Dialogic® Multimedia Kit Software Release 1.0 for PCIe

Release Guide

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Contents

Document Revision History 5

About This Publication 7

Applicability 7
Intended Audience 7
How to Use This Publication 7
Related Information 7

1 Release Overview 9

2 System Requirements 10

2.1 Basic Hardware Requirements 10
2.2 Basic Software Requirements 11

3 Features 12

3.1 Dialogic® Multimedia Accelerator Boards for PCIe 12
3.2 Dialogic® HMP Interface Boards 13
3.3 SS7 Support 13
3.4 3G-324M Interface 14
3.5 RTP Interface 15
3.6 Channel Density Support 17
3.7 Multimedia (Audio/Video) Play and Record 18
3.8 Multimedia (Audio/Video) Transcoding 20
3.9 Multimedia (Audio/Video) Conferencing 21
3.10 Image Overlay 22
3.11 Native RTP Hairpinning 22
3.12 IP Signaling 22
3.13 Tone Management 23
3.14 Remote Media Control Interface 23
3.15 Audio (Voice) Play and Record 23
3.16 OA&M Support Using CLI and SNMP 24

4 Programming Libraries 25

4.1 Dialogic® 3G-324M API Library 25
4.2 Dialogic® Conferencing (CNF) API Library 26
4.3 Dialogic® Device Management API Library 27
4.4 Dialogic® Digital Network Interface API Library 28
4.5 Dialogic® Fax API Library 28
4.6 Dialogic® Global Call API Library 28
4.7 Dialogic® IP Media API Library 29
4.8 Dialogic® Media Toolkit API Library 30
4.9 Dialogic® Multimedia API Library 32
4.10 Dialogic® Standard Runtime API Library 33
4.11 Dialogic® Voice API Library 34
4.12 Dialogic® Demonstration Programs 35

5 **Supported Hardware** 37
5.1 Dialogic® Multimedia Accelerator Boards for PCIe 37
5.2 Dialogic® HMP Interface Boards 37
5.3 Dialogic® SS7 Boards 37
6 **Documentation** 38
6.1 Documentation Feature Support 38
6.2 Release Documentation 40
6.3 Installation and Configuration Documentation 40
6.4 OA&M Documentation 41
6.5 Programming Libraries Documentation 41
6.6 Application Scenario Documentation 41
6.7 Demonstration Software Documentation 42

Document Revision History

This revision history summarizes the changes made in each published version of the Release Guide for Dialogic® Multimedia Kit Software Release 1.0 for PCIe, which is a document that may be subject to updates during the lifetime of the release.

Document Rev 02 – published October 2008

Updated for Service Update 107.

In the Features chapter:

- Added SS7 Support.
- Added support for MONA in 3G-324M Interface.
- Removed Important Usage Guidelines information for 3G-324M connections in 3G-324M Interface, as this inforrmation is included in the new Dialogic® 3G-324M API Programming Guide and Library Reference.
- Added support for 2 to 10 frames per packet for AMR-NB 20 ms frames in Audio Codecs for RTP. Previously only 1 frame per packet was supported.
- Added support for H.263-1998 (H.263+) in Video Codecs for RTP.
- Added Codec for Nb UP table in Channel Density Support; includes new AMR-NB, new G.711, and existing 3G-324M.
- Added support for record/capture a still image and DVR controls in Multimedia (Audio/Video) Play and Record.

In the System Requirements chapter:

- Added optional SS7 boards to Basic Hardware Requirements.

In the Programming Libraries chapter:

- Added support for MONA in Dialogic® 3G-324M API Library.
- Added support for record/capture a still image and DVR controls in Dialogic® Multimedia API Library.

In the Supported Hardware chapter:

- Added Dialogic® SS7 Boards.

In the Documentation chapter:

- Added Dialogic® 3G-324M API Programming Guide and Library Reference (combined programming guide and library reference; supercedes the 3G-324M API Library Reference).
- Added Dialogic® Global Call SS7 Technology Guide.

Document Rev 01 – published September 2008

Initial version of document for Dialogic® Multimedia Kit Software Release 1.0 for PCIe.

About This Publication

The following topics provide information about this publication:

- Applicability
- Intended Audience
- How to Use This Publication
- Related Information

Applicability

This document provides information about the features, system requirements, and release documentation for the Dialogic® Multimedia Kit Software Release 1.0 for PCIe (MMK Software 1.0 for PCIe).

Intended Audience

This document is intended for all users of MMK Software 1.0 for PCIe.

How to Use This Publication

The information in this document is organized into the following sections:

- Chapter 1, "Release Overview" describes the highlights of this release.
- Chapter 2, "System Requirements" describes the hardware and software requirements for this release.
- Chapter 3, "Features" describes the features supported in this release.
- Chapter 4, "Programming Libraries" describes the various development software libraries and demonstration programs that are available as part of this release.
- Chapter 5, "Supported Hardware" lists the hardware supported in this release.
- Chapter 6, "Documentation" provides a list of the documents that accompany this release.

Related Information

See the following for additional information:

- Dialogic® Multimedia Kit Software Release 1.0 for PCIe Release Update for information about known problems, resolved problems, and documentation updates associated with this release. Refer to the Release Update for changes or corrections to the release information. Information is intended to be updated in the Release Update, as needed, during the life cycle of the release.

- *http://www.dialogic.com/manuals/* (for Dialogic® product documentation)
- *http://www.dialogic.com/support/helpweb/* (for Dialogic technical support)
- *http://www.dialogic.com/* (for Dialogic® product information)

Release Overview 1

The Dialogic® Multimedia Kit for PCIe consists of Dialogic® Multimedia Accelerator Boards for PCIe (MMA Boards), which are used for transcoding offload, and associated software. The first software release is the Dialogic® Multimedia Kit Software Release 1.0 for PCIe.

The Dialogic® Multimedia Kit Software Release 1.0 for PCIe provides up to 1000 ports of voice processing functionality or up to 240 ports of multimedia processing functionality. It supports two direct APIs: Dialogic® R4 API for media processing and Dialogic® Global Call API for call control. A standards-based Media Server Markup Language (MSML) interface is also included, providing a rich set of media management capabilities and allowing media processing on the multimedia server from a remote agent such as an application server.

The Dialogic® Multimedia Kit for PCIe supports the industry-standard Session Initiation Protocol (SIP) for call control, with the Real-time Transport Protocol (RTP) and Real-Time Control Protocol (RTCP) for media streaming and control over IP in various audio formats and two video formats.

The Dialogic® Multimedia Kit for PCIe provides media features for voice over IP (VoIP) and for PSTN endpoints. It supports video telephony capabilities based on the 3GPP 3G-324M specification, and advanced video features such as video transcoding, video transrating, image resizing, MPEG-4 play and record, multimedia (audio/video) conferencing, and image overlay.

The Dialogic® Multimedia Kit for PCIe supports Dialogic® HMP Interface Boards (DNI boards) that enable connectivity to PSTN endpoints.

OA&M capabilities are provided through an SNMP interface and a Command Line Interface (CLI).

The Dialogic® Multimedia Kit for PCIe is available in a variety of hardware and media software configurations. For ordering information, see the Dialogic® Multimedia Kit for PCIe data sheet.

Refer to Chapter 3, “Features” for further information about the supported features in this release.

System Requirements 2

This chapter describes the hardware and software requirements for Dialogic® Multimedia Kit Software Release 1.0 for PCIe (MMK Software 1.0 for PCIe).

- Basic Hardware Requirements 10
- Basic Software Requirements. 11

2.1 Basic Hardware Requirements

The basic hardware requirements for this release are:

Rank Mount Server or equivalent PC

- Memory: Minimum of 2 GB of RAM; 4 GB of RAM recommended.
- CPU: Minimum Single Socket Dual-Core Intel Xeon 5080 processor with Hyperthreading (3.73 GHz);
  Dual Socket Quad-Core Intel Xeon X5355 processor (2.67 GHz) or better recommended.
- Disk Space: 500 MB for full installation. Additional disk space may be required depending on multimedia recording needs.
- PCIe slots: half length, full height slot for MMA Boards;
  full length, full height slot for DNI boards

Dialogic® Multimedia Accelerator Boards for PCIe (MMA Boards), one of the following:

- Dialogic® MMAC250PCIEQ Multimedia Accelerator Board for PCIe
- Dialogic® MMAC500PCIEQ Multimedia Accelerator Board for PCIe

Note: Only one MMA Board is supported in a system.

Optional Dialogic® HMP Interface Boards (DNI boards):

- Dialogic® DNI310TEPEHMPQ Digital Network Interface Boards
- Dialogic® DNI610TEPEHMPQ Digital Network Interface Boards
- Dialogic® DNI1210TEPEHMPQ Digital Network Interface Boards
- Dialogic® DNI2410TEPEHMPQ Digital Network Interface Boards

Note: Up to 480 ports of PSTN interface are supported in a system.

Optional Dialogic® SS7 Boards:

- Dialogic® SPCI2S SS7 Board
- Dialogic® SPCI4 SS7 Board
- Dialogic® SS7HDP SS7 Board

2.2 Basic Software Requirements

The following software is supported in this release:

Supported Compilers

- GNU Compiler Collection (GCC) Versions 3.2.3 and 3.4.3

Note: Development tools such as GCC must be installed on your system, even in a runtime installation. MMK Software 1.0 for PCIe uses the GCC to compile certain Dialogic drivers.

Supported Operating Systems

- Red Hat Enterprise Linux Release 4.0 with Update 5 or Update 6 (Advanced Server, Enterprise Server, or Workstation)
- SUSE Linux Enterprise Server 9 Service Pack 4

Note: Only the 32-bit version of the operating systems are supported.

Note: For important information about additional operating system and other requirements, follow the instructions provided in the *Dialogic® Multimedia Kit Software Release 1.0 for PCIe Software Installation Guide.*

Note: A license is required to use MMK Software 1.0 for PCIe. For more information, see *Dialogic® Multimedia Kit Software Release 1.0 for PCIe Software Installation Guide.*

Features 3

This chapter describes the features that are supported in Dialogic® Multimedia Kit Software Release 1.0 for PCIe (MMK Software 1.0 for PCIe).

- Dialogic® Multimedia Accelerator Boards for PCIe 12
- Dialogic® HMP Interface Boards 13
- SS7 Support 13
- 3G-324M Interface 14
- RTP Interface 15
- Channel Density Support 17
- Multimedia (Audio/Video) Play and Record 18
- Multimedia (Audio/Video) Transcoding 20
- Multimedia (Audio/Video) Conferencing 21
- Image Overlay. 22
- Native RTP Hairpinning 22
- IP Signaling. 22
- Tone Management 23
- Remote Media Control Interface. 23
- Audio (Voice) Play and Record 23
- OA&M Support Using CLI and SNMP 24

3.1 Dialogic® Multimedia Accelerator Boards for PCIe

The Dialogic® Multimedia Accelerator Boards for PCIe (MMA Boards) are sold as a bundle with the MMK Software 1.0 for PCIe.

The MMA Boards provide transcoding offload and come with two ethernet ports in a half-length, full height PCI Express form factor. For a list of currently supported MMA Boards, see Section 5.1, “Dialogic® Multimedia Accelerator Boards for PCIe”, on page 37. For technical specifications, see http://www.dialogic.com/products.

For information on configuring MMA Boards, see the *Dialogic® System Configuration Guide*.

3.2 Dialogic® HMP Interface Boards

Dialogic® HMP Interface Boards (DNI boards) provide a native public switched telephone network (PSTN) interface to Dialogic® Host Media Processing (HMP) Software, on which MMK Software 1.0 for PCIe is based.

Built on Dialogic® DM3 architecture, DNI boards enable applications developed using Dialogic® API libraries to connect to the PSTN through T1 or E1 network interfaces.

Each board includes an H.100 CT Bus connector that enables media streaming between Dialogic® software and the CT Bus. A host streaming interface allows DNI boards to communicate with Dialogic® software running on the host via bridge devices.

The Dialogic® Global Call API provides call control functionality on PSTN interfaces. For E1, T1 and ISDN technologies, the libdm3cc.dll library provides this functionality and is dynamically loaded, by specifying GC_DM3CC_LIB when calling the **gc_Start( )** function.

For a list of currently supported boards, see Section 5.2, "Dialogic® HMP Interface Boards", on page 37. For technical specifications, see http://www.dialogic.com/products/ip_enabled/hmp_enabled_boards.htm.

For information on configuring DNI boards, see the *Dialogic® System Configuration Guide*.

3.3 SS7 Support

MMK Software 1.0 for PCIe supports Signaling System 7 (SS7) over IP and TDM interfaces.

Support for SS7 over IP is provided via Signaling Interface Units (SIU) and SIGTRAN (IETF SS7 Signaling over IP). The Dialogic® system provides SIU/SIGTRAN detection, initialization and configuration support.

Support for SS7 over TDM is provided via optional Dialogic® SS7 boards. Dialogic® SS7 boards provide on-board support for SS7 common channel signaling protocols with a number of digital line interfaces (T1/E1/J1) and an H.100 PCM highway that supports connection to a wide range of voice, data, and fax boards.

For a list of currently supported boards, see Section 5.3, "Dialogic® SS7 Boards", on page 37. For technical specifications, see http://www.dialogic.com/products/signalingip_ss7components/signaling_boards.htm.

The Dialogic® Global Call software supports the development of call control applications that use SS7 technology. For more information on using SS7 technology, see the *Dialogic® Global Call SS7 Technology Guide*.

3.4 3G-324M Interface

The 3G-324M technical specification is an umbrella protocol produced by the 3rd Generation Partnership Project (3GPP). An extension to the ITU-T H.324 Recommendation for 3G mobile phone conferencing, the 3G-324M specification includes H.245 for session control; H.223 for bit streams to data packets multiplexer/demultiplexer; H.223 Annex A and B for error handling of low and medium bit error rate (BER) detection, correction, and concealment; and H.324 with Annexes A and C for operating in a wireless environment. H.324 Annex K adds support for Media Oriented Negotiation Acceleration (MONA).

This release supports 3G-324M multimedia sessions over PSTN, as defined in 3GPP Release 99. It also supports 3G-324M multimedia sessions over IP using the Nb UP protocol, as defined in 3GPP Release 4.

For information about the 3G-324M API library, see Section 4.1, "Dialogic® 3G-324M API Library", on page 25.

Features

Features of the 3G-324M interface include:

Audio codecs for 3G-324M

Supported audio codecs are as follows:

- G.723.1 – 6.3 kbps, 30 ms frames, 1 frame per packet
- AMR Narrow Band – 12.2 kbps, 20 ms frames, 1 frame per packet

Note: Using the AMR-NB resource in connection with one or more Dialogic® products mentioned herein 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 http://www.voiceage.com/licensing.php.

Video codecs for 3G-324M

MPEG-4 is supported with these characteristics:

- Simple Profile (SP) – Level L0
- Video picture format – QCIF
- Frame rate – Up to 15 frames per second (fps)
- Bit rate – Up to 64 kbps

H.263 is supported with these characteristics:

- Profile and level – Profile 0, level 10
- Video picture formats – QCIF and sub-QCIF
- Frame rate – Up to 15 frames per second (fps)
- Bit rate – Up to 64 kbps

RTP video data is compliant with Internet Engineering Task Force RFC 2190, RTP Payload Format for H.263 Video Streams.

Video transcoding, video transrating, and image resizing

Video transcoding enables applications to record incoming video in a different format than what is being received from the network and to play back outgoing video in a different format than that of the locally stored file. Transcoding involves decoding and decompressing the original data to a raw intermediate format (YUV format).

Video transrating adjusts the number of video frames per second (and bitrate of the video) between two endpoints to suit the requirements of the device at each endpoint.

Image resizing converts video from one image size to another (for example, from CIF to QCIF) between two endpoints to suit the requirements of the device at each endpoint.

These features are available on MPEG-4 and H.263.

> ***Note:*** In this document, the term "video transcoding" encompasses video transcoding, video transrating, and image resizing.

3.5 RTP Interface

Audio Codecs for RTP

Supported audio codecs for RTP include:

- G.711 – 64 kbps format, mu-law and A-law, and 10, 20, and 30 ms frames
- G.723.1 – 5.3 and 6.3 kbps, 30 ms frames, and 1 or 2 frames per packet
- G.726 – 16, 24, 32 and 40 kbps
  - 20 ms frame size, 1, 2, or 3 frames per packet
- G.729A (compatible with G.729 format) – 8 kbps, 10 ms frames, and 1, 2, 3, or 4 frames per packet
- G.729AB (compatible with G.729B format) – 8 kbps, 10 ms frames, and 1, 2, 3, or 4 frames per packet
- AMR Narrow Band – 20 ms frames, 1 to 10 frames per packet

  AMR-NB data is compliant with RFC 3267, RTP Payload for AMR.

  > ***Note:*** Using the AMR-NB resource in connection with one or more Dialogic® products mentioned herein 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 http://www.voiceage.com/licensing.php.
- EVRC – 20 ms frames, 1, 2, or 3 frames per packet
- GSM EFR – 20 ms frames, 1 frame per packet
- QCELP – 8 and 13 kbps, 20 ms frames, and 1 frame per packet

  QCELP data is compliant with Service Option 60 (SO 60 - Header Removal) of 3GPP2 C.S0047-0, VoIP Link-Layer Assisted Service Options.

Video Codecs for RTP

Supported video codecs for RTP include:

MPEG-4 video codec

MPEG-4 is supported with these characteristics:

Simple Profile (SP), Levels L0, L1

- Video picture formats – Sub-QCIF, QCIF
- Frame rate – Up to 15 fps
- Bit Rate – Up to 64 kbps

Simple Profile (SP), Level L2

- Video picture formats – Sub-QCIF, QCIF
- Frame rate – Sub-QCIF and QCIF up to 30 fps; CIF up to 15 fps
- Bit Rate – Up to 128 kbps

Simple Profile (SP), Level L3

- Video picture formats – Sub-QCIF, QCIF, CIF
- Frame rate – Up to 30 fps
- Bit Rate – Up to 384 kbps

RTP video data is compliant with Internet Engineering Task Force RFC 3016, RTP Payload Format for MPEG-4 Audio/Visual Streams.

H.263 video codec

H.263 is supported with these characteristics:

Profile 0, Level 10

- Video picture formats – Sub-QCIF, QCIF
- Frame rate – Up to 15 fps
- Bit rate – Up to 64 kbps

Profile 0, Level 20

- Video picture formats – Sub-QCIF, QCIF, CIF
- Frame rate – Sub-QCIF and QCIF up to 30 fps; CIF up to 15 fps
- Bit rate – Up to 128 kbps

Profile 0, Level 30

- Video picture formats – Sub-QCIF, QCIF, CIF
- Frame rate – Up to 30 fps
- Bit rate – Up to 384 kbps

RTP video data is compliant with Internet Engineering Task Force RFC 2190, RTP Payload Format for H.263 Video Streams.

H.263-1998 (H.263+) video codec

H.263+ is supported with these characteristics:

Profile 0, Level 10

- Video picture format – QCIF
- Frame rate – Up to 15 fps
- Bit rate – Constant bit rate up to 40 kbps

A nominal bit rate and frame rate to use is 37.8 at 7.5 fps.

RTP video data is compliant with Internet Engineering Task Force RFC 2429 (RFC 4629). Only Baseline Profile is supported. No H.263 annexes are supported.

3.6 Channel Density Support

The following table lists the maximum channel density for supported audio codecs for RTP in Dialogic® Multimedia Kit Software Release 1.0 for PCIe. This maximum density varies with the Dialogic® MMA board in use and the license obtained.

Audio Codec for RTP	Maximum Density
G.711 (10 ms frames)	480
G.711 (20 and 30 ms frames)	1000
G.723.1	360
G.726	480
G.729A / G.729AB	400
AMR Narrow Band	300
EVRC	272
GSM EFR	300
QCELP	272

The following table lists the maximum channel density for supported video codecs for RTP in Dialogic® Multimedia Kit Software Release 1.0 for PCIe. This maximum density varies with the Dialogic® MMA board in use and the license obtained.

Video Codec for RTP	Picture Format	Frames per sec (fps)	Bit Rate (Kbps)	Maximum Density
H.263 / MPEG-4	Sub-QCIF	15	64	120
H.263 / MPEG-4	QCIF	15	64	120
H.263 / MPEG-4	CIF	15	64	30

The following table lists the maximum channel density for supported codecs for Nb UP (native connection only) in Dialogic® Multimedia Kit Software Release 1.0 for PCIe. This maximum density varies with the Dialogic® MMA board in use and the license obtained.

Codec for Nb UP (native only)	Maximum Density
3G-324M	240
G.711 (5 ms frames)	240
G.711 (20 ms frames)	240
AMR Narrow Band (12.2 Kbps)	240

3.7 Multimedia (Audio/Video) Play and Record

The following multimedia features are supported:

Multimedia programming libraries

Several programming libraries provide multimedia-related functionality:

- The Dialogic® Multimedia API library records and plays multimedia data using a multimedia device.
- The Dialogic® Device Management API library connects the multimedia device to other devices such as an IP media device or an m3g (3G-324M) device.
- Multimedia record and playback between the Dialogic® software and remote IP endpoints is accomplished by using the multimedia device and other devices such as IP media devices.
- The Dialogic® IP Media Library API provides IP multimedia session control.
- The Dialogic® Global Call API library provides IP call control for multimedia using SIP and Session Description Protocol (SDP). The Global Call API library must be used in third party call control (3PCC) mode.

Multimedia play and record

Support for the following functionality:

- Record from RTP stream to multimedia file. Play from multimedia file into RTP stream while maintaining synchronization.
- Multimedia API video record and playback with basic playback control and synchronized audio and video.
- Play to and record from SIP devices, depending on capability of device (audio or audio/video). Play video only if no audio is required. Play audio only for non-video devices.

Multimedia file formats

Support for the following file formats for play and record:

- Linear PCM (128 kbps), 16-bit, 8 kHz, mono, LSB-MSB ("little-endian") for audio play and record

Note: Voice API audio files may be used as the audio track in a multimedia session; however, no synchronization between the audio and video file is maintained. In this case, the ipm device in a multimedia session listens to the dx device to which the voice API is playing an audio file. This overrides any audio stream (but not video) from the mm device in the multimedia session. For details, see the *Dialogic® Multimedia API Programming Guide*.

- Dialogic® proprietary native audio file format used for native RTP play and record and for transcoding. For more information, see the *Dialogic® Multimedia API Programming Guide* and *Dialogic® Multimedia File Conversion Tools User Guide*.
- Dialogic® proprietary video file formats used for video transcoding. For more information, see the *Dialogic® Multimedia File Conversion Tools User Guide*.

Multimedia file conversion utility

The **hmp3gp** utility converts multimedia data from Dialogic® proprietary file format to 3rd Generation Partnership Project (3GPP) file format conforming to 3GPP specifications. The reverse direction is also supported.

This utility can be downloaded from the following web site. Check this web site periodically for any updates to the conversion tools and for any corresponding updates to the documentation:
http://www.dialogic.com/products/ip_enabled/download/multimedia/omf.htm

Note: The conversion utility performs CPU-intensive tasks and should only be used when sufficient CPU capacity is available and when it won't impact other operations on the system.

Quality of Service (QOS)

Support for existing QoS audio alarms through the Dialogic® IP Media Library API for the audio portion of a multimedia stream.

Note: QoS alarms and events are not supported for video streams.

Native RTP play and record

The RTP data in both incoming and outgoing directions is not processed or transcoded by Dialogic® software. With this feature, the RTP data is stored directly by and retrieved directly from Dialogic® software without application data handling.

The application must negotiate the proper coder formats when establishing the IP media sessions and match these formats when receiving and sourcing data. Additional media stream data such as RTP timestamps and sequence numbers are made available to the application by the native RTP feature. The RTP packets may be stored exactly as received, with packets out of order or even missing, as long as the RTP stream is retransmitted so that the receiving terminating endpoint can perform necessary packet loss recovery (PLR) activity.

Note: Multimedia (mm) devices do not support EVRC, GSM-EFR and QCELP as audio coders.

This feature is implemented in the Dialogic® Multimedia API Library. For more information, see Section 4.9, “Dialogic® Multimedia API Library”, on page 32.

Multimedia user I/O

Enables applications to directly play and record RTP data via user I/O buffers.

This feature is implemented in the Dialogic® Multimedia API Library. For more information, see Section 4.9, “Dialogic® Multimedia API Library”, on page 32.

Note: This feature is currently not supported. It will be supported in an upcoming Service Update.

Multimedia buffer I/O

Enables applications to directly play and record RTP data via memory.

This feature is implemented in the Dialogic® Multimedia API Library. For more information, see Section 4.9, “Dialogic® Multimedia API Library”, on page 32.

Note: This feature is currently not supported. It will be supported in an upcoming Service Update.

Multimedia runtime control

Allows multimedia play and record functions to be terminated on certain conditions such as digits received.

This feature is implemented in the Dialogic® Multimedia API Library. For more information, see Section 4.9, "Dialogic® Multimedia API Library", on page 32.

Runtime control support in native RTP play and record

The RTP data in both incoming and outgoing directions is not processed or transcoded by Dialogic® software. With this feature, the RTP data is stored directly by and retrieved directly from Dialogic® software without application data handling.

This feature is implemented in the Dialogic® Multimedia API Library.

WAVE file support

The play and record capabilities in the Dialogic® Multimedia API Library support WAVE file format.

Play a still image over a video stream

Only JPEG- and YUV-4:2:0 formatted source material are supported for playing a still image.

This feature is implemented in the Dialogic® Multimedia API Library.

Record or capture a still image from a video stream

This feature provides the ability to capture a frame after a video stream has been paused and save it as an image. This feature is implemented in the Dialogic® Multimedia API Library.

Digital Video Recorder (DVR) controls

These controls enable the user to pause, resume, and seek during video and audio playback. This feature is implemented in the Dialogic® Multimedia API Library.

3.8 Multimedia (Audio/Video) Transcoding

Multimedia transcoding includes both audio transcoding and video transcoding.

Multimedia transcoding features are described as follows:

Video transcoding, video transrating, and image resizing

Video transcoding enables applications to record incoming video in a different format than what is being received from the network and to play back outgoing video in a different format than that of the locally stored file. Transcoding involves decoding and decompressing the original data to a raw intermediate format (YUV format).

Video transrating adjusts the number of video frames per second (and bitrate of the video) between two endpoints to suit the requirements of the device at each endpoint.

Image resizing converts video from one image size to another (for example, from CIF to QCIF) between two endpoints to suit the requirements of the device at each endpoint.

These features are available on MPEG-4 and H.263.

Note: In this document, the term "video transcoding" encompasses video transcoding, video transrating, and image resizing.

Device support for video transcoding
Video transcoding is supported between these devices: 3G-324M (m3g), conferencing (cnf), IP media (ipm), and multimedia (mm) devices.

Device support for audio transcoding
Audio transcoding is supported between these devices: 3G-324M (m3g), conferencing (cnf), digital network interface (dti), IP media (ipm), multimedia (mm), and voice (dx) devices.

Note: Multimedia (mm) devices do not support EVRC, GSM-EFR and QCELP as audio coders.

Audio Codecs for Transcoding

Supported audio codecs for transcoding include:

- G.711
- G.723.1
- G.726
- G.729A
- G.729AB
- AMR Narrow Band
- EVRC
- GSM EFR
- QCELP

Video Codecs for Transcoding

Supported video codecs for transcoding include:

- H.263
- MPEG-4

3.9 Multimedia (Audio/Video) Conferencing

Multimedia conferencing allows a real-time audio/video session between two or more participants in two or more locations. The multimedia conference can occur over the IP network or the PSTN.

Participants in a multimedia conference can be audio only, video only, or audio and video. Multimedia transcoding must be applied for all participants of a conference; apply audio transcoding for audio participants and video transcoding for video participants. For more information on transcoding, see Section 3.8, “Multimedia (Audio/Video) Transcoding”, on page 20.

Users decide on the format of the output screen, and select who will be displayed on the output screen. The number of participants displayed depends on the video layout applied to the conference. Participants may be persistent or may be determined by the active talker algorithm based on user selection.

Multimedia conferencing is implemented in the Dialogic® Conferencing (CNF) API library and video layout is created using the Dialogic® Media Toolkit API library. For more information, see Section 4.2, “Dialogic® Conferencing (CNF) API Library”, on page 26 and Section 4.8, “Dialogic® Media Toolkit API Library”, on page 30.

3.10 Image Overlay

The image overlay feature allows you to place an image (bitmap, YUV or JPEG) over a streaming video. You create a template of the image and define the area on the video screen in which the image will be displayed.

Image overlay is implemented in the Dialogic® Media Toolkit API library. For more information, see Section 4.8, “Dialogic® Media Toolkit API Library”, on page 30.

3.11 Native RTP Hairpinning

Native RTP hairpinning enables applications to form RTP media stream connections between IP media streams, allowing the RTP media stream received from one IP media session to be retransmitted to the outgoing RTP media stream of another IP media session. This is done without processing or transcoding the RTP payload. The RTP packets that are hairpinned may be hairpinned as received, with packets out of order or even missing as long as the RTP stream is retransmitted so that the receiving terminating endpoint can perform necessary packet loss recovery (PLR) type functions.

Native RTP hairpinning is supported for both audio and video RTP streams.

In addition to IPM to IPM connections, native hairpinning connections may also be formed between 3G-324M and IPM devices.

Connections between devices are made using the Dialogic® Device Management API library.

Use cases for native RTP hairpinning include switching type applications, such as an IP-PBX, or streaming data to an external speech server. Benefits of using native RTP hairpinning include increasing achievable system densities, reducing latencies, and improving voice quality by eliminating an additional decode/encode operation.

3.12 IP Signaling

The MMK Software 1.0 for PCIe supports the industry-standard Session Initiation Protocol (SIP).

3.13 Tone Management

Tone management support includes:

- In-Band DTMF detection and generation
- RFC 2833 DTMF detection and generation

3.14 Remote Media Control Interface

The Media Server Markup Language (MSML) is an XML-based media resources control protocol.

The MSML media server software has been designed and implemented as an integral part of the MMK Software 1.0 for PCIe.

If the MMK Software 1.0 for PCIe is installed on a media server (MS), the MSML media server software enables a remote client, also known as an application server (AS), to control media resources.

The MSML media server software is based on the evolving MSML language, as defined in the MSML IETF draft-saleem-msml-06.

The connection between the AS and MS is established using the SIP protocol; thereafter media control commands/responses (in the form of MSML control syntax) are exchanged in SIP messages, such as the INFO message or the 200 OK response.

For more information, see the *Dialogic® MSML Media Server Software User's Guide*.

3.15 Audio (Voice) Play and Record

Supported voice play and record features include:

Voice API audio play and record capability
: The following capability is supported:

  - Playing and recording files in all supported encoding formats, with or without wave headers
  - Indexed play
  - Streaming to board (streams data to the network interface in real time)
  - Transaction record

Voice API audio play and record file formats
: The following file formats are supported:

  - G.711 mu-law and A-law (48 kbps and 64 kbps)
  - OKI ADPCM (24 kbps and 32 kbps)
  - G.726 (16 kbps and 32 kbps)

- Linear PCM (88 kbps and 128 kbps)

For information about multimedia play and record, see Section 3.7, “Multimedia (Audio/Video) Play and Record”, on page 18.

3.16 OA&M Support Using CLI and SNMP

MMK Software 1.0 for PCIe provides two tools for administration, management, and configuration: command line interface (CLI) and Simple Network Management Protocol (SNMP). They both have access to the same configuration and monitoring data. SNMP is MIB-based, and CLI is text command line-based.

For information about using CLI and SNMP, see the *Dialogic® System Configuration Guide*.

For information about a CLI demo that can be used for automating repetitive tasks, see Dialogic® Command Line Interface (CLI) Demo topic in Section 4.12, “Dialogic® Demonstration Programs”, on page 35.

Programming Libraries 4

This chapter describes the development libraries that are supported in Dialogic® Multimedia Kit Software Release 1.0 for PCIe (MMK Software 1.0 for PCIe).

- Dialogic® 3G-324M API Library 25
- Dialogic® Conferencing (CNF) API Library 26
- Dialogic® Device Management API Library 27
- Dialogic® Digital Network Interface API Library 28
- Dialogic® Fax API Library. 28
- Dialogic® Global Call API Library. 28
- Dialogic® IP Media API Library 29
- Dialogic® Media Toolkit API Library 30
- Dialogic® Multimedia API Library. 32
- Dialogic® Standard Runtime API Library 33
- Dialogic® Voice API Library 34
- Dialogic® Demonstration Programs 35

4.1 Dialogic® 3G-324M API Library

The Dialogic® 3G-324M API library provides a standards-compliant interface that enables conversational multimedia communication services to mobile handsets and terminals over circuit-switched networks and packet-switched networks.

The 3G-324M technical specification is an umbrella protocol produced by the 3rd Generation Partnership Project (3GPP). An extension to the ITU-T H.324 Recommendation for 3G mobile phone conferencing, the 3G-324M specification includes H.245 for session control; H.223 for bit streams to data packets multiplexer/demultiplexer; H.223 Annex A and B for error handling of low and medium bit error rate (BER) detection, correction, and concealment; and H.324 with Annexes A and C for operating in a wireless environment. H.324 Annex K adds support for Media Oriented Negotiation Acceleration (MONA).

The 3G-324M API library can be used in conjunction with other API libraries to develop multimedia services such as video conferencing, video-on-demand, surveillance, and multimedia entertainment services.

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

Features

The Dialogic® 3G-324M API library provides the following capabilities:

- Ability to control and manage 3G-324M multimedia sessions

  Note: It does not include a call session control protocol such as SS7 ISUP for establishing a bearer channel connection between 3G-324M endpoints.

- Ability to initiate/terminate a 3G-324M session (including H.245 and H.223)
- Ability to interconnect/disconnect H.223 multiplex inputs and outputs (through device management API library functions)
- Support for Media Oriented Negotiation Acceleration (MONA)
- Support for G.723.1 and AMR Narrow Band (AMR-NB) audio codecs
- Support for H.263 and MPEG-4 video codecs

  Note: Using the AMR-NB resource in connection with one or more Dialogic® products mentioned herein 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 http://www.voiceage.com/licensing.php.

4.2 Dialogic® Conferencing (CNF) API Library

The Dialogic® Conferencing (CNF) API library supports development of multimedia (audio/video) conferencing applications. The conference can take place over an IP network and/or over traditional PSTN lines.

For more information, see the *Dialogic® Conferencing API Programming Guide* and the *Dialogic® Conferencing API Library Reference*.

Note: The maximum number of parties allowed in a single conference is 254.

Features

The Dialogic® Conferencing (CNF) API library provides the following features:

Multimedia (audio/video) conferencing

: Enables images from conference participants to be combined into a single image viewed by the conference participants.

  The design of the video layout is created through the layout builder functions of the Dialogic® Media Toolkit API (see Section 4.8, “Dialogic® Media Toolkit API Library”, on page 30).

Asynchronous programming model

: Enables multiple channels to be handled in a single process and supports higher density conferencing solutions.

Conference bridging

: Multiple conferences can be bridged together so that all parties (also called conferees) in two or more established conferences can communicate with one another.

Coach/pupil feature
: Two selected parties can establish a private communication link within the overall conference. The coach is a private member of the conference and is only heard by the pupil. However, the pupil cannot speak privately with the coach.

DTMF digit detection
: The application can determine whether a party has generated a DTMF digit.

Volume control
: A conferee can adjust the output volume, either by API command or by DTMFs detected on a conferee's input leg.

DTMF tone clamping
: This feature mutes dual tone multi-frequency (DTMF) tones heard during a conference. Tone clamping applies to the transmitted audio going into the conference and does not affect DTMF function. It can be enabled on a board, conference, or party basis.

Automatic gain control (AGC)
: AGC is an algorithm for normalizing an input signal to a target level. The AGC algorithm discriminates between voiced and unvoiced signals within a conference.

Active talker
: The active talker feature sums the three most active talkers in a conference, so that the conversation doesn't get drowned out when too many people talk at once.

Conference monitoring
: Participants have listen-only access to a conference.

Echo cancellation
: This feature reduces echo from the incoming signal, improving the quality of a conference for all participants.

Tariff tone
: A party can receive a periodic tone for the duration of the conference call.

4.3 Dialogic® Device Management API Library

The Dialogic® Device Management API library provides run-time control and management of configurable system devices, including functions to reserve resources and to manage connections between devices for communication.

For more information, see the *Dialogic® Device Management API Library Reference*.

Features

The Dialogic® Device Management API library provides the following capabilities:

Device connection capabilities
: Enables a connection between devices, including one or more transmit ports and one or more receive ports.

Resource reservation capabilities

Manages resources and reserves a resource for use by a specific device, such as when reserving low bit-rate coders.

Native features

Native features include RTP hairpinning and RTP play and record. The device management API library provides the ability to connect devices together natively; the DMFL_TRANSCODE_NATIVE flag in DM_PORT_CONNECT_INFO data structure must be turned on. For more information on native features, see Section 4.7, "Dialogic® IP Media API Library", on page 29 and Section 4.9, "Dialogic® Multimedia API Library", on page 32.

Video transcoding, video transrating, and image resizing

The device management API library provides the ability to connect devices together, as a native connection or for transcoding. Transcoding is invoked when connections are formed. Use the DMFL_TRANSCODE_ON flag in DM_PORT_CONNECT_INFO structure to enable transcoding. For more information on video transcoding, see Section 3.8, "Multimedia (Audio/Video) Transcoding", on page 20.

4.4 Dialogic® Digital Network Interface API Library

The Dialogic® Digital Network Interface API library supports development of applications that require connection to a T1 or E1 network interface. This API library is used by the Dialogic® HMP Interface Boards (DNI boards).

For more information, see the "Digital Network Interface API for DM3" chapter in the *Dialogic® Digital Network Interface API Library Reference*.

4.5 Dialogic® Fax API Library

The Dialogic® Fax API library supports development of a wide variety of fax applications such as fax mail, fax broadcast and fax-on-demand. The fax software includes library functions, device drivers, and firmware files.

See the *Dialogic® Fax Software Reference* for more information.

4.6 Dialogic® Global Call API Library

The Dialogic® Global Call API library provides a uniform call control interface for developing applications for multiple network interface technologies. The Global Call API library supports a variety of protocols.

The Global Call API library provides the following capabilities on MMK Software 1.0 for PCIe:

- Supports SIP and PSTN protocols

- Provides a consistent application interface for the various signaling protocols and technologies
- Runs in third party call control (3PCC) mode only (SIP)

Call control is managed by the Global Call API library, and media exchange is managed by the IP Media Library. The Global Call API library supports multimedia call control over IP when using SIP and SDP.

The generic functionality of the Global Call API is documented in the *Dialogic® Global Call API Library Reference* and the *Dialogic® Global Call API Programming Guide*. Functionality specific to a technology is documented in technology guides: *Dialogic® Global Call IP Technology Guide*, *Dialogic® Global Call ISDN Technology Guide*, and *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*.

Features

Features of the Dialogic® Global Call API include:

Third party call control (3PCC)
: In 3PCC mode, the application can use the Global Call SIP stack without RTP. This allows the application to control calls between two or more other parties without being in the RTP/Media path. This mode also allows an application to have direct control over RTP establishment. The application is responsible for generating and parsing the SDP used for RTP establishment. 3PCC mode is only relevant to SIP call control.

SIP re-INVITE
: In 3PCC mode, ability to generate and accept SIP re-INVITE methods to modify the characteristics of established media sessions.

> ***Note:*** The **ipm_ModifyMedia( )** function, which modifies various properties of an active media session, is not supported in this release as part of the re-INVITE functionality supported in Global Call 3PCC mode. Support for changing the media session parameters, including changing the media codec in a media session to switch the call from audio to video, is provided by interrupting the session using the **ipm_Stop( )** function, making the changes, and then resuming the session with the **ipm_StartMedia( )** function.

4.7 Dialogic® IP Media API Library

The Dialogic® IP Media Library API (IPML) is used to control media on IP devices. Voice over IP applications that use IP signaling stacks other than those supplied with Dialogic® products may use this library for application development.

Note: IP call control with multimedia is provided only when using the Dialogic® Global Call library in 3PCC mode.

For more information, see the *Dialogic® IP Media Library API Library Reference* and the *Dialogic® IP Media Library API Programming Guide*.

Features

The Dialogic® IP Media Library API supports the following features:

Multimedia session control
: Ability to start/stop multimedia session and get/set video related properties.

DTMF mode
: Ability to configure the preferred DTMF mode: RFC 2833, in-band, or out-of-band. Ability to generate and receive DTMF tones to and from the TDM bus.

Quality of Service (QoS)
: Ability to configure the Quality of Service (QoS) alarm threshold and status reporting for audio streams (including the audio streams in multimedia sessions).

Note: QoS alarms and events are not supported for video streams.

Video transcoding, video transrating, and image resizing
: The IPM_VIDEO_CODER_INFO_EX and IPM_VIDEO_CODER_INFO structures specify video codec information for H.263 and MPEG-4.

The DMFL_TRANSCODE_ON flag in the DM_PORT_CONNECT_INFO structure of the device management API library is used to enable transcoding.

For more information on video transcoding, see Section 3.8, "Multimedia (Audio/Video) Transcoding", on page 20.

Native RTP hairpinning
: The RTP media stream received from one IP media session can be retransmitted to the outgoing RTP media stream of another IP media session.

Native RTP play and record
: Audio coder types to support native RTP play and record are in the IPM_AUDIO_CODER_INFO structure. These audio coder types include _NATIVE as part of the value name.

Nb UP protocol support
: An internal media device may be connected to an external IP network device using the 3GPP Nb User Plane (Nb UP) protocol. The Nb UP protocol is a packet-based interface, which is available as an alternative transport for bearer traffic between media gateways in the core network. In this case the Nb UP is intended to transport one or more 3G-324M or audio payloads. Each 3G-324M payload is the same as that used with PSTN TDM links (such as E1) except that the underlying transport technology is based on the Internet Protocol stack.

I-Frame update
: The application can send a request for the ipm device to transmit an I-Frame update (video fast update) to the remote terminal as needed using **ipm_GenerateIFrame( )**.

4.8 Dialogic® Media Toolkit API Library

The Dialogic® Media Toolkit API (MTK) library consists of general-purpose structures and attribute templates as well as API functions for building and manipulating media-related items, such as video layouts and bitmaps. The library also includes functions that allow for the integration of these templates and media-related items with existing Dialogic®

libraries. Generally, templates represent properties of a media-related item, such as a media file or a video layout, and are referenced when setting attributes of a media stream or a video conference.

The Dialogic® Media Toolkit API library includes a main library (mtk) and several sub-libraries as follows: layout builder (lb), overlay builder (ob), and stream manipulation (sm). Each library encapsulates a given type of functionality.

The layout builder functions allow the user to specify the video layout of a video conference or multimedia conference. These functions can be used in conjunction with the Dialogic® Conferencing (CNF) API to develop multimedia conferencing applications.

For more information about this API library, see the *Dialogic® Media Toolkit API Library Reference*.

Features

The Dialogic® Media Toolkit API library supports the following functionality:

Media toolkit (mtk) functions
: Used to create templates for images (bitmap, YUV or JPEG) and frames.

  Note: Only YUV 4:2:0 format is currently supported.

  Note: Image file size should not exceed 148 kbytes.

Layout builder (lb) functions
: Used to specify the video layout of a video conference or multimedia conference.

  - Several layout types are supported including one region (full screen), four regions, six regions, and nine regions. Custom layout types are also supported.
  - Display modes for a participant or party include still image and live streaming.
  - Selection modes supported include active talker and user-selected.

Overlay builder (ob) functions
: Used to define an overlay template to be applied to a streaming device. Attributes of an overlay template include size and position of the bounding frame, overlay fill style, and duration for the overlay to be played.

Stream manipulation (sm) functions
: Used to manage overlays on a streaming device; that is, add overlays to a device and remove overlays from a device. An overlay can be applied to ipm, mm or m3g device types.

For more information about these features, see the *Dialogic® Media Toolkit API Library Reference*.

4.9 Dialogic® Multimedia API Library

The Dialogic® Multimedia API library can be used to play and record digitized multimedia in support of applications providing video services, such as video mail, video color ring, video caller ID, and video location-based services.

For more information, see the *Dialogic® Multimedia API Library Reference* and the *Dialogic® Multimedia API Programming Guide*.

Features

The Dialogic® Multimedia API library provides the following capabilities:

Notification tone
: Transmit a start-of-recording tone to notify the party being recorded. If enabled, the tone is transmitted upon detection of an I-frame (complete video frame) or upon time-out waiting for an I-frame.

Record multimedia (audio/video) data
: Record audio and video data from an IP stream into a file in real time; also provides the capability to record only the audio portion or video portion.

Play multimedia (audio/video) data
: Play back audio and video data from a file to a media session in real time while maintaining synchronization; also provides the capability to playback only the audio portion or video portion.

Native RTP play and record
: The RTP media stream in both incoming and outgoing directions is not processed or transcoded by Dialogic® software. With this feature, the RTP data is stored directly by and retrieved directly from software without application data handling.

Proprietary audio and video file formats
: Proprietary file formats are used for storage and playback. These files can be converted to 3GPP file format. For more information on the proprietary file formats and conversion tools, see the *Dialogic® Multimedia File Conversion Tools User Guide*.

Play audio files
: Play Dialogic® Voice API audio files in a multimedia session where tight synchronization with video is not required (such as for playing with a video menu or status display).

Video transcoding, video transrating, and image resizing
: Characteristics of the video coder are stored in the MM_VIDEO_CODEC structure. Transcoding is enabled in the Dialogic® Device Management API.

: For more information on video transcoding, see Section 3.8, “Multimedia (Audio/Video) Transcoding”, on page 20.

Audio transcoding
: Characteristics of the audio coder are stored in the MM_AUDIO_CODEC and MM_AUDIO_CODEC_OPTIONS_INFO structures.

WAVE file support
The **mm_Play( )** and **mm_Record( )** functions support the WAVE file format.

Play a still image over a video stream
The **mm_Play( )** function supports playing a still image. The MM_MEDIA_IMAGE structure and several structures support this feature.

Record or capture a still image from a video stream
The **mm_Capture( )** function provides the ability to capture a still image. The MM_MEDIA_IMAGE structure and several structures support this feature.

Digital Video Recorder (DVR) controls
The **mm_Seek( )**, **mm_Pause( )**, **mm_Resume( )**, **mm_GetDuration( )**, and **mm_GetElapsedTime( )** functions support DVR controls.

Multimedia user I/O
Enables applications to directly play and record RTP data via user I/O buffers.

Note: This feature is currently not supported. It will be supported in an upcoming Service Update.

Multimedia buffer I/O
Enables applications to directly play and record RTP data via memory.

Note: This feature is currently not supported. It will be supported in an upcoming Service Update.

Multimedia runtime control
Allows multimedia play and record functions to be terminated on certain conditions such as digits received. These conditions are specified in the MM_RUNTIME_CONTROL data structure.

Note: Multimedia (mm) devices do not support EVRC, GSM-EFR or QCELP as audio codecs for audio transcoding or for native play and record.

For more information on the updates, see the *Dialogic® Multimedia API Library Reference*.

4.10 Dialogic® Standard Runtime API Library

The Dialogic® Standard Runtime Library (SRL) API provides a common interface for event handling and other functionality common to Dialogic® devices. The Standard Runtime Library provides the framework for implementing the supported programming models and serves as the central dispatcher for events that occur on all devices. Through the Standard Runtime Library, events are handled in a standard manner.

For more information, see the *Dialogic® Standard Runtime Library API Programming Guide* and the *Dialogic® Standard Runtime Library API Library Reference*.

Features

The Dialogic® SRL API library supports the following features:

Choice of programming models
: Includes support for the following:
- Asynchronous polled model
- Asynchronous with non-signal callback model
- Extended asynchronous model

Event handling
: Provides the ability to handle events on a per device basis, such as enable and disable event handlers.

Event data retrieval
: Provides the ability to retrieve information about the current event.

Device information retrieval
: Standard attribute functions return general information about a device, such as device name, board type, and last error that occurred. Standard attribute functions have ATDV_ as the prefix.

User-specific context
: Enables the application to set and retrieve user-specific context using the **sr_setparm( )** and **sr_getparm( )** functions.

Note: The **sr_getboardcnt( )** function is not supported. Use the SRL Device Mapper functions to return information about the structure of the system such as a list of boards and devices.

4.11 Dialogic® Voice API Library

The Dialogic® Voice API library provides a rich set of features for building a wide range of high-density call processing applications such as voice messaging, interactive voice response, telemarketing/call center, operator services, and more. Features include tone signaling, global tone detection and generation, call progress analysis, and a variety of voice encoding algorithms selectable on a channel-by-channel basis.

For more information, see the *Dialogic® Voice API Library Reference* and the *Dialogic® Voice API Programming Guide*.

Features

The Dialogic® Voice API library provides the following features:

Call progress and call analysis
: Implemented through the **dx_dial( )** function. Handles pre-connect or call progress that reports the status of the call connection, such as busy, no dial tone, or no ringback. Handles post-connect or call analysis that reports the destination party's media type, such as voice, fax, or answering machine.

Tone detection and tone generation

Includes support for the following:

- Dual Tone Multi Frequency (DTMF)
- Global Tone Detection (GTD) user-defined tones
- Global Tone Generation (GTG) user-defined tones, including Cadenced Tone Generation

Data formats for play and record

Includes support for the following:

- G.711 PCM at 6 kHz with 8-bit samples (48 kbps) and 8 kHz with 8-bit samples (64 kbps) using A-law or mu-law coding, VOX and WAVE file formats
- OKI ADPCM at 6 kHz with 4-bit samples (24 kbps) and 8 kHz with 4-bit samples (32 kbps), VOX and WAVE file formats
- PCM at 11 kHz with 8-bit samples (88 kbps) using linear coding, VOX and WAVE file formats
- PCM at 8 kHz with 16-bit samples (128 kbps) using linear coding, VOX file format (Multimedia API audio file format)
- G.726 bit-exact voice coder at 8 kHz with 2- or 4-bit samples (16, 32 kbps), VOX and WAVE file formats

Volume control

Users can adjust the volume of a playback.

Speed control

Users can adjust the speed of a playback via DTMF or via other conditions set using **dx_adjsv( )**. The following coders are supported for speed control:

- 24 kbps and 32 kbps OKI ADPCM (6 kHz 4-bit and 8 kHz 4-bit)
- 48 kbps and 64 kbps G.711 A-law PCM (6 kHz 8-bit and 8 kHz 8-bit)
- 48 kbps and 64 kbps G.711 mu-law PCM (6 kHz 8-bit and 8 kHz 8-bit)
- 128 kbps linear PCM (8 kHz 16-bit)

Note: Before using the speed control feature, you must enable this feature in the [decoder] section of the CONFIG file. For more information, see the *Dialogic® System Configuration Guide*.

Transaction record

Implemented through the **dx_mreciottdata( )** function. Enables the recording of a two-party conversation by allowing two time slots from a single channel to be recorded.

Asynchronous routing

Implemented through the **dx_listenEx( )** and **dx_unlistenEx( )** functions.

4.12 Dialogic® Demonstration Programs

Demonstration programs are provided to demonstrate product functionality and features, and serve as examples of application programming using the supported API libraries. All demo programs are supplied as source code that users may choose to modify to explore other capabilities of the products.

The demo programs are located in */usr/dialogic/demos*.

Note: Demos may require configuration or modification before running.

Dialogic® Multimedia Demo Program

The Dialogic® Multimedia Demo processes a multimedia (audio and video) IP call with SIP endpoints. It uses the following APIs to accomplish the audio/video streaming:

- Dialogic® Device Management API to connect the multimedia device with an IP media device
- Dialogic® Multimedia API to record and play audio/video data
- Dialogic® IP Media Library API for media session control and RTP stream manipulation
- Dialogic® Global Call API in 3PCC mode for SIP call control

Dialogic® 3G-324M Multimedia Gateway Demo Program

The 3G-324M multimedia gateway demo is composed of two parts:

- The m3g-sip_gateway demo application, which demonstrates 3G mobile connectivity to a remote 3G-324M endpoint using the 3G-324M API library. The demo starts a 3G call session and bridges 3G calls to a SIP endpoint.
- The multimedia demo application, when configured to work with the m3g-sip_gateway demo application, provides multimedia streaming to a remote 3G mobile endpoint.

For more information on the 3G-324M multimedia gateway demo, see the *Dialogic® 3G-324M Multimedia Gateway Demo Guide*.

Dialogic® Command Line Interface (CLI) Demo

The CLI demo is a script that can be used to automate repetitive tasks, such as issuing commands to the CLI. The CLI demo can only be used to automate CLI commands. The following files are provided:

- *cte.pl* file, the PERL script which handles CLI session, command parsing/processing, and logging
- **.cfg* files, which represent the basic configuration commands and should be modified to suit your operating environment
- *cte_readme.txt*, which provides instructions for using the *cte.pl*

Note: Before using the CLI demo, the Net-Telnet package and the PERL binary itself must be installed. On most Linux distributions, the PERL binary is installed with a regular OS install, but it is not part of the basic package install. On Windows®, the PERL binary is not part of the OS install, and must be installed separately.

A sequence of CLI commands can be placed into a text file residing on a remote platform, which can then run those commands on the host platform by executing the perlscript locally and using telnet to transmit them to the host platform.

Supported Hardware 5

This chapter lists the boards supported in MMK Software 1.0 for PCIe.

- Dialogic® Multimedia Accelerator Boards for PCIe 37
- Dialogic® HMP Interface Boards 37
- Dialogic® SS7 Boards 37

5.1 Dialogic® Multimedia Accelerator Boards for PCIe

The following Dialogic® Multimedia Accelerator Boards for PCIe are supported in this release:

- Dialogic® MMAC250PCIEQ Multimedia Accelerator Board for PCIe
- Dialogic® MMAC500PCIEQ Multimedia Accelerator Board for PCIe

For more information, see http://www.dialogic.com/products/ip_enabled/high_density_ip_media.htm.

5.2 Dialogic® HMP Interface Boards

The following Dialogic® HMP Interface Boards are supported in this release:

- Dialogic® DNI310TEPEHMPQ Digital Network Interface Boards
- Dialogic® DNI610TEPEHMPQ Digital Network Interface Boards
- Dialogic® DNI1210TEPEHMPQ Digital Network Interface Boards
- Dialogic® DNI2410TEPEHMPQ Digital Network Interface Boards

For technical specifications, see http://www.dialogic.com/products/ip_enabled/hmp_enabled_boards.htm.

5.3 Dialogic® SS7 Boards

The following Dialogic® SS7 PCI boards are supported in this release:

- Dialogic® SPCI2S SS7 Boards
- Dialogic® SPCI4 SS7 Boards
- Dialogic® SS7HDP SS7 Boards

For technical specifications, see http://www.dialogic.com/products/signalingip_ss7components/signaling_boards.htm.

Documentation 6

This chapter provides information about the documentation that supports the Dialogic® Multimedia Kit Software Release 1.0 for PCIe (MMK Software 1.0 for PCIe). This information is organized into the following sections:

- Documentation Feature Support 38
- Release Documentation 40
- Installation and Configuration Documentation 40
- OA&M Documentation 41
- Programming Libraries Documentation. 41
- Application Scenario Documentation 41
- Demonstration Software Documentation 42

6.1 Documentation Feature Support

The following table lists topics and features associated with the MMK Software 1.0 for PCIe features and the user documentation that contains information about these features.

Table 1. Documentation Feature Support

Topic or Feature	MMK Software 1.0 for PCIe Documentation
Release overview, features, API libraries, and system requirements	• Release Guide
Known issues, limitations, new developments, documentation corrections	• Release Update
Installing the software, including pre-installation and post-installation requirements	• Software Installation Guide
License activation	• Software Installation Guide
Resource licenses	• Contact your Dialogic sales representative or authorized Dialogic distributor
Configuring software and boards using the Command Line Interface (CLI) tool or SNMP tool; includes system performance tuning	• System Configuration Guide

Table 1. Documentation Feature Support (Continued)

Topic or Feature	MMK Software 1.0 for PCIe Documentation
3G-324M interface	• 3G-324M API Programming Guide and Library Reference • 3G-324M Multimedia Gateway Demo Guide
Conferencing (CNF)	• Conferencing API Library Reference • Media Toolkit API Library Reference
Diagnostics	• Diagnostics Guide
Digital network interface	• Digital Network Interface Software Reference • Global Call ISDN Technology Guide • Global Call E1/T1 CAS/R2 Technology Guide
Event handling	• Standard Runtime Library API Library Reference • Standard Runtime Library API Programming Guide
Fax using Global Call API	• Global Call IP Technology Guide • Fax Software Reference
Fax using Third-Party Stack	• IP Media Library API Library Reference • IP Media Library API Programming Guide • Device Management API Library Reference • Fax Software Reference
Image overlay	• Multimedia API Programming Guide and Library Reference • Media Toolkit API Library Reference
IP call transfer	• Global Call API Library Reference • Global Call IP Technology Guide
IP multicast	• IP Media Library API Library Reference • IP Media Library API Programming Guide
Multimedia capture and play still image	• Multimedia API Programming Guide and Library Reference
Multimedia Digital Video Recorder (DVR) controls	• Multimedia API Programming Guide and Library Reference
Multimedia (audio/video) play and record	• Multimedia API Programming Guide and Library Reference • Multimedia Demo Guide • Device Management API Library Reference • Multimedia File Conversion Tools User Guide (available with the multimedia file conversion tools download)

Table 1. Documentation Feature Support (Continued)

Topic or Feature	MMK Software 1.0 for PCIe Documentation
Multimedia (audio/video) transcoding	• Multimedia API Programming Guide and Library Reference • Device Management API Library Reference • IP Media Library API Library Reference
Remote media control interface	• MSML Media Server Software User's Guide
SIP call control using Global Call API	• Global Call API Library Reference • Global Call API Programming Guide • Global Call IP Technology Guide
SIP call control using a third-party stack	• IP Media Library API Library Reference • IP Media Library API Programming Guide • Device Management API Library Reference
SS7 technology with Global Call API	• Global Call SS7 Technology Guide • Global Call API Library Reference • Global Call API Programming Guide
Voice (audio) features such as play and record, file formats, transaction record	• Voice API Library Reference • Voice API Programming Guide

6.2 Release Documentation

The following release documentation is provided for this release:

- *Dialogic®* Multimedia Kit Software Release 1.0 for PCIe *Release Guide* (this document)
- *Dialogic® Multimedia Kit Software Release 1.0 for PCIe Release Update*

  Note: The Release Update includes issues that may affect the performance of the Dialogic® software and lists both resolved and known issues. The Release Update also includes corrections and changes to the user documentation that were not made to the documents prior to the release.

6.3 Installation and Configuration Documentation

The following installation and configuration documentation is provided for this release:

- *Dialogic® Multimedia Kit Software Release 1.0 for PCIe Software Installation Guide*
- *Dialogic® System Configuration Guide*
- *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*

6.4 OA&M Documentation

The following OA&M software documentation is provided for this release:

- *Dialogic® Host Media Processing Diagnostics Guide*

6.5 Programming Libraries Documentation

The following programming libraries documentation is provided for this release:

- *Dialogic® 3G-324M API Programming Guide and Library Reference*
- *Dialogic® Conferencing API Library Reference*
- *Dialogic® Conferencing API Programming Guide*
- *Dialogic® Device Management API Library Reference*
- *Dialogic® Digital Network Interface Software Reference*
- *Dialogic® Fax Software Reference*
- *Dialogic® Global Call API Library Reference*
- *Dialogic® Global Call API Programming Guide*
- *Dialogic® Global Call IP Technology Guide*
- *Dialogic® Global Call ISDN Technology Guide*
- *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*
- *Dialogic® Global Call SS7 Technology Guide*
- *Dialogic® IP Media Library API Library Reference*
- *Dialogic® IP Media Library API Programming Guide*
- *Dialogic® Media Toolkit API Library Reference*
- *Dialogic® Multimedia API Programming Guide and Library Reference*
- *Dialogic® Standard Runtime Library API Library Reference*
- *Dialogic® Standard Runtime Library API Programming Guide*
- *Dialogic® Voice API Library Reference*
- *Dialogic® Voice API Programming Guide*

6.6 Application Scenario Documentation

The following application scenario documentation is provided for this release:

- *Dialogic® MSML Media Server Software User's Guide*

6.7 Demonstration Software Documentation

The following demo documentation is provided for this release:

- *Dialogic® Global Call API Demo Guide*
- *Dialogic® Multimedia Demo Guide*
- *Dialogic® 3G-324M Multimedia Gateway Demo Guide*