media only” connection to ClearChannel on a second trunk. The ISDN to ClearChannel Gateway configuration is described in Figure 9: “ISDN to ClearChannel Gateway Configuration” above.

An ISDN to ClearChannel Gateway can be configured, for example, by setting up a Dialogic® HMP Interface Board (DNI Board) or a Dialogic® DM3 Media Board with E1 ISDN signaling on trunk 1, E1 ClearChannel on trunk 2, and the CT Bus configured as A-Law. Alternatively, the board can be set up in a T1 environment with T1 ISDN signaling on trunk 1, T1 ClearChannel on trunk 2, and the CT Bus configured as mu-Law. Using the Dialogic® Global Call API for PSTN network interface connection, an incoming call on the ISDN trunk 1 (dtiB1Tx) is routed through the CTBus to the ClearChannel port on trunk 2 (dtiB2Ty) with the appropriate “listen” commands. For call indication purposes, a UDP Socket protocol, found in the Appendix , “UDP Socket Protocol” can be used to send

remote “Connect” and “Disconnect” command messages to the 3G-324M Gateway application. For example code to demonstrate the Global Call model, please see the `gc_basic_call_model` demo code provided with the Dialogic System Release. An example ISDN to ClearChannel application can be downloaded from the Dialogic FTP site for use with the gateway.

UDP Socket Protocol

This section identifies one example method for providing incoming call signaling identification from a remote signaling gateway to the Dialogic® 3G-324M Multimedia Gateway Demo (3G-324M Demo).

TDM SS7 or ISDN call signaling is beyond the scope of the 3G-324M Demo. Therefore, the 3G-324M Demo provides a UDP socket command method for indicating an incoming TDM call and initiating the 3G-324M protocol. The UDP socket command protocol can be installed at the SS7 signaling entity or at the ISDN to ClearChannel gateway entity. On an incoming call, a remote “connect” message is sent to the 3M-324M Demo to signify that an incoming call has been connected to a particular timeslot. The UDP socket commands provide a remote network indication of the incoming connect or disconnected call similar to a user keyboard indication. The format of UDP socket command for Connect is [`'C <Map ID> <calling number>:<called number>'`] and the format of the disconnect command is [`'D <Map ID>'`]. This is a useful method to indicate an incoming connect or disconnect on a particular DS0. Sample code for this socket command is included below.

```
// Remote Mux3G Gateway Macro Declarations
#define APP_CMD_PORT      40121      // cmd port used by M3G
Gateway
#define REMOTE_HOST      "146.152.80.84" // address of M3G Gateway

//To Connect
// send connect command to Mux3G Gateway
char command[80];
sprintf(command, "C %s %s:%s",pline->devname, "1111111111",
"2222222222");
sendCmdToMux3G(command);

//To disconnect
// send disconnect command to Mux3G Gateway
char command[80];
sprintf(command, "D %s %s",pline->devname);
sendCmdToMux3G(command);

/*****
*****
*   NAME: sendCmdToMux3G
*   DESCRIPTION: Send a UDP Packet to the Mux3G Gateway
*                 to command the gateway to perform an action.
*   INPUT: cmd - command as string
*   RETURNS: none
*****
*****/
void sendCmdToMux3G(char* cmd)
{
    static int initialized = 0;
    static int sockHand;
```

```

struct sockaddr_in server;
int  length;
int  buflen;

if (! initialized)
{
    sockHand = socket(AF_INET ,SOCK_DGRAM , IPPROTO_UDP);
    if (sockHand < 0)
    {
        printandlog(ALL_DEVICES, MISC_WITH_NL, NULL,
"sendCmdToMux3G() socket() failure", 0);
        return;
    }
    initialized = 1;
}

server.sin_family = AF_INET;
server.sin_addr.s_addr = inet_addr(REMOTE_HOST);
server.sin_port = htons(APP_CMD_PORT);
length = sizeof(server);

buflen = strlen(cmd) + 1;

if (sendto(sockHand,
    cmd,
    buflen,
    0,
    (struct sockaddr *)&server,
    length) != buflen)
{
    printandlog(ALL_DEVICES, MISC_WITH_NL, NULL,
"sendCmdToMux3G() sendto() failure", 0);
}
}

```