Dialogic® Host Media Processing Software Release 3.0WIN

Release Guide

December 2010

05-2507-007

Publication Date: December 2010

Document Number: 05-2507-007

Contents

Document Revision History 5

About This Publication 9

Applicability 9
Intended Audience 9
How to Use This Publication 9
Related Information 10

1 Release Overview 11

2 System Requirements 15

2.1 Basic Hardware Requirements 15
2.2 Basic Software Requirements 16
2.3 Equipment and Configurations Tested 17
2.3.1 Multimedia Gateways 18
2.3.2 Multimedia and Voice IP Devices 18
2.3.3 Configurations Tested 19
2.4 Ordering the Product 20

3 Features 23

3.1 Multimedia (Audio/Video) 23
3.2 Secure Real-time Transport Protocol (SRTP) 25
3.3 SIP Transport Layer Security (TLS) 26
3.4 MSML Media Server Software for Remote Applications 26
3.5 Higher Channel Density Support 27
3.6 Dialogic® Conferencing (CNF) API 27
3.7 Support for SS7 Products and SIUs 27
3.8 Analog Device (Springware) Support 29
3.10 Other Features 38

4 Installation and Configuration 43

4.1 Installation 43
4.2 Configuration 44

5 OA&M Software 45

5.1 Administration Software 45
5.2 Dialogic® NCM (Native Configuration Manager) API Library 45
5.3 Diagnostics Software 46

6 Programming Libraries 47

6.1 Dialogic® Audio Conferencing (DCB) API Library 47
6.2 Dialogic® Conferencing API Library 48
6.3 Dialogic® Continuous Speech Processing (CSP) API Library 49
6.4 Dialogic® Device Management API Library 50
6.5 Dialogic® Fax API Library 50
6.6 Dialogic® Global Call API Library 50

6.7 Dialogic® IP Media Library API Library 51
6.8 Dialogic® Learn Mode and Tone Set File APIs 52
6.9 Dialogic® Multimedia API Library 53
6.10 Dialogic® Standard Runtime API Library 53
6.11 Dialogic® Voice API Library 54

7 Supported Applications 55

7.1 MSML Media Server Software 55

8 Supported Hardware 57

8.1 Dialogic® Digital Network Interface Products 57
8.2 Dialogic® Signaling Products 57
8.3 Dialogic® Analog Interface Products 58

9 Demonstration Software 59

9.1 New Demo Programs 59
9.2 Other Supported Demo Programs 60

10 Documentation 65

10.1 Documentation Support for Dialogic HMP Software Release 3.0WIN Features 65
10.2 Release Documentation 68
10.3 Installation and Configuration Documentation 69
10.4 OA&M Documentation 69
10.5 Programming Libraries Documentation 69
10.6 Supported Applications Documentation 70
10.7 Demonstration Software Documentation 70
10.8 Online Help 71

Document Revision History

This revision history summarizes the changes made in each published version of the Release Guide for Dialogic® Host Media Processing Software Release 3.0WIN, which is a document that is subject to updates during the lifetime of the release.

Document Rev 07 - published December 2010

Updated to support the Dialogic® D/4PCIUFEQ and the Dialogic® D/4PCIU4SEQ boards.

In the Release Overview chapter:

- Renamed the "Analog Device Support" section to Analog Device (Springware) Support and added the Dialogic® D/4PCIUFEQ and the Dialogic® D/4PCIU4SEQ boards.
- Added a bullet item about a restriction when assigning Dialogic® HMP Software licenses to a NIC in systems with more than five NICs to the Software Restrictions subheading. (IPY00091711)

In the Features chapter:

- In Support for SS7 Products and SIUs, Features subheading, added a note to the Global Call API Support item that Global Call SS7 binaries are linked with the dynamic linked libraries in the Dialogic® SS7 DSI Development Package. (IPY00081381)
- Renamed the "Analog Device Support" section to Analog Device (Springware) Support and updated the entire section with information about the Dialogic® D/4PCIUFEQ and the Dialogic® D/4PCIU4SEQ boards. Removed restrictions for CNF conferencing and Device Management APIs dev_connect( ) and dev_disconnect( ). Also removed the "Documentation Considerations" section because the analog-specific documents have been incorporated into the Software Release 3.0WIN bookshelf. Previously, they appeared under an Analog Device subheading.

In the Supported Hardware chapter:

- Added the Dialogic® D/4PCIUFEQ and the Dialogic® D/4PCIU4SEQ boards to the Dialogic® Analog Interface Products section.

In the Programming Libraries chapter:

- Added a section for the Dialogic® Learn Mode and Tone Set File APIs.

In the Documentation chapter:

- Added a row for Analog Device (Springware) Support to the "User Documentation Feature Support" table.

- Added the *Dialogic® Springware Architecture Products on Windows Configuration Guide* to the Installation and Configuration Documentation section.
- Added the *Dialogic® Global Call Analog Technology Guide* to the Programming Libraries Documentation section.
- Added the *Dialogic® Learn Mode and Tone Set File API Software Reference* to the Programming Libraries Documentation section.

Document Rev 06 - published September 2010

Removed references to the Dialogic® Digital Station Interface (DSI) boards and the Dialogic® Station Side Interface API Library Reference because these products have been discontinued and are no longer supported.

In the Release Overview chapter:

- Added a Virtualization Support subsection.

In the Features chapter:

- Updated the maximum number of channels in Secure Real-time Transport Protocol (SRTP).
- Updated the maximum number of channels in SIP Transport Layer Security (TLS).
- Updated the Important Considerations topic in the Analog Device (Springware) Support section with information about echo cancellation. Also updated the Voice Encoding Methods (Springware) table.

Document Rev 05 - published July 2010

Updated to support the Dialogic® D/80PCIE-LS Media Board.

In the Release Overview chapter:

- Added an Analog Device Support subsection.

In the Features chapter:

- Added Analog Device (Springware) Support.

In the Supported Hardware chapter:

- Added a Dialogic® Analog Interface Products section.

Document Rev 04 - published March 2010

Updated to support the 32-bit versions of Windows® 7 and Windows Server® 2008 operating systems.

In the Release Overview chapter:

- Updated channel density under the Media Processing Functions heading.

- In the Software Restrictions topic, added a bullet item about Support for Quality of Service (QoS) under Global Call on Windows® 7 or Windows Server® 2008 operating systems.

In the System Requirements chapter:

- Basic Hardware Requirements section:
  - Added a note that a minimum of 2 GB of RAM is required for Windows® 7 or Windows Server® 2008 under memory.
  - Added information about the HPET Timer.
  - Updated the information in the Physical Address Extension (PAE) bullet item.
- Basic Software Requirements section:
  - Added Windows® 7 and Windows Server® 2008 under the Operating System bullet.
  - Added Windows® Web Edition SP2 under the Operating System bullet.
  - Added Windows® XP Professional with Service Pack 3 under the Operating System bullet.
  - Added Development Environment Microsoft® Visual Studio 2005 with Visual Studio 2005 Service Pack 1 and Visual Studio 2005 Service Pack1 Update for Windows® Vista to the bulleted list under compilers.
  - Added two notes under the Compilers bullet regarding compiler versions and User Account Control (UAC) restrictions, respectively.
  - Added Cisco Call Manager 6.0 to row 7 of Table 2. “Devices Tested for Interoperability with Dialogic® HMP Software.”
  - Added a row for up to 1000 user sessions to Table 1, “Processor Recommendations”, on page 16.

In the Features chapter:

- Updated the link to the Multimedia File Conversion Utilities in Section 3.1, “Multimedia (Audio/Video)”.
- Updated channel density in the Higher Channel Density Support section.
- Added a note about processor throttling under the bullet item about the APIC Timer.
- Added support for Development Environment Microsoft® Visual Studio 2005 with Visual Studio 2005 Service Pack 1 and Visual Studio 2005 Service Pack1 Update for Windows® Vista.

In the Installation and Configuration chapter:

- Installation section:
  - Added a note about upgrading on Windows® 7 and Windows Server® 2008.
  - Added information about the HPET Timer.
  - Added a note about the Security Alert- Driver Installation message.

In the OA&M Software chapter:

- New Tools section:
  - Added information about the HPETTOOL Utility.

In the Supported Hardware chapter:

- Removed Section 9.2, "Dialogic® Digital Station Interface Products" because these products have been discontinued and are no longer supported.

Document Rev 02 - published November 2007

Made global changes to reflect the Dialogic brand.

Added this new Document Revision History section.

In the System Requirements section:

- Added information about Kernel Memory and updated the information about Physical Address Extensions in Section 2.1, "Basic Hardware Requirements", on page 15.
- Updated the Operating System information in Section 2.2, "Basic Software Requirements", on page 16.
- Deleted Alcatel 40xxIP e-Reflexes, Alcatel OmniPCX Enterprise call server, and Microsoft Live Communications Server from Table 2, "Devices Tested for Interoperability with Dialogic® HMP Software", on page 18.
- Updated the interoperability status of the respective products from Validated to Tested in Table 2, "Devices Tested for Interoperability with Dialogic® HMP Software", on page 18.

In the Features section:

- Added new Section 3.7, "Support for SS7 Products and SIUs", on page 27.

In the Supported Hardware section:

- Added new Section 8.2, "Dialogic® Signaling Products", on page 57.

In the Documentation section:

- Added SS7 Signaling to Table 5. "User Documentation Feature Support."
- Added the Dialogic® Global Call SS7 Technology Guide to Section 10.5, "Programming Libraries Documentation", on page 69.

Document Rev 01 - published August 2006

Initial Version of document.

About This Publication

The following topics provide information about this publication.

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

Applicability

This Release Guide provides information about the features, system requirements, and release documentation for Dialogic® Host Media Processing Software Release 3.0WIN.

Intended Audience

This document is intended for users of Dialogic® Host Media Processing Software Release 3.0WIN. This includes the following types of customers:

- System Integrators
- Toolkit Developers
- Independent Software Vendors (ISVs)
- Original Equipment Manufacturers (OEMs)

How to Use This Publication

The information found 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 system software and hardware requirements for the Dialogic® Host Media Processing Software.
- Chapter 3, "Features" - describes the features supported in this release.
- Chapter 4, "Installation and Configuration" - provides information about installation and configuration.
- Chapter 5, "OA&M Software" - describes the operation, administration, maintenance, and diagnostics supported in this release.
- Chapter 6, "Programming Libraries" - describes the programming libraries that are available as part of this release.
- Chapter 7, "Supported Applications" - describes the MSML Media Server Software.

- Chapter 9, "Demonstration Software" - describes the demonstration programs provided in this release.
- Chapter 8, "Supported Hardware" - provides a list of all the hardware supported in this release.
- Chapter 10, "Documentation" - provides a list of the documents that accompany this release, either on the CD or downloadable from the Dialogic Support web site.

Related Information

See the following for additional information:

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

Release Overview 1

This chapter provides an overview of the Dialogic® Host Media Processing (HMP) Software as well as a high-level overview of the products and features that are newly supported in Dialogic® Host Media Processing Software Release 3.0WIN.

Dialogic® HMP Software Release 3.0WIN performs media processing tasks on general-purpose servers based on Dialogic® architecture without the need for specialized hardware. The software provides media services that can be used to build flexible, scalable, and cost-effective next-generation IP media servers.

Uses Dialogic® DM3 Architecture

Dialogic® HMP Software is a Dialogic® communications building block technology. When installed on a system, the software performs like a Dialogic® board with DM3 architecture to the customer application, but all media processing takes place on the host processor. To help customers accelerate their time to market and migrate their existing applications to IP, the software also supports two direct APIs: Dialogic® R4 for media processing and Dialogic® Global Call (GC) for call control.

Supports Industry Standards

Dialogic® HMP Software supports the industry-standard H.323 protocol and Session Initiation Protocol (SIP) protocol for call control. Dialogic® HMP Software supports Real-time Transport Protocol (RTP) and Real-Time Control Protocol (RTCP) for media streaming and control over IP in G.711, G.726, G.723.1, G.729, or G.729B audio formats; H.263 video format; and T.38 and V.17 formats for fax over IP. In addition, Dialogic® HMP Software Release 3.0WIN introduces MSML media server software that enables the control of media server resources using MSML syntax embedded in SIP messages.

Media Processing Functions

Dialogic® HMP Software supports voice play and record, conferencing, video messaging, fax, and speech integration functions. Support has been expanded to support up 720 G.711 channels performing voice play and record functions.

Dialogic® HMP Software

Dialogic® HMP Software is implemented as a Windows® operating system kernel-mode driver that runs at real-time priority. The software is optimized to run on Intel processors. Dialogic® HMP Software can be installed and upgraded similar to other software.

Licensing

Dialogic® HMP Software media resources are provided via a license file. This file contains authorization for a combination of call control and media processing features. The license can either be associated with the host machine via its MAC address (Host ID) or, if you are using one or more of the supported Dialogic® DNI or DSI boards, associated with one of the boards in the system. Locking the license to a board allows you to transfer the license to another host by moving the board to that host.

To allow customers the flexibility of choosing combinations of media processing, customers may choose the exact resources they need for their solution. Resources may also be downloaded as the customer's system grows.

Direct PSTN and PBX Connectivity

Dialogic® HMP Software supports a streaming interface to the public switched telephone network (PSTN) by bridging to Dialogic® Digital Network Interface boards.

Analog Device (Springware) Support

The following Springware boards are supported:

- Dialogic® D/4PCIUFEQ
- Dialogic® D/4PCIU4SEQ
- Dialogic® D/80PCIE-LS

While these boards have similar functionality to the current analog Dialogic® JCT boards, they add media streaming capability between their analog interfaces and HMP. The Dialogic® D/80PCIE-LS board also provides HMP media streaming with on-board voice devices. Refer to the Section 3.8, "Analog Device (Springware) Support", on page 29 for more information about these boards. For information about the features of the Dialogic® JCT boards, go to http://www.dialogic.com/products/media/jct/default.htm.

Virtualization Support

Virtualization is supported using VMware® ESXi 4.0 VMware ESXi 4.0 Update 1 Installable. This release offers IP-only support with the following supported versions of Windows Server® 2003:

- Windows Server® 2003 (Standard or Enterprise Edition) with Service Pack 1 or 2
- Windows Server® 2003 R2 (Standard or Enterprise Edition)
- Windows Server® 2003 R2 (Standard or Enterprise Edition) with Service Pack 2

Release Highlights

In addition to the features supported by Dialogic® HMP Software Release 2.0WIN, this release of the Dialogic® HMP Software supports the following new features:

- Multimedia (Audio/Video)

- Secure Real-time Transport Protocol (SRTP)
- SIP Transport Layer Security (TLS)
- MSML media server software for use by remote applications
- Conferencing (CNF) API
- SS7 products and SIUs

Refer to Chapter 3, “Features” for additional information about these new features.

Software Restrictions

The following restrictions apply to Dialogic® HMP Software Release 3.0WIN:

- This software is subject to the U.S. Export Administration Regulations and other U.S. law, and may not be exported or re-exported to certain countries (currently Burma, Cuba, Iran, Libya, N. Korea, Sudan and Syria) or to persons or entities prohibited from receiving U.S. exports (including Denied Parties, Specially Designated Nationals, and entities on the Bureau of Export Administration Entity List or involved with missile technology or nuclear, chemical or biological weapons).
- Support for Quality of Service (QoS) under Global Call: On Windows® 7 or Windows Server® 2008 operating systems, the setting of Type of Service (ToS) or Differentiated Services Code Point (DSCP) in the IP header of outgoing packets is no longer supported by default under the Windows® Sockets (Winsock) API.

  Because the Dialogic® HMP Global Call Software currently uses the Winsock API, setting the ToS or DSCP parameters using IPPARM_IPMPARM/PARMCH_TOS will be ignored by the operating system. To set the type of service values in Windows® 7 or Windows Server® 2008 operating systems, a Quality of Service (QoS) policy must be created with the appropriate DSCP values that can be applied to all applications or a specific application without modifying its code. A QoS policy can be configured and applied in several ways on Windows® 7 or Windows Server® 2008 operating systems. The following example shows configuring a simple QoS policy:

  1. Log in with Administrator privileges.
  2. Launch the Microsoft Management Console by selecting Select Start > Run, typing mmc and pressing Enter.
  3. From File menu, select Add/Remove Snap-in > Group Policy Object Editor (Available snap-ins:) and click Add.
  4. Click Finish on the Select Group Policy Object pop-up window.
  5. Click OK on Add/Remove Snap-ins window to return to the main Console screen.
  6. Expand tree view in the left pane to show the Console Root/Local Computer Policy/Computer Configuration/Windows Setting/Policy-based QoS.
  7. Right click the "Policy-based Qos" item in the tree view and then select Create new Policy from context menu.
  8. In the Policy-based QoS window, enter any policy name text and change the DSCP value. Click Next.
  9. Select how this policy should be applied to applications and then click Next.
  10. Select how you want to apply this policy to source and destination addresses and then click Next.

11. Change “Select the protocol this QoS policy applies to:” from default "TCP" to "UDP". If you want to specify a specific range of source port addresses, select “From this source port number or range:” Specify the range of RTP and RTCP port numbers used by the IPM devices. The default range is from 49152 to the number of IPM devices x 2 (RTP+RTCP). Click Finish.
12. Close the Microsoft Management Console. Click Yes to save the settings.
13. Start the Dialogic® HMP Software.

Refer to the *Dialogic® Global Call IP Technology Guide*, *Dialogic® Global Call API Library Reference*, and the *Dialogic® Global Call API Programming Guide* for information about Global Call. For more information about Windows policy-based Quality of Service (QoS), refer to the Microsoft web site.

- There is a restriction when assigning Dialogic® HMP Software licenses to a NIC in systems with more than five NICs. The HMP license must be tied to one of the first five NICs listed in the NIC priority list, otherwise the license will fail to activate. The NIC priority list can be changed under Advanced Settings of Manage Network Connection in the Windows Control Panel, allowing any NIC to be used for the HMP license.

System Requirements 2

The basic requirements to install and run Dialogic® Host Media Processing Software Release 3.0WIN are described in the following sections:

- Basic Hardware Requirements 15
- Basic Software Requirements 16
- Equipment and Configurations Tested 17
- Ordering the Product 20

2.1 Basic Hardware Requirements

The basic hardware requirements for this release are:

- Processor: See Table 1 for processor recommendations.
- Memory: 512 MB to 2 GB of RAM, depending on channel density and resource configuration.

  Note: A minimum of 2 GB of RAM is required when running Dialogic® HMP Software on Windows® 7 or Windows Server® 2008.
- Kernel Memory: Nonpaged: Dialogic® HMP Software uses this space for allocating memory for its resources. Nonpaged memory requirements do not include the operating system. Operating system usage varies based on system drivers. The Dialogic® HMP Software requires above 200 MB of non-paged memory or system performance may show degradation or failure (Dialogic® Configuration Manager (DCM) will not start).

  Note: Under Windows® 7 and later, the non-paged memory limit has been increased due to optimizations by Microsoft.
- Disk Space: 500 MB of disk space is required for full installation of Dialogic® HMP Software Release 3.0WIN. For multimedia recording, disk space depends on the application program, but can be estimated using the following guidelines:
  - 160 KB required for 10 seconds of audio at maximum of 128 kbps bit rate (linear PCM format)
  - 160 KB required for 10 seconds of video at maximum of 128 kbps bit rate
  - 80 KB required for 10 seconds of video at maximum of 64 kbps bit rate
- HPET Timer: A system with an HPET Timer is required when using Dialogic® HMP Software with Windows® 7 or Windows Server® 2008. For the most desirable results, select systems that have BIOS support for the HPET Timer. Dialogic provides an HPETTOOL utility to assist in determining motherboard compatibility. Refer to the *Dialogic® HMP Software Release 3.0WIN Software Installation Guide* for instructions about using this utility.

Table 1. Processor Recommendations

Number of User Sessions[1]	Processor Type and Clock Speed	
	G.711 (20 msec Frame)	G.729AB[2]
Up to 64	Intel Pentium 4, 2.0 GHz	Dual Intel Xeon, 2.0 GHz
Up to 96	Single Intel Xeon, 2.4 GHz	Dual Intel Xeon, 3.06 GHz
Up to 120	Dual Intel Xeon, 2.4 GHz	Dual Intel Xeon, 3.2 GHz w/HT or Dual Core Extreme 3.2 GHz w/HT
Up to 200	Dual Intel Xeon, 3.06 GHz	Dual Intel Xeon, 3.6 GHz w/HT or Dual Core Extreme 3.2 GHz w/HT
Up to 500	Dual Intel Xeon, 3.6 GHz w/HT or Dual Core Extreme, 3.2 GHz w/HT	—
Up to1000[3]	Quad Core Intel Xeon, W5580, 3.2 GHz w/HT	

1.RTP and Voice functionality
2.Two frames per packet
3.For Windows® 7 or Windows Server® 2008 operating systems

- IP Network Interface: 100 Base-T Network Interface Card (NIC)

  Note: For 120 channels or higher, using a 1000Base-T NIC, while still using a 100Base-T Network, is recommended. In general, better performance will be realized by using a 1000Base-T NIC, even for configurations of less than 120 channels.

- Dialogic® DNI Boards and Physical Address Extension (PAE): PAE is not supported on servers having over 4 GB of RAM with a Dialogic® DNI board PCIe form factor and over 2 GB of RAM with a Dialogic® DNI board PCI form factor. If these configurations are present, then the Dialogic® HMP system will not start. To prevent this, PAE must be disabled.

  Run the following commands to disable PAE for Windows® 7 or Windows Server® 2008 operating systems:

  ```
  bcdedit /set pae ForceDisable
  bcdedit /set nx AlwaysOff
  ```

  Note: Setting “nx” to “AlwaysOff” will disable Data Execution Prevention (DEP). When DEP is enabled, Windows also enables PAE on machines that are capable of supporting PAE. When PAE is enabled, Windows may use 36-bit physical addressing instead of 32-bit, depending on the machine memory.

2.2 Basic Software Requirements

Note: Dialogic® HMP Software Release 3.0WIN has been tested only on systems with NT file systems (NTFS). This release of the Dialogic® HMP Software has not been tested on systems with FAT 32 file systems.

The basic software requirements for this release are:

- Operating System - one of the following:

  Note: Only 32-bit versions of these operating systems are currently supported.

  – Windows® 7

  – Windows Server® 2008 with Service Pack 2

  – Windows® 2003 Web Edition Service Pack 2

  – Windows® XP Professional with Service Pack 3

  – Windows® XP Professional with Service Pack 2

  – Windows Server® 2003 (Standard or Enterprise Edition) with Service Pack 1 or Service Pack 2

  – Windows Server® 2003 R2 (Standard or Enterprise Edition)

  – Windows Server® 2003 R2 (Standard or Enterprise Edition) with Service Pack 2

- Compilers:

  – Development Environment Microsoft® Visual Studio 2005 with Visual Studio 2005 Service Pack 1 and Visual Studio 2005 Service Pack1 Update for Windows® Vista. This compiler is required for application development under Windows® 7 or Windows Server® 2008.

  Note: Microsoft® Visual Studio 2005 (all editions) includes C++ compiler version 8.0. Customer applications with Windows® 7 or Windows Server® 2008 should use version 8. Applications must be compiled as 32-bit binaries.

  Note: By default with Windows® 7 or Windows Server® 2008, executables will run without administration privileges due to User Account Control (UAC) restrictions, even if the user is logged in with administrator privileges. This means that Dialogic applications that required administration privileges and ran successfully on an earlier Windows version may now fail with “permission denied” errors. The developer can overcome this situation by elevating the application execution level by granting an administrator token. One of the mechanisms for doing so is by providing the appropriate application manifest schema settings. Consult the Microsoft® .NET Framework Developer's Guide for details.

  – Microsoft® Visual Studio® 6.0 with Service Pack 5

2.3 Equipment and Configurations Tested

The following topics describe equipment and configurations tested:

- Multimedia Gateways
- Multimedia and Voice IP Devices
- Configurations Tested

2.3.1 Multimedia Gateways

Dialogic® HMP Software Release 3.0WIN has been tested with the following multimedia gateways:

- Dilithium Networks DTG 2000 Multimedia Gateway (SIP version)

  Note: The Dilithium gateway was configured so that a Dialogic® HMP Software Media Server was one of the SIP endpoints running a video messaging test application. A video call was placed from the 3G-324M side of the DTG 2000 and terminated on one of the SIP endpoints of the Dialogic® HMP Software Media Server. A 64 kbps H.263/G.711 video/audio stream was opened and inband DTMF tones were used to control the playing and recording of the media stream on the Dialogic® HMP Software Media Server.

2.3.2 Multimedia and Voice IP Devices

Dialogic® HMP Software has been validated and/or tested (except as noted) for interoperability with the devices listed in Table 2.

Table 2. Devices Tested for Interoperability with Dialogic® HMP Software

Manufacturer & Product	Device Type	SIP	H.323
Avaya 4600 Series	IP Phone	N/A	Tested
Avaya G700 and S8300	IP-PBX	Tested	Tested
Avaya IP Office 4x0	IP-PBX	N/A	Tested
Cisco 7900 Series	IP Phone	Tested	Tested
Cisco 2600, 3600 Series	Gateway/Switch	Tested	Tested
Cisco AS5300/5350/5400	Gateway/Switch	Tested	Tested
Cisco Call Manager 3.3, 4.0, and 6.0	Gateway/Switch	Tested	Tested
Cisco Call Manager Express	Gateway/Switch	Tested	Tested
DyLogic Mirial Standard SIP	Video SoftPhone	Tested	N/A
Dilithium DTG2000 Video Gateway	Gateway	Tested	N/A
Grandstream BudgetTone 1xx Series	IP Phone	Tested	N/A
Dialogic® 1000 Media Gateway Series (formerly Dialogic® PBX-IP Media Gateway)	Gateway	Tested	Tested
Dialogic® 2000 Media Gateway Series (formerly Dialogic® T1/E1-IP Media Gateway)	Gateway	Tested	Tested
Microsoft® NetMeeting®	Soft Client	N/A	Tested
Microsoft® Messenger	Soft Client	Tested	N/A
Polycom Soundpoint IP family phones	Audio	Tested	Tested
RadVision Gatekeeper 4.0.0.28	Audio	N/A	Tested
Notes: Tested = Interoperability has been tested by Dialogic. N/A = Not applicable.			

Table 2. Devices Tested for Interoperability with Dialogic® HMP Software (Continued)

Manufacturer & Product	Device Type	SIP	H.323
Siemens HiPath	IP-PBX	N/A	Tested
SJ Labs SJphone VOIP softphone version 1.10	Soft Client	Tested	N/A
Notes: Tested = Interoperability has been tested by Dialogic. N/A = Not applicable.			

2.3.3 Configurations Tested

The reference configurations listed in Table 3 have been successfully tested with CPU utilization of 50% or less.

Table 3. Resource Configurations Tested

Configuration‡	Processor	R	E	V	C	F	S	I	M	HIB
IVR (Low bit rate)	Dual Intel Xeon, 3.6 GHz (w/HT)	120	120	120	—	120	—	—	—	—
IP Media Server	Intel Pentium 4, 2.0 GHz	23	11	23	—	4	23	—	—	—
IP Media Server	Intel Pentium III, 1.26 GHz	23	11	23	—	4	23	—	—	—
IVR/UM/Conf/Speech High Density Contact Center	Dual Intel Xeon, 3.6 GHz (w/HT)	240	120	240	180	24	120	—	—	DNI1200TEP (2)
Gateway	Intel Celeron D, 2.13 GHz	60	60	60	—	—	—	—	—	DNI601TEP
Gateway	Dual Socket, Dual Core Intel Xeon, 2.8 GHz	360	—	360	—	—	—	—	—	DNI1200TEP (3)
Media Server	Dual Socket, Dual Core Intel Xeon, 2.8 GHz	240	120	240	32	16	240	—	—	—
IVR (G.711)	Dual Socket, Dual Core Intel Xeon, 2.8 GHz	500	—	500	—	—	—	—	—	—
Conferencing	Dual Intel Xeon, 3.6 GHz (w/HT)	400	—	100	400	—	—	—	—	—
Video Services	Intel Pentium D, 3.2 GHz (w/HT)	240	—	240	—	—	—	240	120	—
Legend: R = RTP (basic) E = Enhanced RTP V = Voice C = CNF Conferencing F = Fax S = Speech I = IP Call Control M = Multimedia HIB = Dialogic® HMP Interface Board ‡ = Not all resources are used simultaneously.										

Note: Dialogic® HMP Software Release 3.0WIN has been tested only on systems with NT file systems (NTFS). Dialogic® Host Media Processing Software Release 3.0WIN has not been tested on systems with FAT 32.

2.4 Ordering the Product

The following table lists the resources that are available through flexible licensing.

Table 4. Media and Other Resources

Resource	Resource Code	Number of Resources Available	Resource Description
RTP G.711	r	0-500	Provides the capability of streaming digitized voice over RTP using the G.711 coder with 10, 20, 30 ms frames. The number of RTP resources for any given configuration should be greater than or equal to the number of voice, enhanced voice, conferencing, continuous speech processing, or fax resources (whichever requires the highest number of resources).
Voice	v	0-500	Basic voice ports that allow you to control volume, record with Automatic Gain Control (AGC), and DTMF and user-defined tone detection, including RFC 2833 and H.245 UII. Each voice port requires an RTP G.711 port.
Enhanced RTP	e	0-200	Adds the capability of streaming voice over RTP using the G.723.1, G.729a, and G.729b coders.
Conferencing	c	0-500	Conferences parties using advanced features such as coach/pupil mode, tone clamping, and active talker notification. Conferencing resources require RTP G.711.
Speech Integration (Continuous Speech Processing)	s	0-240	Speech integration capabilities that enable you to integrate the HMP Software product with speech engines for Automatic Speech Recognition (ASR) and Text-To-Speech (TTS) support using the Dialogic® Continuous Speech Processing APIs.
Fax Termination	f	0-120	Supports T.38 and V.17 fax origination and termination sessions. Fax requires RTP G.711.
IP Call Control	i	0-2000	Supports first party and third party IP call control. First party call control manages a media session that is initiated and terminated by Dialogic® HMP Software. Third party call control enables one entity to create, modify, or terminate a media session between two or more endpoints where the call control signaling and media exchange are independently managed.
Multimedia	m	0-120	Supports play, record, and synchronization of voice and H.263 format video in a multimedia stream. Includes video I-frame detection to trigger start of record as well as transmit-oftone notification when recording begins. Multimedia requires RTP G.711

The Dialogic® HMP Software provides a high level of flexibility in choosing media processing configurations. However, all possible combinations of the resources listed in Table 4 are not supported. Refer to the following rules and the automatic license fulfillment tool for determining valid configurations.

Resource Configuration Rules

Use the following rules to determine a valid resource configuration:

- The number of Enhanced RTP resources must be less than or equal to the number of RTP G.711 resources.

  Note: Enhanced RTP resources are used for G.723.1, G.729, and G.729B coders.

- The number of Speech Integration (continuous speech processing) resources must be less than or equal to the number of Voice resources.
- IP Call Control resources must be equal to or greater than the total RTP G.711 resources when selected.
- Multimedia resources require an RTP resource.

For more information about the Dialogic® HMP Software product see http://www.dialogic.com/products/ip_enabled/hmp_software.htm.

Features 3

This chapter lists and describes the new features that are introduced by Dialogic® Host Media Processing Software Release 3.0WIN as well as existing features from previous releases that continue to be supported. This information is provided in the following sections:

- Multimedia (Audio/Video) 23
- Secure Real-time Transport Protocol (SRTP) 25
- SIP Transport Layer Security (TLS) 26
- MSML Media Server Software for Remote Applications. 26
- Higher Channel Density Support 27
- Dialogic® Conferencing (CNF) API 27
- Support for SS7 Products and SIUs 27
- Analog Device (Springware) Support 29
- Other Features 38

3.1 Multimedia (Audio/Video)

Note: All multimedia-related features are new for this release.

- Multimedia record and playback with basic playback control and synchronized audio and video.
- Record from RTP stream to multimedia file. Play from multimedia file into RTP stream while maintaining synchronization between audio and video.
- Supports the H.263 (profile 0, level 30) video codec. Level 30 supports CIF, QCIF and sub-QCIF resolution decoding. It is capable of operation with a bit rate up to 6 x (64,000) = 384,000 bits per second, with a picture decoding rate up to (30,000)/1001 pictures per second.
- Supports the following audio codecs for RTP:
  - G.711
  - G.723.1
  - G.729A (compatible with G.729 format)
  - G.729AB (compatible with G.729B format)

  Note: Audio codec is licensed separately.
- Supports the following video picture formats:
  - Common Intermediate Format (CIF) picture size (352 pixels by 288 pixels)
  - Quarter Common Intermediate Format (QCIF) picture size (176 pixels by 144 pixels)
  - Sub-QCIF picture size (128 pixels by 96 pixels)

- Supports a proprietary video file format and the Linear PCM (128 kbps) audio file format.

- **Multimedia File Conversion Utilities:** These utilities provide off-line conversion of multimedia files. They can be downloaded from the following web site (check this web site periodically for updates to the conversion tools and their capabilities):

  http://www.dialogic.com/support/helpweb/hmp/hmpWin/hmp30/omf/default.htm

  The conversion utilities perform CPU-intensive tasks and should only be used when sufficient CPU capacity is available and when they won't impact other operations on the system. For example, they should not be used while performing audio/video operations or when processing audio/video calls, as this can impact the performance and operation of the system.

  - ***mmconvert***: Conversion of multimedia data from Audio Video Interleave (AVI) Type-2 files with DVSD and DV25 encoded video in PAL 720 x 576 or NTSC 720 x 480 video format, with PCM linear 16-bit (any rate) mono or stereo audio format, to Dialogic proprietary multimedia file format. Conversion includes selectable picture format (sub-QCIF, QCIF, CIF), aspect ratio adaptation, bit rate, and frames per second (30, 15, 10, 6).

  Note: The *mmconvert* utility does not support conversion in the reverse direction (from Dialogic proprietary multimedia file format to AVI file format).

  - ***hmp3gp***: Conversion of multimedia data from Dialogic proprietary multimedia file format to 3rd Generation Partnership Project (3GPP) file format conforming to 3GP Release 4 file format (.3gp), conforming to 3GPP specifications. The generated 3GP file contains two tracks: a video track with H.263 bit-stream video data, and an audio track with Global System for Mobile communication Adaptive Multi-Rate Narrow Band (GSM-AMR-NB) audio encoded at a bitrate of 12.2 kbps. No translating or resizing is done, so the destination frame rate and the picture size will be the same as the source.

  Note: The *hmp3gp* utility does not support conversion in the reverse direction (from 3GPP to Dialogic proprietary multimedia file format).

- Play to and record from SIP devices, depending upon capability of device (audio or audio/video). Play video only if no audio is required. Play audio only for non-video devices.

- Supports existing Quality of Service (QoS) audio alarms through the Dialogic® IPML API for the voice portion of multimedia stream.

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

- Play Dialogic® Voice API audio files in a multimedia session. You can play Dialogic® Voice API audio files in a multimedia session as long as tight synchronization with video is not required (as when playing with a video menu or status display). In this case, the "ipm" device in a multimedia session will listen to the "dxxx" device to which the Dialogic® Voice API is playing an audio file. This overrides any audio stream (but not video) from the "mm" device in the multimedia session.

- Licensing of Dialogic® HMP Software multimedia resources. One complete multimedia session requires one multimedia resource license and one RTP resource license. The multimedia resource provides the video/audio play/record capability, while the RTP resource provides one RTP channel for the audio stream and one RTP channel for the video stream.

- The RTP channel for a video stream cannot be used for an audio stream.
- The programming libraries that provide multimedia-related enhancements and additions are identified in the following outline of multimedia functionality:
  – The Dialogic® Multimedia API Library records and plays the multimedia data using a multimedia device.
  – The Dialogic® Device Management API Library is used to connect the multimedia device with an IP media device.
  – Multimedia record and playback between Dialogic® HMP Software and remote IP endpoints is accomplished by using both the multimedia and IP media devices.
  – The Dialogic® IP Media Library API Library provides the IP multimedia session control.
  – The Dialogic® Global Call API Library must be used in third party call control (3PCC) mode. The Global Call API library provides multimedia support through SIP third party call control. A convenience API is available to process the Session Description Protocol (SDP).

  See the *Dialogic® Multimedia API Library Reference* and the *Dialogic® Multimedia API Programming Guide* for details about the Dialogic® Multimedia API. See the *Dialogic® Host Media Processing Software Release 3.0WIN Administration Guide* for Licensing information.

3.2 Secure Real-time Transport Protocol (SRTP)

Real-time Transport Protocol (RTP) is commonly used for the transmission of real-time audio or video streams in Internet telephony applications. Secure RTP (SRTP) is an enhancement to RTP that provides confidentiality, message authentication, and replay protection for RTP and the companion Real Time Control Protocol (RTCP). “Replay protection” is protection against the interception of an RTP packet and later reintroduction into the packet stream.

SRTP can be thought of as residing between an RTP application and the transport layer. On the sending side, SRTP intercepts an RTP packet and forwards an equivalent SRTP packet. At the receiving side, SRTP receives an SRTP packet and the equivalent RTP packet is passed up the protocol stack.

In SRTP, the payload (and padding) of RTP and RTCP packets may be authenticated or encrypted. RTCP packets are always authenticated, but encryption on RTP and RTCP packets and authentication on RTP packets is optional.

Many different encryption and message authentication algorithms exist, but RFC 3711 specifies the following default pre-defined algorithms:

- For encryption, the pre-defined cipher is the Advanced Encryption Standard (AES) operating in Segmented Integer Counter Mode.

  Note: Currently, only Segmented Integer Counter Mode is supported.

- For message authentication and integrity, the pre-defined authentication transform is HMAC-SHA1 as described in the *HMAC: Keyed-Hashing for Message Authentication* IETF publication, RFC 2104.

An important part of any encryption scheme is the generation of the keys used to encrypt the information. SRTP relies on an external key management system to provide the master key and master salt. A master key is a random bit string from which session keys (used directly in the cryptographic transforms) are derived. A master salt is also a random bit string used to provide even greater security.

SRTP incorporates a "key derivation algorithm" that uses the master key, master salt and packet index to generate the session keys that are used directly for encryption or message authentication. The rate at which new session keys are applied, that is, the "key derivation rate" can also be defined.

Note: The maximum number of SRTP channels in a system is 500.

For additional information, see Section 6.7, "Dialogic® IP Media Library API Library", on page 51 and the *Dialogic® IP Media Library API Programming Guide and Library Reference*.

3.3 SIP Transport Layer Security (TLS)

Transport Layer Security (TLS) provides the ability to authenticate and encrypt TCP-based call control using a variety of different key exchange, authentication, encryption, and message authentication code algorithms. TLS provides for its own authentication and key management, as well as encryption. TLS can provide a secure way for two devices mutually using SRTP to exchange the necessary setup information, including the SRTP keys (using SDP Secure Descriptions). This capability is only required when the Dialogic® HMP Software call control stack is used. When using a call control stack external to Dialogic® HMP Software, you will need to provide your own call control security.

Note: The maximum number of TLS channels in a system is 500.

For additional information about TLS, see the *Dialogic® Global Call IP for Host Media Processing Technology Guide*.

3.4 MSML Media Server Software for Remote Applications

Dialogic® HMP Software Release 3.0WIN supports MSML media server software that uses the common interfaces provided by the Global Call call control software and the Session Initiated Protocol (SIP) to allow media processing on a Media Server (MS) from a remote agent such as an Application Server (AS).

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

3.5 Higher Channel Density Support

Dialogic® HMP Software Release 3.0WIN supports up to 720 instances of G.711-based voice resources in an IVR configuration and up to 200 instances of low bit-rate voice resources.

For additional information about channel densities supported, see Chapter 2, “System Requirements”.

3.6 Dialogic® Conferencing (CNF) API

Dialogic® HMP Software Release 3.0WIN supports a new Dialogic® CNF Conferencing API that adds:

- Asynchronous programming model support - This model enables multiple channels to be handled in a single process and supports higher density conferencing solutions.
- Additional flexibility in making and breaking conference connections.
- Ability to apply echo cancellation via an application command to an incoming PSTN audio media stream whenever that stream is directed to a conferencing resource.

Note: The maximum number of parties allowed in a single conference is 254. This applies to conferencing applications developed using the Dialogic® CNF conferencing API as well as the Dialogic® DCB conferencing API.

For additional information about CNF conferencing, see the *Dialogic® Conferencing API Library Reference* and the *Dialogic® Conferencing API Programming Guide*.

3.7 Support for SS7 Products and SIUs

Dialogic® SS7 boards provide on-board support for SS7 common channel signaling protocols with a number of digital line interfaces (T1/E1/J1) and a H.100 or H.110 PCM highway that supports connection to a wide range of voice, data, and fax boards. Dialogic® HMP Software Release 3.0WIN supports the following Dialogic® SS7 PCI boards:

Dialogic® SPCI2S
: This is a PCI board that features two T1/E1 interfaces, an H.100 PCM highway, two synchronous serial interfaces (V.11/V.35), and four SS7 links.

Dialogic® SPCI4
: This is a PCI board that features four T1/E1 interfaces, an H.100 PCM highway, wo synchronous serial interfaces (V.11/V.35), and four SS7 links.

Dialogic® SS7HDP
: This is an SS7 PCI board that provides up to four T1/E1 interfaces, V.11(V.35-compatible) serial ports, an H.110 PCM highway, and 64 SS7 links.

Signaling Interface Units (SIUs)

Dialogic® Signaling Interface Units (SIUs) are SS7 server solutions that provide a convenient and cost-effective way to add SS7 connectivity to an existing multi-chassis system or to enable an application with the necessary protocols for mobile wireless or intelligent networks. Dialogic® HMP Software Release 3.0WIN provides support for the following SIUs:

Dialogic® SS7G21 (in SIU Mode)

The Dialogic® SS7G21 is fitted with Dialogic® SPCI4 or SPCI2S boards with a system maximum of 12 SS7 links and provides a form, fit, and function replacement for the Dialogic® SIU520 signaling gateway on a higher performance platform. A Dialogic® SS7G21 unit may be purchased with either:

- 1, 2, or 3 Dialogic® SPCI2S boards (4 SS7 links, 2 T1/E1 interfaces, two V.11 serial ports per board).
- 1, 2, or 3 Dialogic® SPCI4 boards (4 SS7 links, 4 T1/E1 interfaces per board). Supplied in a 2U carrier-grade chassis, the Dialogic® SS7G21 unit provides SS7 connectivity for multichassis call control, wireless, or Intelligent Networking (IN) applications.

Dialogic® SS7G22 (in SIU Mode)

The Dialogic® SS7G22 is fitted with Dialogic® SS7HDP boards and offers significantly greater performance and link density than the Dialogic® SS7G21unit. A Dialogic® SS7G22 unit may be purchased with 1, 2, or 3 Dialogic® SS7HDP boards (64 SS7 links, 4 T1/E1 interfaces per board) with a system maximum of 128 SS7 links. Supplied in a 2U carrier-grade chassis, the Dialogic® SS7G21 unit provides SS7 connectivity for multichassis call control, wireless, or Intelligent Networking (IN) applications.

Features

Dialogic® HMP Software Release 3.0WIN provides the following features for Dialogic® SS7 boards and SIUs:

Dialogic® SPCI2S

This is a PCI board that features two T1/E1 interfaces, an H.100 PCM highway, two synchronous serial interfaces (V.11/V.35), and four SS7 links.

Dialogic® SPCI4

This is a PCI board that features four T1/E1 interfaces, an H.100 PCM highway, and four SS7 links.

Dialogic® SS7HDP

This is an SS7 PCI board that provides up to four T1/E1 interfaces, V.11 (V.35-compatible) serial ports, an H.110 PCM highway, and 64 SS7 links.

Global Call API support

Supports the development of call control applications that use SS7 technology. See the Global Call SS7 Technology Guide for more information.

> ***Note:*** Global Call SS7 binaries are linked with the dynamic linked libraries in the Dialogic® SS7 DSI Development Package. Global Call SS7 customers must use the Dialogic® SS7 DSI Development Package version 5.0 or later. If an earlier version is used, the Global Call SS7 server will not start during download.

Dialogic® HMP Software Release 3.0WIN works in conjunction with the Dialogic® SS7 Development Package, which is not part of the Dialogic® HMP software. The Dialogic® SS7 Development package is available from the following URL:

http://www.dialogic.com/support/helpweb/signaling

This package must be installed to provide support for the following SS7 layers and protocols:

- MTP2
- MTP3
- ISUP
- TUP (ITU and China GF001-9001 variants)
- SCCP
- TCAP
- IS41
- MAP
- INAP

For more information, see the “Support for SS7 Functionality” section of the *Dialogic® Host Media Processing Software Release 3.0WIN Release Update*.

3.8 Analog Device (Springware) Support

The Dialogic® HMP Software Release 3.0WIN supports the following Dialogic® Springware boards: Dialogic® D/80PCIE-LS and the Dialogic® D/4PCIUFEQ and Dialogic® D/4PCIU4SEQ (herein referred to as D/4PCIU boards). These boards are analog loop start, RoHS commercial product 6/6 boards, used for developing advanced communications applications extending the HMP media capabilities to the analog loop-start PSTN network.

The Dialogic® D/4PCIUFEQ board provides four-port basic voice processing and DSP-based Group 3 fax support (DSP fax or SoftFax). The Dialogic® D/4PCIU4SEQ board provides four-port basic voice processing with continuous speech processing (CSP). Both are half-length PCI Express form factor boards. The boards have a subset of the features and functionality of the current analog Dialogic® JCT boards. They also provide HMP media streaming between their four analog interfaces and HMP; however, they have no CT Bus connectivity.

Note: The Dialogic® D/4PCIU board will not function with any other board in the system, and must be the only board in the system. This is a permanent, stand-alone restriction.

The Dialogic® D/80PCIE-LS Media Board is a full-length PCI Express form factor, eight-port, full-featured JCT board. In addition, it provides media streaming between HMP and its eight analog interfaces and eight on-board voice devices.

The following sections provide information specific to using these Springware products with HMP software.

3.8.1 Bridge Device

The analog devices, like other Dialogic® HMP interface boards, have a bridge device that enables media streaming between HMP and the board. The boards have their bridges enabled by default.

With the Dialogic® D/80PCIE-LS board, both the board's analog interfaces and on-board voice devices are connected to the CT Bus. The CT Bus provides the fabric for intra-board connectivity or, in the case of other CT Bus boards in the system, for inter-board connectivity. In addition, the bridge device can be used to stream media to and from HMP. Traditional CT Bus routing and HMP streaming are accomplished using the standard run-time routing APIs. On-board voice and analog interface half-duplex streaming connections from HMP media and IP media devices are performed using the Voice API TDM Routing functions. HMP half-duplex streaming connections from on-board voice and analog interface devices are performed using the traditional TDM Routing APIs for their respective technologies.

Alternatively, half- or full-duplex bridging connections between HMP and bridge-capable, on-board devices on these analog-interface boards can be accomplished using the Dialogic® Device Management API function **dev_Connect( )**. On analog boards, the SRL device handle has the dual purpose of acting on either the proper voice device, or the analog-interface device. In order to differentiate analog voice devices from analog front-ends, a new connection type value, DM_ANALOG_INTF, is added to the **dev_Connect( )** function's connType parameter. This value specifies that the Springware voice device handle passed as either devHandle1 or devHandle2 is treated as an analog front-end.

For example, DM_ANALOG_INTF needs to be ORed with either DM_FULLDUP or DM_HALFDUP for the application to consider the analog voice device passed to **dev_Connect( )** as an analog front-end. If DM_ANALOG_INTF is not ORed, then the analog voice device will be considered a voice device.

Note: The **dev_Disconnect( )** function must be called to break a connection made using the **dev_Connect( )** function before a new connection can be established.

The following connections are supported:

Analog Front-End and CNF Audio Conference Party

A full-duplex or half-duplex connection between an audio conferencing party device (CNF API) and a Springware front-end device on board. Requires a valid audio conferencing party device handle obtained through the **cnf_OpenParty( )** function and a valid voice device handle obtained through the **dx_open( )** function. Both synchronous and asynchronous modes are supported. In the half-duplex connection, either type of device can listen to the other.

Springware Voice and CNF Audio Conferencing Party

A full-duplex or half-duplex connection between an audio conferencing party device (CNF API) and a Springware voice device on board. Requires a valid audio conferencing party device handle obtained through the **cnf_OpenParty( )** function and a valid voice device handle obtained through the **dx_open( )** function. Both synchronous and asynchronous modes are supported. In the half-duplex connection, either type of device can listen to the other.

Analog Front-End and HMP Voice Device

A full-duplex or half-duplex connection between a HMP voice device (Voice API) and an analog front-end device on board. Requires a valid HMP voice device handle obtained through the **dx_open( )** function and a valid analog voice device handle obtained through the **dx_open( )** function. Both synchronous and asynchronous modes are supported. In the half-duplex connection, either type of device can listen to the other.

Notes:1. Do not use the **dev_Connect( )** function to connect two on-board devices. One of the device handles passed to the function must me an HMP device. If two analog device handles are passed to **dev_Connect( )**, the function will return an error.

2. The **dev_Connect( )** function does not support analog device connections to or from any HMP device other than a CNF device.

3.8.2 D4PCIU Board Restrictions

The D4PCIU board does not have an onboard CT Bus. The board is capable of routing host-based HMP resources to and from the analog front-end interface using the HMP soft Bus. Each analog front-end interface establishes a permanent bridge connect to a unique HMP soft Bus in the board-to-HMP soft Bus direction as returned by the **ag_getxmitslot( )** function on the associated voice device. In the HMP soft Bus-to-board direction, an **ag_listen( )** function invoked on the associated voice device is responsible for routing from an HMP soft Bus time slot via a bridge connect. The routing can be removed by a disconnect resulting from the **ag_unlisten( )** function.

On the D4PCIU board, the **ag_unlisten( )** function has a dual-purpose functionality. Before it can create a bridge from HMP resources, it must first break the transmission of audio from the analog front-end's associated voice device. This is because an analog channel can either receive audio from the HMP soft Bus bridge or the associated voice channel. Conversely, the **ag_unlisten( )** function not only breaks a previously existing bridge from HMP resources, but it also switches back on-board audio transmission from the analog front-end's associated voice device. Therefore, before a new analog call can be processed by the analog front-end interface, the **ag_unlisten( )** function must be called on the associated voice device to be able to transmit digits (required by Global Call).

The following restrictions apply to D4PCIU boards:

- The boards will not function with any other board in the system, and must be the only board in the system. This is a permanent, stand-alone restriction.
- Transaction record is not supported.
- The Voice API library routing functions **dx_getxmitslot( )**, **dx_listen( )**, and **dx_unlisten( )** are not supported. Calls to these functions will return the error, EDX_BADPROD.
- When using the Device Management API **dev_Connect( )** function for bridging between HMP and the board's analog interfaces, be sure DM_ANALOG_INTF is added to the **dev_Connect( )** function's connType parameter. If this value is not added, the function will fail because bridging the on-board voice devices is not supported on these boards.

3.8.3 Important Considerations

CT Bus TDM routing for on-board voice and analog interfaces is now extended to and from HMP devices in order to provide media streaming.

Transaction Record

Transaction record enables the recording of a two-party conversation by allowing data from two time division multiplexing (TDM) bus time slots to be recorded with a single channel using the voice API **dx_mreciottdata( )** function.

Note: Transaction record is not supported on the Dialogic® D4PCIU boards.

With HMP and the Dialogic® D/80PCIE-LS board, TDM bus time slots can be of two types:

– CT Bus: Transmit TDM time slots from HMP-enabled analog or digital boards

– HMP soft Bus: Transmit TDM time slots from HMP media or IP devices

Transaction record remains a valid feature of the Dialogic® D/80PCIE-LS board whenever voice data belongs to CT Bus TDM bus time slots; however, it is not functional when voice data from either of the two receive TDM time slots is from HMP soft Bus TDM bus time slots.

In other words, if the valid channel device handle in the **dx_mreciottdata( )** function is from a Dialogic® D/80PCIE-LS board, the SC_TSINFO sc_tsarrayp values must contain two valid CT Bus TDM time slots from transmit TDM time slots from analog or digital network interface HMP-enabled boards. The function will fail when either of the SC_TSINFO sc_tsarrayp values belong to transmit TDM bus time slots from HMP host devices.

Whenever transaction recording from an HMP soft Bus TDM time slot, i.e., from an HMP media or IP device, is required, use an HMP voice device instead. In this case, there is no limitation and both time slots are supported. The following table summarizes:

Voice Device Handle Technology	TDM Time Slot One sc_tsarrayp[0]	TDM Time Slot One sc_tsarrayp[1]	Result
HMP	HMP soft Bus	CT Bus	Supported
	CT Bus	HMP soft Bus	Supported
	CT Bus	CT Bus	Supported
	HMP soft Bus	HMP soft Bus	Supported
D/80PCIE-LS	HMP soft Bus	CT Bus	Not Supported
	CT Bus	HMP soft Bus	Not Supported
	CT Bus	CT Bus	Supported
	HMP soft Bus	HMP soft Bus	Not Supported

Echo Cancellation

Several parameter updates are recommended when using the Dialogic® Springware boards in gateway mode (bridging to and from IP devices) to reduce echo that may be originated in the board's analog interfaces.

First, echo cancellation should be enabled on the HMP IP devices globally and statically by updating the HMP *.config* file as shown below. Refer to the *Dialogic® IP Media Library API Programming Guide and Library Reference* for more information.

```
[IPVSC]
SetParm=0x1b12, 1  !Enable EC on HMP.
SetParm=0x1b13, 512  !Set EC tail length to 128(16ms), 512(64ms)
```

Once echo cancellation is enabled, the system has to be reconfigured. Follow the steps below:

1. Run the following command to generate the *.fcd* file:
   ```
   fcdgen <license filename>.config -o  <license filename>.fcd
   ```
2. Re-initialized the HMP system using DCM as explained in the product Configuration Guide.
3. Once complete, the application needs to control echo cancellation on the IP devices at run time. This is accomplished using the **ipm_SetParm( )** function on an IP media channel basis as shown in the code excerpt below. In this case, echo cancellation is enabled using the corresponding device handle (nDeviceHandle) of an IP media channel bridged in full-duplex with an analog device in gateway mode.
   ```
   ipm_SetParm(nDeviceHandle, PARMCH_ECACTIVE,1)         //Enables EC
   for this channel

   ipm_SetParm(nDeviceHandle, PARMCH_ECNLP_ACTIVE,1)    // Enables
   Non-Linear processing for this channel
   ```

   It is also recommended that the application reduce the volume of receive audio through the IP device. This can be accomplished at run-time, on an IP media channel basis, as shown below. In this excerpt, there is a -7 dB reduction from the default.

```
ipm_SetParm(nDeviceHandle, PARMCH_RX_ADJVOLUME,-7)      // Reduces
the volume of Inbound IP traffic before it goes into the analog
board by 7dB
```

Refer to the *Dialogic® IP Media Library API Programming Guide and Library Reference* for more information about using the **ipm_SetParm( )** function and parameters.

Alternatively for 1PCC, Global Call has a convenient method for setting the IP Media Library parameters using the IPSET_CONFIG SetID and IPPARM_IPMPARM ParmID. Please refer to Section 4.30 “Setting Dialogic® IP Media Library Parameters” in the *Dialogic® Global Call IP Technology Guide* for details.

3.8.4 Supported Coders

The supported analog devices provide fully functional on-board voice devices that support voice encoding methods and sampling rates. Refer to the *Dialogic® Voice API Programming Guide* for the voice encoding methods for these Dialogic® Springware devices.

3.8.5 Configuration Considerations

This section provides configuration considerations applicable when using the Dialogic® Springware boards.

Firmware Load File

The firmware files available for the Dialogic® D/4PCIU boards and the Dialogic® D/80PCIE-LS board are as follows:

D/4PCIUFEQ
: d4pciu.fwl   Provides four channels of basic voice processing and fax.

D/4PCIU4SEQ
: d4ucsp.fwl   Provides four channels of basic voice processing, and continuous speech processing (CSP).

D/80PCIE-LS
: D8xjct.fwl   Provides eight channels of basic voice processing and fax. (Default)
: D81jcsp.fwl   Provides eight channels of basic voice processing, fax, and continuous speech processing (CSP).

Within the Dialogic Configuration Manager (DCM), each board has a set of property sheets that display a set of board’s configuration parameters. Each property sheet displays a different set of parameters based on the functionality they affect. To access a board’s property sheets, double-click on the board model name in the system window. The Misc property sheet is displayed by default. The Misc property sheet, contains the

FirmwareFile parameter. This is where a non-default parameter for the firmware file would be selected.

See the *Dialogic® Springware Architecture Products on Windows Configuration Guide* for more information about configuration procedures and firmware load files.

HMP Resource Licensing

The same procedures for associating an HMP resource license to boards should be followed for the Dialogic® Springware boards. The *Dialogic® HMP Software Release 3.0WIN Administration Guide* documents the steps required for HMP resource license configuration and activation. If the analog board is the only board in the system, then its input data must be used for locking HMP resources to it, otherwise its selection is optional. Please see the "Obtaining a License File that is Locked to a Board" section of the Administration Guide for more instructions.

Analog Line Adaptation Utility (LineAdapt)

The Dialogic® D/80PCIE-LS board (North American version) supports the analog line adaptation utility (LineAdapt). This configuration utility is available for tuning the impedance level on analog front-ends to reduce transmitter side line echo due to degraded analog telephone lines that deviate from their designed impedance range. Information about using the LineAdapt utility can be found in the *Dialogic® Springware Architecture Products on Windows Configuration Guide*.

Note: The Line Adapt Utility is not supported when using the Dialogic® D4PCIU boards.

3.9 Virtualization Support

This feature specifically focuses on the VMware® ESXi 4.0 installable product which provides a native (or full) virtualization layer running on physical servers for abstracting processor, memory, storage, and resources into multiple virtual machines. For more information about virtualization, refer to the VMware web site at *www.vmware.com*.

Notes:

1. It is assumed that the reader is familiar with common terms used to describe basic virtualization concepts, such as guest operating system, host, hypervisor, etc.
2. Virtualization is not supported on thin-blade configurations.

3.9.1 VMware® ESXi 4.0 Virtualization Support

Dialogic® HMP virtualization refers to the capability of running a separate instance of the Dialogic® HMP software release on the "guest" operating system of one or more virtual machines being hosted on the same physical platform (i.e., server). Each Dialogic® HMP software release has a separate runtime license, a number of dedicated resources, and requires a dedicated application (written to standard Dialogic® HMP Global Call and R4 Media API) to manage the resources.

HMP virtualization is implemented using VMware® ESXi 4.0 Update 1 Installable. VMware® ESXi partitions a physical server into multiple secure and portable virtual machines that can run side by side. Each virtual machine represents a complete system—with processors, memory, networking, storage and BIOS—so that an operating system and software applications can be installed and run in the virtual machine without any modification.

Refer to the VMware® ESXi 4.0 documentation at *http://www.vmware.com/support/pubs/* for more information.

The density achieved when operating in an virtual environment is directly dependent on the configuration settings of the virtual machine (i.e., CPU, memory, etc.) and the host platform hardware. Users should view the configuration settings provided as guidelines and not absolute, based on the target platform hardware characteristics in which feature validation was performed. Customizing the settings for optimal performance based on needs of the controlling application and host platform should be done by knowledgeable and experience personnel familiar with VMware® ESXi products.

3.9.2 Configuring HMP Virtualization

To configure Dialogic® HMP software to run as close as it would in a physical server configuration, the hypervisor should be configured to distribute the host hardware CPU processor, memory, storage, and networking resources to enable the real-time processing of RTP, media, and call control on all instances of the Dialogic® HMP software. The following subsections examine the critical parameters to achieve this goal. Please refer to the vSphere Resource Management Guide found at *http://www.vmware.com/pdf/vsphere4/r40_u1/vsp_40_u1_resource_mgmt.pdf* for a thorough explanation of the terms and concepts utilized herein.

CPU Affinity Settings

To run real-time software on VMware® ESXi, use CPU affinity. This is the recommended method for real-time voice since each virtual processor can get CPU resources directly from one or more of the available host CPUs, reducing the likelihood that virtual processors are rescheduled to give CPU time to another virtual machine.

Each virtual machine is more isolated, which helps real-time software run as though it were in a physical server environment. Due to HMP software’s intensive use of the operating system kernel resources, it is also highly recommended to set aside one physical (host) CPU to the VMware® ESXi 4.0 hypervisor. This host CPU should not be part of the affinity setting of any of the virtual machines.

For example, on a dual-processor, four-core host system without hyper-threading system, there will be eight physical CPUs available to VMware® ESXi. In this scenario, two virtual machines are configured with two virtual processors each. The system administrator could set the first virtual machine CPU affinity to physical CPUs 0 through 3 (total 4), and the second virtual machine CPU with affinity to physical CPUs 4 through 6 (total 3); this leaves physical CPU 7 unassigned and available to the VMware® ESXi hypervisor.

Virtual machine configuration is accomplished using the vSphere vCenter or via the VMware CLI. Refer to the vSphere Basic System Administration or equivalent guide at *http://www.vmware.com/pdf/vsphere4/r40_u1/vsp_40_u1_admin_guide.pdf* for vSphere vCenter information. For VMware CLI instructions, refer to *http://www.vmware.com/pdf/vsphere4/r40_u1/vsp_40_u1_vcli.pdf*.

Notes: ***1.*** Be careful not to cross physical processor boundaries when assigning CPU affinity to virtual machines, so that all host CPUs assigned to a virtual machine belong to the same host physical processor.

2. On NUMA host servers, it is recommended to keep all physical CPUs affine to a virtual machine residing in the same NUMA node in order to avoid a performance penalty when accessing non-local memory.

Timing Configuration

For optimal virtual machine timing and HMP operation in a virtualized environment, it is recommended that VMware Tools are installed in each virtual machine.

- Install VMware Tools in each virtual machine. Refer to the VMware ESXi Setup Guide for the installation procedure.
- Use the vShpere vCenter utility (or VMware CLI) to access the host system Time Configuration. Provide the address of an appropriate NTP Server in the Date and Time Options, and restart the NTP service to apply the changes.

Note: VMware Tools includes an optional clock synchronization feature "Time Synchronization between the virtual machine and the ESX Server" that can be enabled in the virtual machines, and could conflict with the native synchronization software. Be aware that having both enabled could affect the virtual machine's operating system's ability to correct long-term wall-clock drift, hence affect HMP audio quality.

Resource Budgeting

The same HMP requirements for system resources are required when operating in a VMware® ESXi environment. Refer to the Dialogic® Host Media Processing Software Release 3.0WIN Release Guide for those requirements.

The user is responsible for distributing the host system so enough resources are available to the virtual machines at all times. In addition to the CPU affinity and timing settings discussed, VMware® ESXi and vSphere provide a vast number of virtual machine configuration parameters that affect the configuration and behavior of virtual resources, such as reservation, shares, and resource pools that are outside of the scope of this document but are very important in providing a virtual environment to HMP as close as possible to a physical server environment.

Network Configuration

By default, VMware® ESXi provides one virtual switch that handles all virtual machine network traffic according to each virtual machine's IP and MAC addresses and VMware® ESXi management network traffic. Virtual machines can be assigned to virtual networks, and these to virtual switches in various network topologies, utilizing all available host physical network interfaces. The system integrator should carefully consider the virtual network layout based on the aggregated network traffic of all virtual machines and the capabilities and number of the physical network interfaces.

3.9.3 Density Limits

Aggregate density limits were tested at the currently supported limits as physical platforms. It is important to note that density projections are platform specific and are susceptible to the performance capabilities of the underlying hardware platform (host), and to the number of virtual machines. Initial density results show that the aggregate density of virtual machines running on the same host may be slightly less than the total capacity of the physical server. This is the result of additional overhead associated with each virtual machine.

3.10 Other Features

In addition to the new features, the following features are also supported in this release:

- Enhanced Licensing Capability

  Support for the following licensing enhancements:

  – Host based licensing - license is associated with particular machine based on the machine's MAC address

  – Board-based licensing - license is associated with a particular board based on the board's serial number

  For additional information, see the *Dialogic® Host Media Processing Software Release 3.0WIN Administration Guide*.

- Host Streaming Interface

  Dialogic® HMP Software Release 3.0WIN can create media stream connections between external board-based channels on the Dialogic® Digital Network Interface (DNI) board to the HMP host.

  The Host Streaming Interface feature allows Dialogic® DNI boards to communicate with HMP software running on the host. The Dialogic® DNI boards contain bridge devices that are used to perform this streaming.

  For additional information, see the *Dialogic® Host Media Processing Configuration Guide*.

- Interface Board Support

  Dialogic® HMP Software Release 3.0WIN supports the following interface boards:

  – Dialogic® Digital Network Interface Boards

  For Dialogic® Network Interface Boards, the Dialogic® Global Call API is used to provide 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 more information about the interface boards, see the *Dialogic® Host Media Processing Configuration Guide* and the Dialogic® Configuration Manager (DCM) Online Help.

- Echo Cancellation

  Includes the following echo cancellation support for circuit switched connections that may contain echo on the received media stream:
  - Provides echo cancellation via runtime API command for the:
    - Dialogic® HMP Software signal detector of the Dialogic® HMP Voice resource
    - Dialogic® HMP Software conferencing resource
    - Dialogic® HMP Software CSP resource
    - Dialogic® HMP Software IPM resource (for E1/T1 connections)
  - Compliance with G.168
  - Support for echo cancellation on tail lengths up to 64 msec

  Note: The signal detector only supports tail lengths of 8 msec.

  See the appropriate API library documentation for more information about echo cancellation support in that API library. See the *Dialogic® Host Media Processing Software Configuration Guide* for information about implementing echo cancellation.

- SIP Re-INVITE

  Support for subsequent INVITE requests, also known as re-INVITE requests, on existing SIP dialogs (calls). A Global Call application using the SIP protocol can originate a re-INVITE request to a remote endpoint, receive a re-INVITE request from the remote endpoint, and accept or reject that received re-INVITE.

  The support of re-INVITE is implemented by means of SIP-specific APIs in the Global Call library. These APIs are documented in detail in the *Dialogic® Global Call IP Technology Guide*.

- Early Media Call Setup

  When using IP technology, the ability to configure an end point for half-duplex (or full-duplex) media streaming and subsequently reconfigure the end point for full-duplex (or half-duplex) media streaming is supported. See the *Dialogic® IP Media Library API Programming Guide* and the *Dialogic® Global Call IP Technology Guide* for additional information.

- Supported Codecs for IP (RTP) Encoding/Decoding:
  - G.711 (64 kbps format) mu-law and A-law (10, 20, and 30 ms frames)

  Note: Frames of 10 ms are not supported on configurations that exceed 240 channels.

  - G.723.1 (5.3 and 6.3 kbps format) 30 ms frames (1, 2, or 3 frames per packet)
  - G.729A (compatible with G.729 format) and G.729AB (compatible with G.729B format) (8 kbps format) 10 ms frames (2, 3, or 4 frames per packet)
  - G.726 IP Coder

- IP Call Control:
  - Support for H.323 and SIP protocols via Global Call
  - SIP call transfer
  - H.450.2 call transfer (H.323)
  - SIP outbound proxy
  - SIP over TCP
  - SIP request retry
  - MIME-encoded SIP message bodies

- SIP INFO messages
- SIP OPTION messages
- SIP SUBSCRIBE and NOTIFY messages
- Getting RTP addresses of a call
- Getting SIP-specific origination and destination addresses for a call
- Host LAN cable disconnect alarms
- SIP generic headers
- SIP register and unregister
- SIP digest authentication
- Support for SIP message headers greater than 255 bytes

(See the *Dialogic® Global Call IP Technology Guide* for information.)

- APIs:
  - IP Media library to support third party call control (3PCC) mode and third-party protocol stacks for call control over IP (See the *Dialogic® IP Media Library API Programming Guide and Library Reference*.)
  - R4 Media Processing for Voice (See the *Dialogic®* V*oice API Library Reference* and the *Dialogic® Voice API Programming Guide* for information.)
  - R4 Media Processing for Conferencing (See the *Dialogic® Audio Conferencing API Library Reference* and *Dialogic® Audio Conferencing API Programming Guide* for information about DCB conferencing. See the *Dialogic® Conferencing API Library Reference* and *Dialogic® Conferencing API Programming Guide* for information about CNF conferencing.)
  - R4 Media Processing for Fax (See the *Dialogic® Fax Software Reference* for information.)
  - R4 Media Processing for Continuous Speech Processing (See the *Dialogic® Continuous Speech Processing API Library Reference* and *Dialogic® Continuous Speech Processing API Programming Guide* for information.)
  - Global Call for call control (See the *Dialogic® Global Call API Library Reference* and *Dialogic® Global Call API Programming Guide* for information.)
  - Standard Runtime Library for event handling (See the *Dialogic® Standard Runtime Library API Library Reference* and *Dialogic® Standard Runtime Library API Programming Guide* for information.)
  - Device Management API for coder reservation and T.38 connection (See the *Dialogic® Device Management API Library Reference* for more information.)
- IP Multicast (transmit and receive) support
- Tone Management:
  - In-Band DTMF detection/generation
  - RFC 2833 DTMF detection/generation
  - H.245 User Input Indication (UII) (H.323) reception/transmission
  - User-defined Global Tone Detection (GTD) and Global Tone Generation (GTG)
- Player/Recorder Formats:
  - G.711 mu-law and A-law (48 kbps and 64 kbps)
  - OKI ADPCM (24 kbps and 32 kbps)
  - Linear PCM (88 kbps and 128 kbps)

  - G.726 (16 kbps and 32 kbps)
- Play/Record Capability:
  - Playing and recording files in all supported encoding formats with or without Wave headers
  - Automatic Gain Control
  - Volume Control
  - Indexed Play
  - Streaming to Board (streams data to the network interface in real time)
- Call progress analysis:
  - Cadence detection
  - Frequency detection
  - Loop current detection
  - Positive voice detection
  - Positive answering machine detection
  - Fax tone detection
- Audio conferencing:
  - Active Talker status
  - Digit Detection with tone clamping
  - Monitoring
  - Coach/Pupil monitoring
- Speech Integration (Continuous Speech Processing)
- T.38 Fax origination/termination
- V.17 Fax origination/termination
- Transaction Record - enables the recording of a two-party conversation by allowing two time slots from a single channel to be recorded. (See the *Dialogic® Voice API Library Reference* and *Dialogic® Voice API Programming Guide* for information about this feature.)
- Programmatic control of inbound RTP stream gain and outbound RTP stream volume (See the *Dialogic® IP Media Library API Programming Guide and Library Reference* for more information.)
- Support of event notification for RTP and RTCP traffic stopping and starting (See the *Dialogic® IP Media Library API Programming Guide and Library Reference* for information about this feature.)
- More flexible VoIP Quality of Service (QoS) support by modifying the default Registry setting of the Type Of Service (TOS) byte during installation to support TOS setting through IPML APIs at run time. (See the *Dialogic® IP Media Library API Programming Guide and Library Reference* for information about this feature.)
- Manual, Semi-automatic, and Automatic startup modes for starting the System Service (See the *Dialogic® Host Media Processing Software Release 3.0WIN Software Installation Guide* for information about this feature.)
- Increased usability of Dialogic® HMP Software on Windows® platforms with Advanced Configuration and Power Interface (ACPI) by integrating automated Advanced Programmable Interrupt Controller (APIC) Timer compatibility check into the installation. (See the *Dialogic® Host Media Processing Software Release 3.0WIN*

Software Installation Guide and the Compatibility Notes section of the Release Update for information about this feature.)

Note: Processor throttling can be left on with Windows® 7 or Windows Server® 2008 as these operating systems do not use the APIC Driver.

- Support for Development Environment Microsoft® Visual Studio 2005 with Visual Studio 2005 Service Pack 1 and Visual Studio 2005 Service Pack1 Update for Windows® Vista for development on Windows® 7 or Windows Server® 2008
- Ability to configure UDP/RTP port range

Installation and Configuration 4

This chapter describes the installation and configuration software features that are supported by Dialogic® Host Media Processing Software Release 3.0WIN.

- Installation. 43
- Configuration 44

4.1 Installation

The installation of Dialogic® Host Media Processing Software Release 3.0WIN is a complete installation.

If Dialogic® Host Media Processing (HMP) Software Release 1.1, 1.3 or 2.0WIN is installed on your system, you must uninstall this version before installing Dialogic® HMP Software Release 3.0WIN. However, if Dialogic® HMP Software Release 3.0WIN is on your system, you can upgrade to a later Dialogic® HMP Software Release 3.0WIN Service Update without uninstalling the existing version. During installation, the new software will determine if there is a previous version of Dialogic® HMP Software Release 3.0WIN installed, or that no Dialogic® HMP Software is currently installed, and correctly install the appropriate Dialogic® HMP 3.0 Software.

Notes:

1. If upgrading the Windows® Operating System from a version prior to Windows® 7 and Windows Server® 2008, the Dialogic® HMP Software must be uninstalled prior to upgrade.
2. Dialogic® HMP Software with Windows® 7 or Windows Server® 2008 requires a motherboard with an HPET Timer. Run the HPETTOOL Utility to determine motherboard compatibility. For the most desirable results, select systems that have BIOS support for the HPET Timer and declare the HPET Timer in the ACPI table. Refer to the *Dialogic® HMP Software Release 3.0WIN Software Installation Guide* for instructions about using the HPETTOOL Utility.
3. You may see a **Security Alert- Driver Installation** message during installation of the Dialogic® HMP Software. This is just a warning message, and the Dialogic® HMP Software will install properly after you click **Yes**. For more information, refer to the *Dialogic® Host Media Processing Software Release 3.0WIN Software Installation Guide*.

Web Only User Documentation

With Dialogic® Host Media Processing Software Release 3.0WIN, the user documentation will be delivered only on the web. This is to ensure that only the most current user documentation is available. Because the user documentation is available on the web, "Documentation" will no longer be available as an installable option as it has been in the past. When you install the Dialogic® HMP Software Release 3.0WIN, a shortcut to the web site containing the user documentation will be placed on the Start menu as follows:

Start > Programs > Dialogic HMP > Documentation > User Documentation.

User documentation that was installed with a previous release will be removed.

For complete installation information, see the *Dialogic® Host Media Processing Software Release 3.0WIN Software Installation Guide*.

4.2 Configuration

This section describes the configuration software capabilities that are supported in the Dialogic® Host Media Processing Software Release 3.0WIN. Configuration is performed after the system software is installed, using the Dialogic® Configuration Manager (DCM) utility. The Dialogic® DCM utility allows you to configure:

- Dialogic® HMP Software
- Bridge Devices
- Dialogic® Digital Network Interface boards
- TDM bus

In addition, you may modify FCD file parameters associated with a Dialogic® Digital Network Interface board by editing the board's *.config* file.

For detailed configuration information, see the DCM Online Help and the *Dialogic® Host Media Processing Software Configuration Guide*.

Trunk Configuration

The trunks on a Dialogic® Digital Network Interface board can be configured for both the media load and, on a trunk-by-trunk basis, either a T1 or E1 protocol. Trunk configuration can be accomplished either through the Dialogic® Configuration Manager (DCM) utility using the Trunk Configuration Property sheet or programmatically using the Dialogic® NCM API function **NCM_ApplyTrunkConfiguration( )**.

For information about configuring the trunks using DCM, see the DCM Online Help. For information about configuring the trunks programmatically, refer to the *Dialogic® Native Configuration Manager API Library Reference* and the *Dialogic® Native Configuration Manager API Programming Guide*.

OA&M Software 5

This section describes the OA&M (operation, administration, and maintenance) software features that are supported in Dialogic® Host Media Processing Software Release 3.0WIN.

- Administration Software 45
- Dialogic® NCM (Native Configuration Manager) API Library 45
- Diagnostics Software 46

5.1 Administration Software

There have been no new administration capabilities or enhancements to existing capabilities since the Dialogic® Host Media Processing (HMP) Software Release 2.0WIN release. For more information about the administration software, see the *Dialogic® Host Media Processing Administration Guide*.

New Tools

For the Windows® 7 and Windows Server® 2008 operating systems, the HPETTOOL Utility is introduced to determine motherboard compatibility with the HPET Driver. This utility is installed with the Dialogic® HMP software and located in the C:\Program Files\Dialogic\HMP\bin directory. See the *Dialogic® HMP Software Release 3.0WIN Software Installation Guide* for more information about this utility.

New API Libraries

There have been no new API libraries or new features of existing API libraries since the Dialogic® Host Media Processing (HMP) Software Release 2.0WIN release.

5.2 Dialogic® NCM (Native Configuration Manager) API Library

The Dialogic® NCM API library provides an interface for developing customized system configuration and administration applications. The Dialogic® NCM API functions operate on the complete system, individual Dialogic® boards, or the TDM bus settings.

Refer to the *Dialogic® Native Configuration Manager API Library Reference* and the *Dialogic® Native Configuration Manager API Programming Guide* for complete information about the Dialogic® NCM API Development Software

New Features

There have been no new features since the Dialogic® Host Media Processing (HMP) Software Release 2.0WIN release.

5.3 Diagnostics Software

This section describes the new diagnostic capabilities and tools available for Dialogic® Host Media Processing Software Release 3.0WIN. For more information about the diagnostics software, refer to the *Dialogic® Host Media Processing Diagnostics Guide*.

Runtime Trace Facility (RTF) Tool

A new version of the Runtime Trace Facility tool is supported. The RTF tool provides a mechanism for tracing the execution path of runtime libraries that are supported by HMP 3.0. Support for the RTF tool is as follows:

- Rtftrace command is used to stop/start the RTF tool's tracing capabilities.
- Tool provides centralized logging for key OA&M components (OAMSYSLOG) and IP libraries.
- You can run the RTF tool in preservation mode. Preservation mode allows you to save specified RTF trace information into a separate, preserved log file while the RTF engine continues to output active trace information into the default log file. The RTF engine will not overwrite, delete or append to the preserved log file after it has been saved.

Programming Libraries 6

This chapter describes the various development libraries and demonstration programs that are available as part of Dialogic® Host Media Processing Software Release 3.0WIN.

- Dialogic® Audio Conferencing (DCB) API Library 47
- Dialogic® Conferencing API Library 48
- Dialogic® Continuous Speech Processing (CSP) API Library 49
- Dialogic® Device Management API Library 50
- Dialogic® Fax API Library. 50
- Dialogic® Global Call API Library 50
- Dialogic® IP Media Library API Library 51
- Dialogic® Learn Mode and Tone Set File APIs 52
- Dialogic® Multimedia API Library 53
- Dialogic® Standard Runtime API Library 53
- Dialogic® Voice API Library 54

6.1 Dialogic® Audio Conferencing (DCB) API Library

The Dialogic® Audio Conferencing (DCB) API library supports development of host-based conferencing applications. The Dialogic® Audio Conferencing API library provides many features that can be used to develop customized audio conferencing servers.

The Dialogic® Audio Conferencing software includes library functions, device drivers, and firmware.

Note: Dialogic® DCB Conferencing API support is not planned for subsequent releases. You should create all new conferencing applications using the Dialogic® CNF Conferencing API library.

Dialogic® Audio Conferencing API Functions Not Supported by Dialogic® HMP Software

The following Dialogic® Audio Conferencing API functions are not supported by HMP:

- **dcb_GetAtiBitsEx()**

Refer to the *Dialogic® Audio Conferencing API Programming Guide* and the *Dialogic® Audio Conferencing API Library Reference* for additional information.

New Features

There have been no new features implemented since Dialogic® Host Media Processing (HMP) Software Release 2.0WIN.

6.2 Dialogic® Conferencing API Library

The new Dialogic® CNF Conferencing API library supports development of conferencing applications on Dialogic® HMP Software. The conference can take place over an IP network and/or over traditional public switched telephone network (PSTN) lines.

Refer to the *Dialogic® Conferencing API Programming Guide* and the *Dialogic® Conferencing API Library Reference* for additional information.

Features

The new Dialogic® Conferencing API library supports the following features:

Asynchronous programming model support
: This model enables multiple channels to be handled in a single process and supports higher density conferencing solutions.

Support for conferees from multiple sources
: Participants in a conference may come from a variety of sources, such as a voice device and an IP media device. The software is designed for flexibility to grow and support additional sources.

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.

6.3 Dialogic® Continuous Speech Processing (CSP) API Library

The Dialogic® Continuous Speech Processing (CSP) API Library supports development of host-based automatic speech recognition (ASR) applications. CSP provides many features such as high-performance echo cancellation, voice energy detection, barge-in, voice event signaling, pre-speech buffering, and full-duplex operation.

The Dialogic® CSP software includes library functions, device drivers, firmware, and demonstration programs.

Dialogic® CSP API Functions Not Supported by Dialogic® HMP Software

The following Dialogic® CSP API functions are not supported by Dialogic® HMP Software Release 3.0WIN:

- **ec_rearm()**

Refer to the *Dialogic® Continuous Speech Processing API Programming Guide* and *Dialogic® Continuous Speech Processing API Library Reference* for more information.

New Features

There have been no new features implemented since Dialogic® Host Media Processing (HMP) Software Release 2.0WIN.

6.4 Dialogic® Device Management API Library

The Dialogic® Device Management API library provides run-time control and management of configurable system devices, which includes functions to reserve resources and manage the connections between devices for communication and sharing of resources. The Dialogic® Device Management API functions enable use of the T.38 fax IP-only resource, which provides the ability to originate and terminate T.38 fax over IP connections only. The API also includes functions to reserve low bit rate codecs for an IP media device on Dialogic® HMP Software.

Refer to the *Dialogic® Device Management API Library Reference* for more information.

New Features

No new device management API features have been added in Dialogic® Host Media Processing Software Release 3.0WIN.

6.5 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.

Refer to the *Dialogic® Fax Software Reference* for more information.

New Features

No new fax API features have been added in Dialogic® Host Media Processing Software Release 3.0WIN.

6.6 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 Dialogic® Global Call API library supports a variety of protocols.

The Dialogic® Global Call API library:

- Is designed to support H.323, SIP, and PSTN protocols
- Provides a consistent application interface for the various protocols and technologies

Dialogic® Global Call API Functions Not Supported by Dialogic® HMP Software

The following Dialogic® Global Call API functions are not supported by Dialogic® HMP Software:

- **gc_Attach()**
- **gc_CallProgress()**
- **gc_GetANI()**
- **gc_GetBilling()**
- **gc_GetCallProgressParm()**
- **gc_GetConfigData()**
- **gc_GetDNIS()**
- **gc_GetInfoElem()**
- **gc_GetNetworkH()**
- **gc_GetUserInfo()**
- **gc_GetVoiceH()**
- **gc_LoadDxParm()**
- **gc_Open()**
- **gc_QueryConfigData()**
- **gc_ReleaseCall()**
- **gc_ReqANI()**
- **gc_SetBilling()**
- **gc_SetCallProgressParm()**

The generic functionality of Global Call is documented in the *Dialogic® Global Call API Library Reference* and the *Dialogic® Global Call API Programming Guide*. HMP-specific functionality is documented in the *Dialogic® Global Call IP Technology Guide*.

New Features

There have been no new features implemented since Dialogic® Host Media Processing (HMP) Software Release 2.0WIN.

6.7 Dialogic® IP Media Library API Library

The Dialogic® IP Media Library API 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.

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

New Features

The Dialogic® IP Media Library API supports the following new features in this release. These features are new since Dialogic® Host Media Processing (HMP) Software Release 2.0WIN:

Secure RTP

The payload (and padding) of RTP packets may be encrypted using SRTP. The IP Media Library now includes the **ipm_SecurityGenMasterKeys()** function that generates master and salt keys.

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

Video Support

The IP media library supports start/stop multimedia sessions and get/set video related properties.

6.8 Dialogic® Learn Mode and Tone Set File APIs

The Learn Mode API is supported on Dialogic® Springware boards only.

Note: For more information about the Learn Mode and Tone Set File APIs, refer to the *Dialogic® Learn Mode and Tone Set File API Software Reference*.

Features

The Learn Mode API provides the ability to characterize a call progress tone from a PBX, key system or PSTN and to obtain a complete tone description.

Note: You can use the PBX Expert utility provided with this system release to accomplish PBX tone learning and tone set file management. See the online help provided with this utility for more information.

The Tone Set File API provides the capability to do the following:

- Store unique call progress tone data obtained from the Learn Mode API
- Change default tone definitions that are provided by the voice library
- Create a new tone definition and add it to the tone template for use with call progress analysis
- Store an unlimited number of tone sets on your system (subject to storage constraints)
- Combine up to 10 sets of tone characteristics into a single, consolidated tone set
- Support up to 10 sets of tone characteristics (that is, for up to 10 different PBX or key systems)

6.9 Dialogic® Multimedia API Library

The Dialogic® Multimedia API is 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.

Multimedia library functionality is documented in the *Dialogic® Multimedia API Programming Guide and Library Reference*.

Features

The Dialogic® Multimedia API library provides the following capabilities:

Real-time recording
: 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. Optionally, you can 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.

Real-time playback
: 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.

Play Voice API audio files in multimedia session
: Play 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).

See *Multimedia File Conversion Utilities* in Section 3.1, "Multimedia (Audio/Video)", on page 23.

6.10 Dialogic® Standard Runtime API Library

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

Dialogic® SRL API Functions Not Supported by Dialogic® HMP Software

The following Dialogic® SRL API functions are not supported by Dialogic® HMP Software:

- **sr_getboardcnt()**

Refer to the *Dialogic® Standard Runtime Library API Programming Guide* and the *Dialogic® Standard Runtime Library API Library Reference* for more information.

New Features

No new Dialogic® SRL API features have been added in Dialogic® Host Media Processing Software Release 3.0WIN.

6.11 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.

Refer to the *Dialogic® Voice API Library Reference* and the *Dialogic® Voice API Programming Guide* for more information.

New Features

The Dialogic® Voice API library supports the following new features in this release. These features are new since Dialogic® Host Media Processing (HMP) Software Release 2.0WIN:

Support for speed control

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

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

Note: Before using the speed control feature, you must enable this feature in the [decoder] section of the CONFIG file. The speed control feature is disabled by default to preserve MIPS usage and enhance system performance. For more information about enabling speed control, see the *Dialogic® Host Media Processing Software Configuration Guide*.

Supported Applications 7

This chapter describes applications supported by Dialogic® Host Media Processing Software Release 3.0WIN.

- MSML Media Server Software 55

7.1 MSML Media Server Software

The MSML Media Server Software has been designed and implemented as an integral part of the Dialogic® HMP Software.

When the system software 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 and MOML languages, as defined in their respective IETF drafts. The current implementation is based on Media Sessions Markup Language (MSML) IETF Draft version -06 and Media Objects Markup Language (MOML) IETF Draft version -06.

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

Features

The MSML Media Server Software features are being introduced in a phased approach. The supported features map to corresponding IETF drafts as follows:

- MSML Core Module
- MSML Stream Management Module (conferencing not included)
- MSML Dialog Module
- MOML Core Module
- MOML Group Module (parallel topology only)
- MOML Basic Primitives Module
- MOML Transform Primitives Module (gain only)

Functionality that is not supported by the current implementation includes:

- MSML Stream Management Module (audio and video conferencing)
- MOML Speech Module
- MOML Fax Module

Supported Hardware 8

This section lists the boards supported by Dialogic® Host Media Processing Software Release 3.0WIN. The boards are organized into the following categories:

- Dialogic® Digital Network Interface Products 57
- Dialogic® Signaling Products 57
- Dialogic® Analog Interface Products 58

8.1 Dialogic® Digital Network Interface Products

The Dialogic® digital network interface products consist of the following models:

- DNI300TEPHMP
- DNI601TEPHMP
- DNI1200TEPHMP

8.2 Dialogic® Signaling Products

The Dialogic® Signaling Products consist of the following subcategories:

- Dialogic® SS7 Boards
- Dialogic® Signaling Gateways (SS7 Signaling Interface Units [SIUs])

Dialogic® SS7 Boards

The Dialogic® SS7 boards consist of the following models:

- SPCI2S
- SPCI4
- SS7HDP

Dialogic® Signaling Gateways (SS7 Signaling Interface Units [SIUs])

The Dialogic® SS7 SIUs consist of the following models:

- SS7G21 (in SIU Mode)
- SS7G22 (in SIU Mode)

8.3 Dialogic® Analog Interface Products

The supported Dialogic® Analog Interface products consist of the following models:

- D/4PCIUFEQ
- D/4PCIU4SEQ
- D/80PCIE-LS

Demonstration Software 9

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

This chapter provides information about demonstration programs provided in Dialogic® Host Media Processing (HMP) Software Release 3.0WIN.

- New Demo Programs 59
- Other Supported Demo Programs 60

9.1 New Demo Programs

The following new demo programs are included in this release. These demo programs are new since Dialogic® Host Media Processing (HMP) Software Release 2.0WIN. The demo programs are located in the \demos directory under the environment variable for the directory in which the Dialogic® HMP Software was installed.

Multimedia Demo

The Multimedia demo application features the video capabilities of the Dialogic® Host Media Processing Software. The application is based on the Dialogic® Global Call API for Session Initiation Protocol (SIP) call control and uses the Dialogic® IP Media Library for Real-Time Transport Protocol (RTP) media manipulation, the Dialogic® Multimedia API for playing and recording audio/video streams, and Dialogic® Voice API for Dual-Tone Multi-Frequency (DTMF) detection and generation.

The Multimedia demo application demonstrates two use cases:

- Video mail - users can record an audio/video clip and play it back at a later time
- Video portal - users can select video clips from a menu

The key features of the Multimedia demo are:

- Terminates and answers an incoming SIP call
- Plays video clips (menus and/or short clips)
- Records an audio/video message to disk
- Detects inband and RFC 2833 DTMF digits in the RTP stream

See the *Dialogic® Multimedia Demo Guide* for more information.

MSML Demo

The Remote Media Control Client Sample Application provides a means to demonstrate and exercise features of an MSML Media Server as supported on Dialogic® HMP products. Remote control of a MSML Media Server by application servers is accomplished using the SIP-based Media Session Markup Language (MSML) and Media Object Markup Language (MOML) protocols. Additionally, the sample application demonstrates best known methods (BKM) for software architecture, design, and API usage.

The sample application server implements simple voice and video mail functionality using MSML scripts and media recording files to perform all media functions on the Media Server. The Application Server uses HMP in SIP 3PCC mode for call control.

See the readme.txt file provided with the Remote Media Control Application Server Sample Application for additional information about the MSML Demo.

CNF Conferencing Demo

The CNF Conferencing Demo is an audio conferencing application based on the Dialogic® CNF Conferencing API. The demo uses the Dialogic® Global Call API to handle call control, the Dialogic® Voice API to detect digits and play files, and the Dialogic® CNF Conferencing API to maintain conferences.

Once you have installed Dialogic® Host Media Processing (HMP) Software Release 3.0WIN, see:

Demos\conferencing\CnfConferencingDemo\doc\ConferenceDemo.doc

for detailed information about the CNF Conferencing Demo.

9.2 Other Supported Demo Programs

The following demo programs continue to be supported in this release:

Ansrmt Voice Demo

The Ansrmt demonstration program is a multithreaded application based on the Dialogic® Voice API library that is developed using the synchronous programming model. Unlike asynchronous application models that use events and event handlers to manage specific devices and events on an "interrupt" basis, Ansrmt uses a single thread per voice channel that allows voice channel processing to run uninterrupted until completion.

The Ansrmt demo program illustrates the voice recording and playback feature. You can listen to a prompt, record a message, and play back that message.

See the Ansrmt Voice Demo online Help for more information.

Audio Conferencing (DCB) Demo

The Audio Conferencing demo is a simple audio conferencing application that is implemented using the HMP software. The Dialogic® Audio Conferencing demo directly supports H.323 and SIP call control signaling protocols through use of the Dialogic® Global Call API.

The Audio Conferencing demo application is written in asynchronous mode, using a single process, single-threaded program that handles events using the polled mode. Conferencing features are accessed using the Audio Conferencing (DCB) API. The Dialogic® Global Call API is used for implementing call control and the Dialogic® Voice API is used for basic voice functionality.

See the *Dialogic® Audio Conferencing API Demo Guide* for more information.

Note: This Demo is not supported by the Dialogic® Interface boards.

Continuous Speech Processing (CSP) Demo

The CSP demo is a single-threaded program based on the Dialogic® Continuous Speech Processing API that illustrates CSP features such as voice activity detection, barge-in, pre-speech buffering, and echo cancellation. You can run the CSP demo in two different modes: manual mode and diagnostics mode.

In the Manual mode, the demo is a single-channel, interactive demo which allows you to barge in on a prompt that is being played. Messages are displayed on your screen as the demo progresses. This mode illustrates the operation of the **ec_stream( )** function.

In the Diagnostic mode, the demo is a non-interactive demo that exercises the CSP parameters.

See the *Dialogic® Continuous Speech Processing API Demo Guide* for more information.

Note: This Demo is not supported by the Dialogic® Interface boards.

Global Call API Demo

The Global Call API demo program sets up and tears down calls on the virtual boards and channels specified by the user. The program demonstrates call control functionality only and uses the Global Call basic call state model. Using the Global Call API demo program configuration file, the user can specify:

- The channels to be used by the demo
- The protocol (H.323 or SIP) to be used by each device
- The protocol type (inbound or outbound) for each device
- The IP destination address to associate with each device
- The transmit (Tx) and receive (Rx) codec parameters (type, rate and Voice Activity Detection [VAD])

When the Global Call API demo program is run, one device waits for calls while another device makes calls. The sequence of function calls, events received, and the call states are displayed as the program proceeds. When the user presses Ctrl-C to interrupt the process, the program prints a summary of the activity including information such as, the total number of inbound calls, the total number of outbound calls, the amount of time the demo program was running.

See the *Dialogic® Global Call API Demo Guide* for more information.

Note: This Demo is not supported by the Dialogic® Interface boards.

IP Gateway (Global Call) Demo

The IP Gateway (Global Call) demo program is an object-oriented host-based application that demonstrates using the Dialogic® Global Call API to build a PSTN-IP gateway. The demo source code can be used as sample code for those who want to begin developing an application from a working application. The demo is not designed to implement a complete gateway and lacks features such as least-cost routing.

The IP Gateway (Global Call) Object Oriented demo program supports the following features:

- Accepts IP calls
- Places IP calls
- Accepts PSTN calls
- Places PSTN calls

- Configuration file
- Command line options
- Output log files
- Printing to the monitor
- QoS

See the *Dialogic® IP Gateway (Global Call) Demo Guide* for more information.

IP Media Server for HMP Demo

The IP Media Server for HMP demo is an object-oriented host-based application that demonstrates using the Dialogic® Global Call API to build an IP media server, providing voice and fax services via IP technology. The demo source code can be used as sample code for those who want to begin developing an application from a working application.

The Dialogic® IP Media Server for HMP demo supports the following features:

- Voice service
- Fax service
- CSP barge in
- Configuration file
- Command line options

See the *Dialogic® IP Media Server Demo Guide* for more information.

Note: This Demo is not supported by the Dialogic® Interface boards.

Station Interface (SiTest) Demo

The SiTest demo program uses the Station Interface API to drive proprietary digital stations using the Dialogic® DSI162HMP or Dialogic® DSI162LGNHMP boards. The demo is a console-based application that requires two command line arguments – one that specifies the board (0-based) and one that specifies the station interface(1-based) to be used: For example, executing the command “SiTest 2 5” executes on the ssiB3C5 device. The default parameters that SiTest.exe will execute with are 0 and 1, namely the first station on the first board (ssiB1C1).

When the program is executed, it will query and output the capabilities (display, ringer, etc.) of the specified station, open the station, enable all asynchronous events, and subsequently wait for events on the station. Pressing keys on the digital phone, lifting the handset, disconnecting the phone, etc. will all generate unsolicited event messages to the console. The SiTest Demo program is terminated by pressing any key on the console keyboard.

Additionally, the SiTest demo provides a fully compiled Station Interface API project, complete with source code, project files, etc. in Visual C++® 6.0. This program can be accessed by opening the SiTest.dsw workspace in <Installation Path>\demos\SiTest

Note: To open the Station Interface project, you will need Visual C++® 6.0 installed on the machine.

Browsing to Project -> Settings will reveal the recommended compile and link options for multithreaded Station Interface API applications. The source code included in SiTest.cpp provides a simple starting point for developing applications using the Dialogic® Station Interface API.

VoiceDemo

The VoiceDemo demonstration program uses the asynchronous programming model. It is a multithreaded, multi-channel demo based on the Dialogic® Voice API that illustrates key voice features, such as play, record, volume adjustment, and make call.

The demo consists of two threads. The main thread creates a child window for each voice device that is opened. The purpose of the second thread is to poll for device events using the Standard Runtime Library function sr_waitevt( ). After an event is received, a message is displayed in the child window, and menu items are available for the next function selection.

See the VoiceDemo online Help for more information.

Xaansr Voice Demo

The Xaansr demonstration program is a multithreaded voice application based on the Voice API and is developed using the extended asynchronous polled programming model. Xaansr is state driven and uses two threads to handle events: one thread for voice events and another thread for network events.

The Xaansr demo program illustrates the voice recording and playback feature. You can listen to a prompt, record a message, and play back that message.

See the Xaansr Voice Demo online Help for more information.

Documentation 10

This chapter provides information about the documentation that supports Dialogic® Host Media Processing Software Release 3.0WIN. This information is organized into the following sections:

- Documentation Support for Dialogic HMP Software Release 3.0WIN Features65
- Release Documentation 68
- Installation and Configuration Documentation 69
- OA&M Documentation 69
- Programming Libraries Documentation 69
- Supported Applications Documentation 70
- Demonstration Software Documentation 70
- Online Help 71

10.1 Documentation Support for Dialogic HMP Software Release 3.0WIN Features

The following table lists the Dialogic® Host Media Processing Software Release 3.0WIN features and the user documentation containing the information about these features.

Table 5. User Documentation Feature Support

Dialogic® HMP Software Release 3.0WIN Feature	User Documentation
Dialogic® HMP Software Installation	• Dialogic® Host Media Processing Software Release 3.0WIN Installation Guide • Installation Online Help
Dialogic® HMP Software Configuration	• Dialogic® Host Media Processing Configuration Guide • Dialogic® Native Configuration Manager (NCM) API Library Reference • Dialogic® Native Configuration Manager (NCM) API Programming Guide • Dialogic® DCM Configuration Manager Online Help

Table 5. User Documentation Feature Support (Continued)

Dialogic® HMP Software Release 3.0WIN Feature	User Documentation
Resource Licensing	• Dialogic® Host Media Processing Software Release 3.0WIN Administration Guide • Dialogic® Host Media Processing Software License Transition Guide
IP Call Control Using Global Call • RFC2833 • H.245 UII • Low Bit Rate Coder Reservation	• Dialogic® Global Call API Library Reference • Dialogic® Global Call API Programming Guide • Dialogic® Global Call IP Technology Guide • Dialogic® IP Media Server Demo Guide
IP Call Transfer	• Dialogic® Global Call API Library Reference • Dialogic® Global Call IP Technology Guide
Digital Network Interface	• Dialogic® Digital Network Interface Software Reference • Dialogic® Global Call ISDN Technology Guide • Dialogic® Global Call E1/T1 CAS/R2 Technology Guide
SS7 Signaling	• Dialogic® Global Call API Library Reference • Dialogic® Global Call SS7 Technology Guide
Media Streaming Compatibility with a Third-Party Stack for IP Call Control • RFC2833 • H.245 UII • Low Bit Rate Coder Reservation	• Dialogic® IP Media Library API Programming Guide and Library Reference • Dialogic® Device Management API Library Reference
IP Multicast	• Dialogic® IP Media Library API Programming Guide and Library Reference

Table 5. User Documentation Feature Support (Continued)

Dialogic® HMP Software Release 3.0WIN Feature	User Documentation
Multimedia/Video	• Dialogic® Multimedia API Programming Guide and Library Reference • Dialogic® Multimedia Demo Guide • Dialogic® Device Management API Library Reference
Remote Media Processing	• Dialogic® MSML Media Server Software User's Guide • Dialogic® Global Call API Library Reference • Dialogic® Global Call API Programming Guide • Dialogic® Global Call IP Technology Guide
Voice Features	• Dialogic® Voice API Library Reference • Dialogic® Voice API Programming Guide
CNF Conferencing	• Dialogic® Conferencing API Library Reference • Dialogic® Conferencing API Programming Guide
DCB Conferencing	• Dialogic® Audio Conferencing API Library Reference • Dialogic® Audio Conferencing API Programming Guide • Dialogic® Audio Conferencing Demo Guide
Speech Integration	• Dialogic® Continuous Speech Processing API Library Reference • Dialogic® Continuous Speech Processing API Programming Guide • Dialogic® Continuous Speech Processing API Demo Guide
Fax Using Global Call API	• Dialogic® Global Call IP Technology Guide • Dialogic® Fax Software Reference

Table 5. User Documentation Feature Support (Continued)

Dialogic® HMP Software Release 3.0WIN Feature	User Documentation
Fax Using Third-Party Stack	• Dialogic® IP Media Library API Programming Guide and Library Reference • Dialogic® Device Management API Library Reference • Dialogic® Fax Software Reference
Analog Device (Springware) Support	• Dialogic® Springware Architecture Products on Windows Configuration Guide • Dialogic® Global Call Analog Technology Guide • Dialogic® Learn Mode and Tone Set File API Software Reference
Event Handling	• Dialogic® Standard Runtime Library API Library Reference • Dialogic® Standard Runtime Library API Programming Guide • Dialogic® Event Service API Library Reference • Dialogic® Event Service API Programming Guide
Diagnostics	• Dialogic® Host Media Processing Diagnostics Guide

10.2 Release Documentation

The following system documentation is provided for this release:

- *Dialogic® Host Media Processing Software Release 3.0WIN Release Guide* (this document). †
- *Dialogic® Host Media Processing Software Release 3.0WIN Release Update* †

  Note: The Release Update is not part of the online bookshelf, but is posted on the Support web site. This document includes issues that may affect the performance of the Dialogic® HMP Software and lists both resolved and known issues. The Release Update also includes corrections and changes to the user documentation that could not be made to the documents prior to the release.

Note: A dagger (†) next to a document title indicates that the document is new or has been updated since Dialogic® Host Media Processing (HMP) Software Release 2.0WIN.

10.3 Installation and Configuration Documentation

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

- *Dialogic® Host Media Processing Software Release 3.0WIN Software Installation Guide* †
- *Dialogic® Host Media Processing Configuration Guide* †
- *Dialogic® Springware Architecture Products on Windows Configuration Guide* †
- *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*†

Note: A dagger (†) next to a document title indicates that the document is new or has been updated since Dialogic® Host Media Processing (HMP) Software Release 2.0WIN.

10.4 OA&M Documentation

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

- *Dialogic® Host Media Processing Software Release 3.0WIN Administration Guide* †
- *Dialogic® SNMP Agent Software Administration Guide* †
- *Dialogic® Host Media Processing Diagnostics Guide* †
- *Dialogic® Event Service API Library Reference*
- *Dialogic® Event Service API Programming Guide*
- *Dialogic® Native Configuration Manager API Library Reference* †
- *Dialogic® Native Configuration Manager API Programming Guide* †

Note: A dagger (†) next to a document title indicates that the document is new or has been updated since Dialogic® Host Media Processing (HMP) Software Release 2.0WIN.

10.5 Programming Libraries Documentation

The following development software documentation is provided to support this release:

- *Dialogic® Audio Conferencing API Library Reference*
- *Dialogic® Audio Conferencing API Programming Guide*
- *Dialogic® Conferencing API Library Reference* †
- *Dialogic® Conferencing API Programming Guide* †
- *Dialogic® Continuous Speech Processing API Library Reference*
- *Dialogic® Continuous Speech Processing API Programming Guide*
- *Dialogic® Device Management API Library Reference* †
- *Dialogic® Digital Network Interface Software Reference*
- *Dialogic® Fax Software Reference*

- *Dialogic® Global Call Analog Technology Guide* †
- *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 Programming Guide and Library Reference* †
- *Dialogic® Learn Mode and Tone Set File API Software 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* †

Note: A dagger (†) next to a document title indicates that the document is new or has been updated since Dialogic® Host Media Processing (HMP) Software Release 2.0WIN.

10.6 Supported Applications Documentation

The following supported application documentation is provided to support this release:

- *MSML/MOML Media Server Software User's Guide* †

Note: A dagger (†) next to a document title indicates that the document is new or has been updated since Dialogic® Host Media Processing (HMP) Software Release 2.0WIN.

10.7 Demonstration Software Documentation

The following demo documentation is provided for this release:

- *Dialogic® Audio Conferencing API Demo Guide*
- *Dialogic® Continuous Speech Processing API Demo Guide*
- *Dialogic® Global Call API Demo Guide*
- *Dialogic® IP Media Server Demo Guide*
- *Dialogic® IP Gateway (Global Call) Demo Guide*
- *Dialogic® Multimedia Demo Guide* †

Note: A dagger (†) next to a document title indicates that the document is new or has been updated since Dialogic® Host Media Processing (HMP) Software Release 2.0WIN.

10.8 Online Help

The following online help is provided for this release:

- *Dialogic® Configuration Manager (DCM) Online Help* †
- *Dialogic® Installation Online Help* †
- *Dialogic® Ansrmt Voice Demo Online Help*
- *Dialogic® Xaansr Voice Demo Online Help*
- *Dialogic® VoiceDemo Online Help*

Note: A dagger (†) next to a document title indicates that the document is new or has been updated since Dialogic® Host Media Processing (HMP) Software Release 2.0WIN.