Dialogic® System Release 6.0 PCI for Windows®

Release Update

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About This Publication

This section contains information about the following topics:

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

Purpose

This Release Update addresses issues associated with Dialogic® System Release 6.0 PCI for Windows® (sometimes also referred to herein as “System Release 6.0 PCI Windows”). In addition to summarizing issues that were known as of the Release’s general availability, it is intended that this Release Update will continue to be updated to serve as the primary mechanism for communicating new issues, if any, that may arise after the release date.

Intended Audience

This Release Update is intended for users of System Release 6.0 PCI Windows.

How to Use This Publication

This Release Update is organized into four sections (click the section name to jump to the corresponding section):

- Document Revision History: This section summarizes the ongoing changes and additions that are made to this Release Update after its original release. This section is organized by document revision and document section.
- Post-Release Developments: This section describes significant changes to the system release subsequent to the general availability release date. For example, the new features provided in Service Updates are described here.
- Release Issues: This section lists issues that may affect the system release hardware and software. The primary list is sorted by issue type, but alternate sorts by defect number, by product or component, and by Service Update number are also provided.
- Documentation Updates: This section contains corrections and other changes that apply to the System Release documentation set that were not made to the documents prior to the release. The updates are organized by documentation category and by individual document.

Related Information

See the following for additional information:

- For information about the products and features supported in this release, see the *Dialogic® System Release 6.0 PCI for Windows® Release Guide,* which is included as part of the documentation bookshelf for the release.
- For further information on issues that have an associated defect number, you may use the Defect Tracking tool at *http://membersresource.dialogic.com/defects/*. When you select this link, you will be asked to either LOGIN or JOIN.
- *http://www.dialogic.com/support/* (for Dialogic technical support)
- *http://www.dialogic.com/* (for Dialogic® product information)

Document Revision History

This Revision History summarizes the changes made in each published version of the Release Update for Dialogic® System Release 6.0 PCI for Windows®, which is a document that has been and is intended to be periodically updated throughout the lifetime of the release.

Document Rev 62 - published January 30, 2008

Updated for Service Update 181.

In the Post-Release Developments section:

- Added Runtime Control of Single or Double Hookflash on Consultation Drop for FXS/LS Protocol.
- Deleted the detailed descriptions about some Dialogic® Global Call SS7 features that were previously included in this section, because this information has been incorporated into the updated *Dialogic® Global Call SS7 Technology Guide* that is now on the online documentation bookshelf.

In the Release Issues section, added the following resolved problems: IPY00040874, IPY00041079, IPY00041421, IPY00041426.

In the Documentation Updates section:

- Added documentation update to the Dialogic® Global Call E1/T1 CAS/R2 Technology Guide because of a new feature in the Service Update.
- Deleted the corrections for the Dialogic® Global Call SS7 Technology Guide, because these corrections have been incorporated into an updated document that is now on the online documentation bookshelf.
- Added documentation update to the Dialogic® Voice API Library Reference for the **dx_getdig( )** function (IPY00038453).
- Deleted the sections for the *Dialogic® Conferencing (CNF) API Library Reference* and *Dialogic® Conferencing (CNF) API Programming Guide*. These documents have been removed from the online documentation bookshelf because the CNF API is no longer supported in Dialogic® System Release 6.0 PCI for Windows®.

Document Rev 61 - published December 28, 2007

Updated for Service Update 178.

In the Release Issues section, added the following resolved problems: IPY00039334, IPY00040536, IPY00041078, IPY00041082, IPY00041178, IPY00041209, IPY00041233, IPY00041345.

In the Documentation Updates section:

- Deleted the corrections for the *Dialogic® Audio Conferencing API Library Reference* and *Dialogic® Audio Conferencing API Programming Guide*, because these corrections have been incorporated into updated documents that are now on the online documentation bookshelf.
- Deleted some of the corrections for the *Dialogic® Continuous Speech Processing API Library Reference* and *Dialogic® Continuous Speech Processing API Programming Guide*, because these corrections have been incorporated into updated documents that are now on the online documentation bookshelf.
- Deleted the corrections for the *Dialogic® Standard Runtime Library API Library Reference* and *Dialogic® Standard Runtime Library API Programming Guide*, because these corrections have been incorporated into updated documents that are now on the online documentation bookshelf.

Document Rev 60 - published November 15, 2007

Updated for Service Update 174.

In the Post-Release Developments section:

- Added Analog Call Transfer Support on Dialogic® Springware Boards.
- Added Windows Server® 2003 R2 SP2 under New Operating System Support.

In the Release Issues section, added the following resolved problems: IPY00038391, IPY00039490, IPY00039661, IPY00040096, IPY00040685, IPY00040798, IPY00040832. Also added IPY00040179 (resolved in Service Update 171).

In the Documentation Updates section:

- Added documentation updates to the following documents because of new features in the Service Update: *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*, *Dialogic® Global Call API Library Reference*, *Dialogic® Global Call Analog Technology Guide*.
- Added documentation updates to the *Dialogic® Springware Architecture Products on Windows® Configuration Guide*, Dialogic® Configuration Manager (DCM) Online Help, and *Dialogic® Continuous Speech Processing API Programming Guide* regarding use of the **EC_Resource** and **CSPExtraTimeSlot** parameters on Dialogic® Springware Boards (IPY00041018).
- Added documentation updates to the *Dialogic® Fax Software Reference* for additional return values for **ATFX_RESLN( )** and other related changes (IPY00040796).
- Added documentation update to the *Dialogic® GDK Programming Reference Manual* about the **fn_received** queue record field (IPY00040964).
- Added documentation update to the *Dialogic® Global Call ISDN Technology Guide* for additional firmware-related cause values when using Dialogic® DM3 Boards (IPY00041046).
- Added documentation updates to the *Dialogic® Voice API Library Reference* and *Dialogic® Voice API Programming Guide* for functions that are no longer supported (**r2_creatfsig( )** and **r2_playbsig( )**).

Document Rev 59 - published October 9, 2007

Updated for Service Update 171.

In the Post-Release Developments section, added Support for Windows Vista® Operating System.

In the Release Issues section

- Added the following resolved problems: IPY00039476, IPY00040052.
- Added the following known problems: IPY00040083, IPY00040086.

Made global changes to reflect Dialogic brand.

Document Rev 58 - published September 14, 2007

Updated for Service Update 167.

In the Post-Release Developments section:

- Added Dialogic® DM3 Media Channel Reset Capability (Stuck Channel Recovery).
- Under AMD Opteron Server Support, deleted the note about unsupported hardware; the issues have been resolved.

In the Release Issues section, added the following resolved problem: IPY00039014.

In the Documentation Updates section, added documentation updates to the following documents because of a new feature in the Service Update: *Dialogic® Continuous Speech Processing API Library Reference*, *Dialogic® Voice API Library Reference*.

Document Rev 57 - published September 6, 2007

Additional update for Service Update 166.

In the Post-Release Developments section, added AMD Opteron Server Support.

Document Rev 56 - published August 30, 2007

Updated for Service Update 166.

In the Release Issues section, added the following resolved problems: IPY00038190, IPY00038981, IPY00039068, IPY00039412, IPY00039427, IPY00039538, IPY00039586.

In the Documentation Updates section, deleted some of the corrections for the *Dialogic® Voice API Library Reference*, because these corrections have been incorporated into an updated document that is now on the documentation bookshelf.

Document Rev 55 - published August 20, 2007

Updated for Service Update 165.

In the Post-Release Developments section, added Global DPD Enabled on Dialogic® Springware Boards.

In the Release Issues section:

- Added the following resolved problems: IPY00037918, IPY00038545, IPY00038572, IPY00039155, IPY00039331, IPY00039341, IPY00039492.
- Eliminated the link to view issues sorted by PTR number. (PTR numbers have been superseded by defect numbers. The PTR numbers still appear in the Release Issues table for historical purposes, but a version of the table sorted by PTR number is no longer provided.)

In the Documentation Updates section, added a documentation update to the *Dialogic® Voice API Programming Guide* because of a new feature in the Service Update.

Document Rev 54 - published August 3, 2007

Updated for Service Update 162.

In the Release Issues section, added the following resolved problems: IPY00038551, IPY00038792, IPY00038946, IPY00039032, IPY00039179, IPY00039249.

Document Rev 53 - published July 20, 2007

Updated for Service Update 160.

In the Post-Release Developments section:

- Added Enhanced Special Information Tones on Dialogic® DM3 Boards Using Voice and Global Call APIs.
- Added Troubleshooting Information for RTF Logs.
- Added Remote Diagnostics Package.
- Under Enhanced Diagnostics, added PSTN Diagnostics (pstndiag) and Status Monitor (statusmon).
- Under Enhanced Diagnostics, added more tools that can now be executed from the New Dialogic® Diagnostics Management Console.
- Updated the Support for PCI Express Boards - Dialogic® Springware Boards section for the Dialogic® D/42JCT-EW and Dialogic® D/82JCT-EW PBX Integration Boards.
- Under Telecom Subsystem Summary Tool (its_sysinfo), added information about the new Windows® Package Info section.
- Deleted the section about compliance with ITU-T Q.454 and Q.455; this feature is not supported.

In the Release Issues section:

- Added the following resolved problems: IPY00037319, IPY00037643, IPY00037789, IPY00037923, IPY00038298, IPY00038407, IPY00038419, IPY00038433, IPY00038435, IPY00038494, IPY00038499, IPY00038524, IPY00038533, IPY00038539, IPY00038611, IPY00038612, IPY00038708, IPY00038836, IPY00038849, IPY00038894, IPY00038979, IPY00038991, IPY00038998.
- Revised the information for IPY00036665 (resolved in Service Update 160, not in Service Update 155).

In the Documentation Updates section, added documentation updates to the following documents because of new features in the Service Update: *Dialogic® System Software Diagnostics Guide*, *Dialogic® Global Call API Programming Guide*, *Dialogic® Voice API Library Reference*, *Dialogic® Voice API Programming Guide*.

Document Rev 52 - published June 25, 2007

Updated for Service Update 155.

In the Post-Release Developments section:

- Added New Parameter for Adjusting Silence Threshold on Dialogic® DM3 Boards.
- In the Support for PCI Express Boards - Dialogic® DM/V-B Products section, made minor changes to terminology in the Media Loads table.

In the Release Issues section:

- Added the following resolved problems: IPY00036665, IPY00037262, IPY00037493, IPY00038206, IPY00038280, IPY00038317. Also added IPY00037861 (resolved in Service Update 154).
- Added the following known (permanent) problem: IPY00037706.

In the Documentation Updates section:

- Added updates to the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* about NFAS D channel backup (DCBU) supported on 4ESS, 5ESS, and NI-2.
- Added an update to the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* about active talker and scaling in conferences.
- Added an update to the *Dialogic® Fax Software Reference* about the default fax font (IPY00037855).

Document Rev 51 - published June 13, 2007

Additional updates for Service Update 154.

In the Post-Release Developments section, updated the Support for PCI Express Boards - Dialogic® Springware Products section for the D/240JCT-T1-EW and D/300JCT-E1-EW PCI Express Boards.

In the Release Issues section, added the following resolved problems: IPY00010514 (PTR 35342), IPY00028248 (PTR 33718), IPY00028516 (PTR 35001), IPY00028549 (PTR 35901), IPY00028555 (PTR 36110), IPY00031590 (PTR 36755).

Document Rev 50 - published June 1, 2007

Updated for Service Update 154.

In the Post-Release Developments section:

- Added Support for PCI Express Boards - Dialogic® Station Interface Boards.
- In the Support for PCI Express Boards - Dialogic® DM/V-B Products and Support for PCI Express Boards - Dialogic® Springware Products sections, revised names of the PCI Express Boards to indicate their item market names.

In the Release Issues section:

- Added the following resolved problems: IPY00032797, IPY00033228, IPY00036855, IPY00037161, IPY00037166, IPY00037351, IPY00037372, IPY00037373, IPY00037467, IPY00037507, IPY00037777, IPY00037817, IPY00037818, IPY00038074, IPY00038119, IPY00038130, IPY00038235, IPY00038244.
- Added the following known problem: IPY00037923.
- Added the following known (permanent) problem: IPY00006127 (PTR 33837).

In the Documentation Updates section:

- Added an update for the **gc_InitXfer( )** function under *Dialogic® Global Call API Library Reference* (IPY00038401).
- Added an update for the **dx_setevtmsk( )** function under *Dialogic® Voice API Library Reference* (IPY00038053).

Document Rev 49 - published April 20, 2007

Updated for Service Update 148.

In the Post-Release Developments section:

- Updated the Support for PCI Express Boards - Dialogic® Springware Products section for the D/480JCT and D/600JCT PCI Express Boards.
- Added Windows® Server 2003 SP2 under New Operating System Support.

In the Release Issues section, added the following resolved problems: IPY00034857, IPY00036469, IPY00036919, IPY00037183, IPY00037318, IPY00037356, IPY00037396, IPY00037432, IPY00037483, IPY00037607, IPY00037632, IPY00037633, IPY00037708, IPY00037746, IPY00037767.

In the Documentation Updates section:

- Added a documentation update to the *Dialogic®* System Release 6.0 PCI for Windows® *Release Guide* for Windows Server® 2003 SP2 support.

- Added an update to the Media Load table under *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Document Rev 48 - published March 22, 2007

Updated for Service Update 144.

In the Post-Release Developments section:

- Added Support for PCI Express Boards - Dialogic® DM/V-B Products.
- Added Support for Dialogic® D/4PCI Board.
- Added New Parameter for Adjusting Silence Threshold on Dialogic® DM3 Boards.

In the Release Issues section:

- Added the following resolved problems: IPY00036504, IPY00036861.
- Added the following known problem: IPY00035574.

In the Documentation Updates section, added information about binary log files to the *Dialogic® System Software Diagnostics Guide* (IPY00037518).

Document Rev 47 - published March 13, 2007

In the Post-Release Developments section, updated the Support for PCI Express Boards section for additional PCI Express Boards: Dialogic® D/4PCIU4S, D/4PCIUF, D/41JCT-LS, and VFX/41JCT-LS.

Document Rev 46 - published March 5, 2007

Updated for Service Update 142.

In the Release Issues section:

- Added the following resolved problems: IPY00006707 (PTR 33803), IPY00007470 (PTR 32437), IPY00009499 (PTR 33932), IPY00028633 (PTR 35748), IPY00036280, IPY00036345, IPY00036347, IPY00036423, IPY00036448, IPY00036830, IPY00036833, IPY00036865, IPY00036886, IPY00037004. Also added IPY00034365 (resolved in Service Update 139).

Note: The fix for defect **IPY00036345** may have an impact on existing Dialogic® Springware applications; refer to the defect description in the Release Issues section.

- Added the following known (permanent) problem: IPY00037015.

Document Rev 45 - published February 5, 2007

Updated for Service Update 139.

In the Post-Release Developments section, added File Management Enhancements for Dialogic® ISDNtrace Tool.

In the Release Issues section, added the following resolved problems: IPY00031534, IPY00036044, IPY00036101, IPY00036247, IPY00036248, IPY00036337, IPY00036418, IPY00036949.

In the Documentation Updates section:

- Added IPY00006024 (PTR 29612) under *Dialogic® PBX Integration Board User's Guide*.
- Added documentation updates to the following document because of new features in the Service Update: *Dialogic® System Software Diagnostics Guide*.

Document Rev 44 - published January 3, 2007

Updated for Service Update 134.

In the Release Issues section, added the following resolved problems: IPY00034413, IPY00034841, IPY00035350, IPY00035613, IPY00035822, IPY00035831, IPY00035875. Also, added IPY00028444 (PTR 35763) (resolved in Service Update 124).

Document Rev 43 - published December 18, 2006

Updated for Service Update 133.

In the Post-Release Developments section:

- Added Support for Dialogic® DI/0408-LS-AR2 Board.
- Added Change in ipmedia.log Implementation.
- Added Adjusting Pre-Record Beep Tone Characteristics through the CONFIG File.
- Added Reduced Dial Tone Delay with MWI.

In the Release Issues section, added the following resolved problem: IPY00036073.

Document Rev 42 - published November 15, 2006

Updated for Service Update 131.

In the Post-Release Developments section:

- Added Enhanced Diagnostics.
- Added Support for PCI Express Boards.
- Deleted some of the detailed descriptions about diagnostics features that were previously included in this section, because this information is now superseded by the updated *Dialogic® System Software Diagnostics Guide* that is now on the documentation bookshelf.

In the Release Issues section, added the following resolved problems: IPY00006790 (PTR 35137), IPY00033492, IPY00034404, IPY00034495, IPY00034606, IPY00034738, IPY00034816, IPY00035148, IPY00035451, IPY00035506.

In the Documentation Updates section:

- Deleted the corrections for the *Dialogic® System Release 6.0 PCI for Windows® Software Installation Guide* because these corrections have been incorporated into an updated document that is now on the documentation bookshelf.
- Deleted the relevant corrections for the *Dialogic® System Software Diagnostics Guide* because these corrections have been incorporated into an updated document that is now on the documentation bookshelf.

Document Rev 41 - published October 18, 2006

Updated for Service Update 125.

In the Release Issues section, added the following resolved problems: IPY00033102, IPY00034079, IPY00034105, IPY00034378, IPY00034618, IPY00034678. Also, added IPY00032664 (resolved in Service Update 105) and IPY00032875 (resolved in Service Update 116).

Document Rev 40 - published September 26, 2006

Updated for Service Update 124.

In the Post-Release Developments section:

- Added PDK Trace Supports CAS/R2MF/Tone Tracing.
- Added Compliance with ITU-T Q.454 and Q.455.
- Added Ability to Lower or Disable White Noise.
- Added Optional Use of Sharing of Timeslot (SOT) Algorithm.
- Under Dialogic® Global Call Software Support for Time Slots on Dialogic® SS7 Boards Running in DTI Mode, deleted the restriction that opening trunk devices is not supported. Trunk devices can be opened.
- Under Notification of Layer 1 Alarm Events on Dialogic® SS7 Boards, revised the Alarm Handling for SS7 Boards section to indicate that GCEV_ALARM events are disabled by default and must be enabled via **gc_SetAlarmConfiguration( )**.

In the Release Issues section, added the following resolved problems: IPY00033163, IPY00033698. Also, added IPY00033244 (resolved in Service Update 113).

In the Documentation Updates section:

- Added documentation updates to the following documents because of new features in the Service Update: *Dialogic® System Software for DM3 Architecture Products on Windows® Diagnostics Guide* and *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.
- Added updates about fixed-line short message service (SMS) support on Dialogic® Springware Boards under *Dialogic® Voice API Programming Guide*.

Document Rev 39 - published August 22, 2006

Updated for Service Update 118.

In the Release Issues section:

- Added the following resolved problem: IPY00030001 (PTR 36796).
- Added the following known (permanent) problem: IPY00031563 (PTR 36612).

Document Rev 38 - published August 7, 2006

Updated for Service Update 116.

In the Release Issues section, added the following resolved problems: IPY00034050 (PTR 36636). Also added IPY00034018 (fixed in Service Update 115).

Document Rev 37 - published August 4, 2006

Updated for Service Update 115.

In the Post-Release Developments section:

- Added New FSK Transmit and Receive Signal Level Parameters.
- Added Support for Reporting Billing Type.
- Added Runtime Control of Double Answer for R2MF.
- Added Enhanced ISDN Trace Functionality for DPNSS Tracing.

In the Release Issues section, added the following resolved problems: IPY00007931 (PTR 23718), IPY00033499.

In the Documentation Updates section:

- Added IPY00006258 (PTR 36353) under *Dialogic® PBX Integration Board User's Guide*.
- Added documentation updates to the following documents because of new features in the Service Update: *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*, *Dialogic® Global Call API Library Reference*, *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*.

Document Rev 36 - published July 26, 2006

Updated for Service Update 113.

In the Post-Release Developments section, under New Features in Dialogic® Global Call Protocols Package, added five more new protocols (Bulgaria R2, Croatia R2, Kuwait R2, Lithuania R2, Uzbekistan R2) and new parameters for Nortel Meridian Lineside E1 protocol.

In the Release Issues section:

- Added the following resolved problems: IPY00031559 (PTR 36828), IPY00031560 (PTR 36801), IPY00032793, IPY00033009, IPY00033584.
- Added three known (permanent) problems regarding Runtime Trace Facility: IPY00032730, IPY00032735, IPY00032742.

In the Documentation Updates section:

- Added an update to the **NCM_ApplyTrunkConfiguration( )** function under *Dialogic® Native Configuration Manager API Library Reference*.
- Added IPY00006540 (PTR 34211) under *Dialogic® Global Call ISDN Technology Guide*.
- Added IPY00033772 under *Dialogic® Voice API Library Reference*.

Document Rev 35 - published July 5, 2006

Updated for Service Update 111.

In the Post-Release Developments section, added information about the following:

- Notification of Layer 1 Alarm Events on Dialogic® SS7 Boards.
- Dialogic® Global Call Software Support for Time Slots on Dialogic® SS7 Boards Running in DTI Mode.
- Time Stamp for Tone-On/Off Events.

In the Release Issues section, added the following resolved problems: IPY00031588 (PTR 36770), IPY00033410.

In the Documentation Updates section, added information about the following:

- Notification of Layer 1 alarm events on Dialogic® SS7 Boards in the *Dialogic® Global Call SS7 Technology Guide* and *Dialogic® Global Call API Library Reference*.
- Dialogic® Global Call Software support for time slots on Dialogic® SS7 Boards running in DTI mode in the *Dialogic® Global Call SS7 Technology Guide*.
- Time stamp for Tone ON/OFF events in the *Dialogic® Voice API Library Reference.*

Document Rev 34 - published June 28, 2006

Updated for Service Update 110.

In the Release Issues section, added the following resolved problems: IPY00029931 (PTR 36809), IPY00031597 (PTR 36527), IPY00032715.

In the Documentation Updates section:

- Added IPY00033335 under *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

- Added IPY00006520 (PTR 36259), IPY00006556 (PTR 35326), and IPY00006570 (PTR 35992) under *Dialogic® Fax Software Reference*.
- Added IPY00006537 (PTR 35666), IPY00006580 (PTR 34546), IPY00006581 (PTR 35616), and IPY00006594 (PTR 36685) under *Dialogic® Voice API Programming Guide.*

Document Rev 33 - published June 12, 2006

Updated for Service Update 108.

In the Post-Release Developments section, added information about the New Fax Parameter for Modem Receive Level.

In the Release Issues section, added the following resolved problems: IPY00006562 (PTR 35636), IPY00028341 (PTR 35790), IPY00030882 (PTR 36057), IPY00031529 (PTR 36814), IPY00031535 (PTR 36852), IPY00031536 (PTR 36637), IPY00031561 (PTR 36775), IPY00032244 (PTR 36750), IPY00032363, IPY00032794, IPY00032796, IPY00032803, IPY00033013, IPY00033029, IPY00033122, IPY00033185. Revised information about IPY00028341 (PTR 35790) - resolved in Service Update 108, not Service Update 65.

In the Documentation Updates section, added information about setting parameters to receive fax under *Dialogic® Fax Software Reference*.

Document Rev 32 - published May 26, 2006

Updated for Service Update 105.

In the Post-Release Developments section, added information about the following:

- Ability to Send and Receive DPNSS End to End Messages, which is the ability to send and receive raw DPNSS end to end message using API control on Dialogic® DM3 Boards.
- Enable RTF Logging on Dialogic® DM3 Libraries by entering module names in the RTF config file.

In the Release Issues section, added the following resolved problem: IPY00031550 (PTR 36859).

In the Documentation Updates section, added information about the following:

- New message type and event for DPNSS end to end messages.
- Enable RTF logging on Dialogic® DM3 libraries.

Document Rev 31 - published May 15, 2006

Updated for Service Update 104.

In the Post-Release Developments section, added information about the following:

- PDK Configuration Property Sheet which is a new property sheet in DCM.
- Automatic FCD File Generation, which provides an enhanced way to generate an updated FCD file.
- Centralized Logging using Runtime Trace Facility (RTF), which logs OA&M components to the RTF.
- New Option for Dialogic® dm3post Utility, which provides an option to run POST on a chassis level.
- New OAMIPC Mechanism Replaces CORBA, which will no longer be used during installation.
- Support for Mixed ISDN and Clear Channel on Additional Dialogic® DM3 Boards, which is the ability to mix ISDN (Net5) and clear channel on the same board on a trunk by trunk basis.
- Detection of Unsupported Boards.

In the Release Issues section:

- Added the following resolved problem: IPY00032271 (PTR 36699). Also added IPY00006348 (PTR 36782) (fixed in Service Update 103).
- Added the following known problem: IPY00033013.

In the Documentation Updates section, added information about the following:

- PDK Configuration property sheet because of a new feature in DCM. Added document update for *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* and *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*.
- Java Runtime Environment error messages. Added document update for *Dialogic® System Release 6.0 PCI for Windows® Software Installation Guide*.
- Automatic FCD File Generation. Added document update for *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.
- Centralized logging using Runtime Trace Facility (RTF). Added document update for *Dialogic® System Software for DM3 Architecture Products on Windows® Diagnostics Guide*.
- New Option for dm3post Utility. Added document update for *Dialogic® System Software for DM3 Architecture Products on Windows® Diagnostics Guide*.
- New OAMIPC Mechanism replaces CORBA. Added document update for *Dialogic® System Release 6.0 PCI for Windows® Software Installation Guide*.
- Support for Mixed ISDN and Clear Channel on Additional DM3 Boards. Added document update for *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Document Rev 30 - published May 3, 2006

Updated for Service Update 100.

In the Post-Release Developments section, added PBX Integration Support for Nortel BCM.

In the Release Issues section, added the following resolved problems: IPY00006712 (PTR 36790), IPY00006846 (PTR 36711), IPY00028547 (PTR 35670), IPY00031562 (PTR 36766).

In the Documentation Updates section:

- Added a documentation update to the *Dialogic® PBX Integration Board User's Guide* because of a new feature in the Service Update.
- Added documentation updates about the PhysicalSlotNumber and PciID parameters under *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* and DCM Online Help.
- Added IPY00006588 (PTR 36210) under *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* and *Dialogic® Global Call API Programming Guide*.
- Added IPY00032691 under *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*.
- Added IPY00029956 (PTR 36646) under *Dialogic® Global Call IP Technology Guide*.
- Added IPY00006590 (PTR 36501) under *Dialogic® Global Call ISDN Technology Guide*.

Document Rev 29 - published April 21, 2006

Updated for Service Update 98.

Note: The Release Issues section has been modified to show issues by Change Control System defect number and by PTR number. Issues reported prior to March 27, 2006, will be identified by both numbers. Issues reported after March 27, 2006, will only have a defect number.

In the Post-Release Developments section:

- Updated the Dynamically Retrieving and Modifying Selected Protocol Parameters When Using Dialogic® DM3 Boards section to add information about Retrieving or Modifying CDP Variable Values and Extension of GC_RTCM_EVTDATA. Also updated the Restrictions and Limitations.
- Added information about a new media load, Media Load QSB-ML10-LC, under New Media Loads for Dialogic® DMV1200BTEP Boards. Also revised the information about Media Load QSB-U3 to indicate that CSP streaming to CT Bus is no longer supported with this media load.

In the Release Issues section:

- Added the following resolved problems: IPY00006345 (PTR 36788), IPY00006647 (PTR 36598), IPY00006856 (PTR 36800), IPY00006862 (PTR 36830), IPY00010760 (PTR 36647), IPY00010900 (PTR 36349), IPY00011037 (PTR 36677), IPY00031596 (PTR 36840), IPY00031791 (PTR 36793), IPY00032239 (PTR 36769).

- Added the following known problems: IPY00006353 (PTR 36792), IPY00006393 (PTR 36758), IPY00006407 (PTR 36806), IPY00031561 (PTR 36775), IPY00032271 (PTR 36699).

In the Documentation Updates section:

- Added documentation update to the following document because of a new feature in the Service Update: *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.
- Added documentation updates for the *Dialogic® Digital Network Interface Software Reference*.
- Added IPY00031917 (PTR 27337) under *Dialogic® Fax Software Reference*.

Document Rev 28 - published March 23, 2006

Updated for Service Update 95.

In the Post-Release Developments section, added Windows Server® 2003 R2 under New Operating System Support.

In the Release Issues section, added the following resolved problems: 36640, 36688, 36698, 36735, 36780, 36810.

In the Documentation Updates section, added a documentation update to the following document because of a new feature in the Service Update: *Dialogic®* System Release 6.0 PCI for Windows® *Release Guide*.

Document Rev 27 - published March 16, 2006

Updated for Service Update 94.

In the Post-Release Developments section:

- Added Automatic Registration of DebugAngel Service.
- Added Windows® 2000 Update Rollup 1 for SP4 under New Operating System Support.
- Added the Dialogic® D/42-NE2 PCI PBX Integration Board under New Boards Supported.

In the Release Issues section:

- Added the following resolved problems: 35746, 36319, 36587, 36666. Also added 32842 (fixed in Service Update 70).
- Added the following known (permanent) problem: 36722.

In the Documentation Updates section:

- Added documentation updates to the following documents because of new features in the Service Update: *Dialogic®* System Release 6.0 PCI for Windows® *Release Guide*, *Dialogic® System Software for DM3 Architecture Products on Windows® Diagnostics*

Guide, *Dialogic® D/42 Series Software API Reference*, *Dialogic® D/42 Series User's Guide*.

- Added documentation update to the *Dialogic® Global Call IP Technology Guide* about the IP_H221NON.STANDARD data structure.

Document Rev 26 - published March 2, 2006

Updated for Service Update 92.

In the Post-Release Developments section, added Enhancements to Runtime Trace Facility (RTF) Logging.

In the Release Issues section:

- Added the following resolved problems: 35117, 36548, 36584, 36633, 36681, 36799. Also added 33144 (fixed in Service Update 18) and 33173 (fixed in Service Update 84).
- Added the following known (permanent) problem: 36119

In the Documentation Updates section:

- Added documentation updates to the following documents because of new features in the Service Update: *Dialogic® System Software for DM3 Architecture Products on Windows® Diagnostics Guide, Dialogic® Global Call API Programming Guide*
- Added PTR# 36260 under *Dialogic® Native Configuration Manager API Library Reference*.
- Added PTR# 36726 under *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*.
- Added PTR# 35565 under *Dialogic® Modular Station Interface API Library Reference*.

Document Rev 25 - published February 14, 2006

Updated for Service Update 90.

In the Release Issues section:

- Added the following resolved problems: 36134, 36302, 36329, 36416, 36606. Also added 33099 (fixed in Service Update 39).
- Added the following known (permanent) problems: 35879, 36716

In the Documentation Updates section:

- Added documentation updates to the *Dialogic®* System Release 6.0 PCI for Windows® *Release Guide*, *Dialogic® Conferencing (CNF) API Library Reference*, and *Dialogic® Conferencing (CNF) API Programming Guide* because of upcoming changes in support for the CNF API.
- Added PTR# 36674 under *Dialogic® Fax Software Reference*.
- Added PTR# 36660 under *Dialogic® Voice API Library Reference*.

Document Rev 24 - published February 2, 2006

Updated for Service Update 89.

In the Post-Release Developments section, added Dynamically Retrieving and Modifying Selected Protocol Parameters When Using Dialogic® DM3 Boards.

In the Documentation Updates section, added documentation updates to the following documents because of new features in the Service Update: *Dialogic® Global Call API Library Reference*, *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*, *Dialogic® Global Call ISDN Technology Guide*

Document Rev 23 - published January 31, 2006

Updated for Service Update 88.

In the Release Issues section, added the following resolved problem: 36333

In the Documentation Updates section:

- Added PTR# 36671 under *Dialogic®* System Release 6.0 PCI for Windows® *Release Guide*.
- Added PTR# 36278 under *Dialogic® System Software for DM3 Architecture Products on Windows® Diagnostics Guide*.

Document Rev 22 - published January 11, 2006

Updated for Service Update 87.

In the Post-Release Developments section:

- Added Analog Line Adaptation Utility (LineAdapt).
- Added New QSIG Channel Mapping Parameter for Dialogic® E1 Boards.

In the Release Issues section, added the following resolved problem: 36371

In the Documentation Updates section, added documentation updates to the following documents because of new features in the Service Update: *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*, *Dialogic® Springware Architecture Products on Windows® Configuration Guide*

Document Rev 21 - published January 6, 2006

Updated for Service Update 84.

In the Post-Release Developments section:

- Added IP Support on Dialogic® DI0408LSAR2 Boards.
- Added Dialogic® DI0408LSAR2 Product Support for Host Systems with Multiple NICs.

- Added Support for QSIG NCAS Calls on Dialogic® DM3 Boards.
- Added Loop Current Reversal Detection on the Dialogic® DMV160LP Board.
- Added Adjusting DTMF Characteristics through the CONFIG File.
- Added Single Board Start/Stop for Selected Dialogic® JCT Boards.
- Added New Media Load for Dialogic® DMV3600BP Boards.
- Revised Mixing ISDN and CAS on Dialogic® DM/V-B Boards section to mention that A-law/Mu-law conversion is supported.

In the Release Issues section:

- Added the following resolved problems: 31991, 33750, 34095, 34159, 34284, 35423, 35430, 35634, 35809, 35832, 35921, 36020, 36021, 36042, 36063, 36085, 36090, 36108, 36129, 36159, 36197, 36204, 36213, 36237, 36248, 36256, 36295, 36310, 36316, 36335, 36356, 36429
- Added the following known (permanent) problem: 34616

In the Documentation Updates section:

- Added PTR# 36373 under *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*.
- Added documentation updates to the following documents because of new features in the Service Update: *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*, *Dialogic® Springware Architecture Products on Windows® Configuration Guide*, *Dialogic® Board Management API Library Reference*, DCM Online Help, *Dialogic® System Software for PCI Products on Windows® Administration Guide*, *Dialogic® Global Call Analog Technology Guide*, *Dialogic® Global Call IP Technology Guide*, *Dialogic® Global Call ISDN Technology Guide*

Document Rev 20 - published November 4, 2005

Updated for Service Update 74.

In the Post-Release Developments section:

- Added SIP Call Transfer.
- Added Early Media.

In the Release Issues section:

- Added the following resolved problems: 32144, 34532, 34915, 35169, 35339, 35619, 35620, 35967, 36092, 36209
- Added the following known (permanent) problem: 36079

In the Documentation Updates section:

- Added PTR# 34210 under *Dialogic® Audio Conferencing API Library Reference* and *Dialogic® Audio Conferencing API Programming Guide*.
- Added PTR# 33036 under *Dialogic® Fax Software Reference*.
- Added PTR# 32087 under *Dialogic® Global Call IP Technology Guide*.

- Added PTR# 33826 under *Dialogic® IP Media Library API Programming Guide*.
- Added PTR# 34119 under *Dialogic® Standard Runtime Library API Programming Guide*.
- Added PTR# 33806 under *Dialogic® Voice API Library Reference*.

Document Rev 19 - published October 17, 2005

Updated for Service Update 71.

In the Post-Release Developments section, added Dialogic® Global Call SS7 Enhancements.

In the Release Issues section, added the following resolved problems: 33717, 34816, 35102, 35650

Document Rev 18 - published October 10, 2005

Updated for Service Update 70.

In the Post-Release Developments section:

- Added Conference Bridging on Dialogic® Station Interface Boards.
- Added New Parameter for Order of DNIS and ANI.

In the Release Issues section:

- Added the following resolved problems: 32415, 32772, 32855, 33413, 34569, 34858, 34886, 35327, 35417, 35538, 35839, 35851, 35898, 35937, 35991, 36043, 36081, 36091
- Added the following known (permanent) problem: 33991

In the Documentation Updates section:

- Added PTR# 36031 and 36105 under *Dialogic®* System Release 6.0 PCI for Windows® *Release Guide*. (For 36031, also added a note in Section 1.34, “New Operating System Support”, on page 159.)
- Added PTR# 35769 under *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*. Also added a documentation update to this guide because of a new feature in the Service Update.
- Added a documentation update to the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* because of a new feature in the Service Update.
- Added documentation updates to the *Dialogic® Continuous Speech Processing API Library Reference* and *Dialogic® Continuous Speech Processing API Programming Guide* about valid values for DXCH_EC_TAP_LENGTH on Dialogic® Springware Boards.
- Added PTR# 34237 and 35965 under *Dialogic® Global Call API Library Reference*.
- Added PTR# 35268 under *Dialogic® Global Call IP Technology Guide*.

Document Rev 17 - published September 2, 2005

Updated for Service Update 65.

In the Post-Release Developments section, added New Channel Block Timer for NTT Protocol.

In the Release Issues section, added the following resolved problems: 34814, 35011, 35270, 35330, 35566, 35671, 35704, 35775, 35790, 35799, 35825, 35875

In the Documentation Updates section, added a documentation update to the *Dialogic® Springware Architecture Products on Windows® Configuration Guide* because of a new feature in the Service Update.

Document Rev 16 - published August 19, 2005

Updated for Service Update 64.

In the Post-Release Developments section, added Mixing ISDN and CAS on Dialogic® DM/V-B Boards.

In the Release Issues section, added the following resolved problem: 35148

In the Documentation Updates section:

- Added a documentation update to the following document because of a new feature in the Service Update: *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.
- Added PTR# 35249 and 35844 under *Dialogic® Global Call ISDN Technology Guide*.
- Added documentation update about Application Development Guidelines under *Dialogic® Continuous Speech Processing API Programming Guide* and *Dialogic® Voice API Programming Guide*.

Document Rev 15 - published August 12, 2005

Updated for Service Update 63.

In the Release Issues section, added the following resolved problems: 32759, 34878, 35105, 35390, 35507, 35572, 35573, 35597, 35768

Document Rev 14 - published July 29, 2005

Updated for Service Update 62.

In the Release Issues section:

- Added the following known problems: 35105, 35148, 35572, 35573. Also added a known problem (no PTR number) with the Host Install affecting the use of PDKManager after an update install.

- Added the following resolved problems: 31675, 32313, 32712, 33514, 34160, 35104, 35134, 35170, 35232, 35281, 35321, 35412, 35431, 35438, 35458. In addition, the known problem with the update install from Service Update 58 has been resolved.

In the Documentation Updates section:

- Added PTR# 32933 under *Dialogic®* System Release 6.0 PCI for Windows® *Release Guide*.
- Added PTR# 33555/34771 under *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.
- Added PTR# 34244 under *Dialogic® Continuous Speech Processing API Library Reference* and *Dialogic® Continuous Speech Processing API Programming Guide*.
- Added documentation update about alarm handling for Dialogic® DM3 Boards under *Dialogic® Global Call ISDN Technology Guide*.

Document Rev 13 - published July 19, 2005

In the Release Issues section, added a known problem (no PTR number) with the Host Install.

Note: This problem only occurs when performing an **update install** (not a full install) of Service Update 58. Please check the Release Issues section for known problems with Host Install, and perform the workaround that is given.

Document Rev 12 - published July 12, 2005

Updated for Service Update 58.

In the Post-Release Developments section:

- Added Implementation of ROLM Call Waiting LED.
- Added information about two new media loads, QSB-U3 and QSB-ML10, under New Media Loads for Dialogic® DMV1200BTEP Boards.
- Added a new section, New Media Load for Dialogic® DMV600BTEP Boards, with information about media load DSB-U2.

In the Release Issues section, added the following resolved problem: 35154

In the Documentation Updates section:

- Added documentation updates to the following documents because of new features in the Service Update: *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*, *Dialogic® PBX Integration Board User's Guide*, *Dialogic® PBX Integration Software Reference*
- Provided additional information about event cause values (PTR# 34490) under *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*.

Document Rev 11 - published June 24, 2005

Updated for Service Update 56.

In the Post-Release Developments section:

- Revised Windows® Hardware Quality Labs (WHQL) Certification section to indicate that WHQL certification for Dialogic® System Release 6.0 PCI for Windows® Service Update is not currently valid; the product is getting recertified.
- Added Enhanced Special Information Tone Frequency Detection on Dialogic® DM3 Boards.
- Added Enhanced GCAMS on Dialogic® DM3 Boards.
- Added Telecom Subsystem Summary Tool (its_sysinfo).
- Revised New Features in Global Call Protocols Package for the latest features that are now available.
- Added support for Windows Server® 2003 SP1 under New Operating System Support.
- Added information about a new media load, 10b, under New Media Loads for Dialogic® DMV1200BTEP Boards.

In the Release Issues section:

- Added the following resolved problems: 30233, 31912, 32103, 32265, 32458, 32539, 32953, 33019, 33199, 33249, 33385, 33685, 33816, 33939, 33998, 34032, 34050, 34175, 34269, 34274, 34329, 34344, 34397, 34427, 34476, 34495, 34503, 34516, 34537, 34543, 34575, 34586, 34587, 34640, 34663, 34664, 34685, 34719, 34753, 34788, 34805, 34862, 34921, 34972, 34985, 34999, 35012, 35013, 35035, 35042, 35049, 35077, 35130, 35132, 35157, 35159, 35190, 35210
- Added the following known problem: 33137
- Added the following known (permanent) problems: 32588, 35118

In the Documentation Updates section:

- Added documentation updates to the following documents because of new features in the Service Update: *Dialogic®* System Release 6.0 PCI for Windows® *Release Guide*, *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*, *Dialogic® Global Call API Library Reference*, *Dialogic® Voice API Library Reference*, *Dialogic® Voice API Programming Guide*
- Added PTR# 33698 and 33699 under *Dialogic® GDK 5.0 Installation and Configuration Guide for Windows®*.
- Added documentation update about RTF to *Dialogic® System Software for DM3 Architecture Products on Windows® Diagnostics Guide*.
- Added documentation update about multithreaded programming to *Dialogic® Audio Conferencing API Programming Guide*.
- Added PTR# 33852 and made a correction to the GCLIB_MAKECALL_BLK data structure reference page under *Dialogic® Global Call API Library Reference*.
- Added PTR# 33202 under *Dialogic® Global Call Analog Technology Guide*.

- Added PTR# 29448, 34490, and 35050 under *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*.
- Added PTR# 34285 and 34840 under *Dialogic® Voice API Library Reference*.

Document Rev 10 - published March 22, 2005

Updated for Service Update 39.

In the Release Issues section:

- Added the following resolved problems: 34121, 34241, 34345, 34393, 34478
- Added "SU No." column to the Issues table to show the Service Update number for resolved PTRs. Also added a link to view the Issues table sorted by Service Update number.

In the Documentation Updates section, added information about support for Intel Hyper-Threading Technology in the *Dialogic®* System Release 6.0 PCI for Windows® *Release Guide*.

Document Rev 09 - published February 25, 2005

Updated for Service Update 37.

In the Post-Release Developments section, added the following new features:

- Windows® Hardware Quality Labs (WHQL) Certification
- Single Echo Canceller Convergence
- New Features in Dialogic® Global Call Protocols Package

In the Release Issues section:

- Added the following resolved problem: 34319
- Added the following known problem: 34764

In the Documentation Updates section:

- Added documentation updates about DCM to *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* and *Dialogic® Springware Architecture Products on Windows® Configuration Guide*.
- Added PTR# 32847 to *Dialogic® Continuous Speech Processing API Library Reference*.
- Added PTR# 32607/21073 to *Dialogic® Continuous Speech Processing API Programming Guide*.
- Added documentation updates about ECCH_XFERBUFFERSIZE to *Dialogic® Continuous Speech Processing API Library Reference* and *Dialogic® Continuous Speech Processing API Programming Guide*. Also added documentation updates about single echo canceller convergence because of a new feature in the Service Update.

- Added PTR# 32544, PTR# 32501, and PTR# 32616 to *Dialogic® Global Call API Library Reference*.
- Added PTR# 32481 to *Dialogic® Global Call API Programming Guide*.
- Added PTR# 32379 to *Dialogic® Global Call Analog Technology Guide*.
- Added PTR# 32966 to *Dialogic® Standard Runtime Library API Programming Guide* and *Dialogic® Voice API Programming Guide*.
- Added PTR# 32681 to *Dialogic® Voice API Library Reference* and *Dialogic® Voice API Programming Guide*.
- Added PTR# 32643, PTR# 32106, and PTR# 30881 to *Dialogic® Voice API Library Reference*.

Document Rev 08 - published January 21, 2005

Updated for Service Update 30.

In the Post-Release Developments section, added the following new features:

- Windows® XP SP2 Support
- New Station Interface Alarms

In the Release Issues section, added the following resolved problems: 30390, 31583, 32188, 32590, 32827, 33772

In the Documentation Updates section, added documentation updates to the following documents because of new features in the Service Update: *Dialogic® Modular Station Interface API Library Reference*, *Dialogic® Modular Station Interface API Programming Guide*

Document Rev 07 - published December 23, 2004

Updated for Service Update 27.

In the Post-Release Developments section, added support for the Dialogic® D/4PCIU4S Media Board.

In the Release Issues section, added the following resolved problems: 32571, 33981, 33994, 34048, 34054, 34063

Document Rev 06 - published December 9, 2004

Updated for Service Update 25.

In the Release Issues section, added the following resolved problems: 31747, 32343, 32978, 33782, 34053

Document Rev 05 - published November 15, 2004

Updated for Service Update 22.

In the Post-Release Developments section:

- Revised the information about installing the Service Update.
- Added the following new features:
  - Support for ANI Category Digit Retrieval on Dialogic® DM3 Boards
  - New Media Load for Dialogic® DMV1200BTEP Boards

In the Release Issues section:

- Added the following resolved problems: 27539, 28620, 31632, 31633, 31661, 31896, 32060, 32318, 32979, 33011, 33200, 33501, 33690
- Added the following known problems: 33019, 34054
- Deleted some PTRs that were not applicable to this release.

In the Documentation Updates section:

- Added documentation updates to the following documents because of new features in the Service Update: *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*, *Dialogic® Global Call API Library Reference*, *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*
- Deleted the corrections for the *Dialogic® System Release 6.0 PCI for Windows® Software Installation Guide*, because these corrections have been incorporated into the updated document that is now on the documentation bookshelf.
- Added correction for PTR# 33615 in the *Dialogic®* System Release 6.0 PCI for Windows® *Release Guide*.
- Added correction about CSP support on Dialogic® DM/IP Boards in the *Dialogic®* System Release 6.0 PCI for Windows® *Release Guide*.

Document Rev 04 - published October 15, 2004

Added a new section, Post-Release Developments, to describe the new features provided in Service Update 18.

Added the following resolved problems to the Release Issues section: 17567, 25633, 27336, 27563, 27764, 28550, 29328, 29445, 29859, 31242, 31333, 31530, 31777, 31778, 31782, 31840, 31844, 31850, 31945, 32014, 32026, 32065, 32104, 32108, 32111, 32161, 32192, 32209, 32275, 32303, 32411, 32416, 32435, 32441, 32443, 32444, 32510, 32547, 32554, 32557, 32601, 32615, 32625, 32651, 32678, 32696, 32704, 32725, 32733, 32765, 32773, 32810, 32846, 32858, 32913, 33053, 33056, 33069, 33070, 33146, 33156, 33334, 33351, 33389, 33425, 33443, 33444, 33502, 33519, 33543, 33596, 33665, 33694

Added the following known problems to the Release Issues section: 32882, 33625, 33633, 33730, 33939

In the Documentation Updates section:

- Added documentation updates to the following documents because of new features provided in the Service Update: *Dialogic®* System Release 6.0 PCI for Windows® *Release Guide*, *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*, *Dialogic® Board Management API Library Reference*, *Dialogic® Global Call API Library Reference*, *Dialogic® Global Call Analog Technology Guide*, *Dialogic® Voice API Library Reference*
- Revised the correction that was previously entered for PTR# 33046 under *Dialogic®* System Release 6.0 PCI for Windows® *Release Guide*.

Document Rev 03 - published June 7, 2004

Added PTR# 31812/32282 in the Documentation Updates section under *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Added PTR# 33046 in the Documentation Updates section under *Dialogic®* System Release 6.0 PCI for Windows® *Release Guide*.

Added PTR# 32824 in the Documentation Updates section under *Dialogic® Global Call IP Technology Guide*.

Document Rev 02 - published March 29, 2004

Added PTR# 32418 in the Documentation Updates section under *Dialogic® Global Call ISDN Technology Guide*.

Added PTR# 27774 in the Documentation Updates section under *Dialogic® Voice API Library Reference*.

Added a reference to the Media Load Densities on Dialogic® DMV-B Multifunction Series Boards technote in the Release Issues table.

Removed the workaround statement for PTR# 32144 in the Release Issues table. Further testing revealed that the workaround is not feasible and the issue may still arise.

Document Rev 01 - published March 4, 2004

Initial version of document.

Post-Release Developments

This section describes significant changes to the system release subsequent to the general availability release date.

1.1 Service Update for Dialogic® System Release 6.0 PCI for Windows®

A Service Update for Dialogic® System Release 6.0 PCI for Windows® is now available. Service Updates provide fixes to known problems, and may also introduce new functionality. New versions of the Service Update are planned to be released periodically. It is intended that this Release Update will document the features in the Service Updates.

Depending on whether you already have a version of Dialogic® System Release 6.0 PCI for Windows® on your system, installing the Service Update will give you either a **full install** or an **update install**:

- If you don't have an existing version of System Release 6.0 PCI Windows on your system, installing the Service Update gives you a **full install** of the system release. You can select the features that you want to install, for example, Development

Package, Core Runtime Package, ISDN Protocols, Demos, SNMP Component Manager, Global Call Protocols, and Documentation.

Note: With the Service Update, the Global Call Protocols Package can now be installed as part of System Release 6.0 PCI Windows. Previously, this package was installed separately.

The Development Package and Demos are available in the Developer Edition only, not in the Redistributable Edition.

- If you have an existing version of System Release 6.0 PCI Windows on your system, installing the Service Update gives you an **update install**. The update install gives you the latest software for the features that you selected when you did the full install of the system release that is currently on your system. If you want additional features, such as the Global Call Protocols Package, you can use the Modify or Change option as explained in the Installation Guide.

A new *Dialogic® System Release 6.0 PCI for Windows® Software Installation Guide* has been added to the documentation bookshelf to describe the full install and update install procedures. The *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for the Global Call Protocols Package has also been added to the bookshelf.

Note: Since the Global Call Protocols Package is now included with this Service Update version of System Release 6.0 PCI Windows, the stand-alone protocols package should not be used. (If you already have the stand-alone protocols package installed, you will be prompted to remove it before installing the Service Update.) Do not install the stand-alone protocols package after installing the Service Update (full install or update install), or your software may become non-functional.

See the new *Dialogic® System Release 6.0 PCI for Windows Software Installation Guide* on the documentation bookshelf for complete, detailed information about installing the software.

1.2 Runtime Control of Single or Double Hookflash on Consultation Drop for FXS/LS Protocol

With the Service Update, runtime control of sending either a single or double hookflash when dropping a consultation call on a supervised transfer is now supported for Dialogic® DM3 Boards using the United States T1 FXS/LS Bidirectional protocol.

1.2.1 Feature Description

The signal pattern normally used by the FXS/LS protocol to drop a supervised transfer consultation call is a *single hookflash*. For PBXs that require a *double hookflash* to drop a consultation call, this can be set in the country dependent parameters (CDP) file for the FXS/LS protocol, *pdk_us_ls_fxs_io.cdp*, by enabling the **CDP_AllowDblHookflashOnConsultationDrop** parameter. (This parameter is disabled by default.) CDP file parameters are set on a board basis. Parameter settings are static and apply to all calls (per board).

However, some PBXs may require *either a single or double hookflash* depending on the circumstances of the call. For example, a particular PBX may require:

- Single hookflash on consultation call drop if the call went through
- Double hookflash on consultation call drop if the call was in progress but did not go through and never got connected (for example, call progress failure or call abort before connect)

Note: These are only examples; the circumstances requiring a single or double hookflash can vary depending the PBX. It is up to the application developer to determine when to apply a single or double hookflash in any scenario or deployment.

For PBXs that require either a single or double hookflash, applications must be able to:

- Programmatically select either single or double hookflash when dropping a consultation call in a supervised transfer
- Change this behavior on a call-by-call basis

Call Transfer Overview

An overview of call transfer is given elsewhere in this Release Update. See Section 1.3.1, "Call Transfer Overview", on page 38.

New Parameter for Single or Double Hookflash

Runtime control of single or double hookflash is implemented using the Dialogic® Global Call **gc_SetConfigData( )** function. The parameter settings in **gc_SetConfigData( )** are limited to the current call, that is, to the call reference number (CRN) specified as the **target_id** in **gc_SetConfigData( )**. The CRN should be that of the consultation call. The application should call **gc_SetConfigData( )** with the correct hookflash value before calling **gc_DropCall( )** on the consultation call.

The **gc_SetConfigData( )** function uses a GC_PARM_BLK data structure that contains the configuration information. A new parmID, **GCPARM_CONSDROP_HKFLASH_OVERRIDE**, is used to set the single or double hookflash. As its name implies, this is a parameter to override the **CDP_AllowDblHookflashOnConsultationDrop** parameter in the CDP file. It does so only on a temporary basis and for a single consultation call. (See the Implementation Guidelines section below for further information about related parameters in the CDP file.)

The GC_PARM_BLK structure is populated using the **gc_util_insert_parm_val( )** function with the following values:

- **parm_blkpp** = pointer to the address of a valid GC_PARM_BLK structure where the parameter and value are to be inserted
- **setID** = GCSET_CALLINFO
- **parmID** = GCPARM_CONSDROP_HKFLASH_OVERRIDE

- **data_size** = sizeof(int)
- **data** = One of the following values:
  - GCPV_SINGLE_HKFLASH - single hookflash
  - GCPV_DBL_HKFLASH - double hookflash
  - GCPV_DISABLED - not set

Once the GC_PARM_BLK has been populated with the desired values, the **gc_SetConfigData( )** function can be issued to perform the configuration. The parameter values for the **gc_SetConfigData( )** function are as follows:

- **target_type** = GCTGT_CCLIB_CRN
- **target_id** = the call reference number (CRN) of the consultation call
- **target_datap** = pointer to the GC_PARM_BLK structure
- **time_out** = time-out in seconds
- **update_cond** = when to update (GCUPDATE_IMMEDIATE or GCUPDATE_ATNULL)
- **request_idp** = pointer to the location for storing the request ID
- **mode** = async or sync

For more detailed information about the **gc_SetConfigData( )** function parameters, see the *Dialogic® Global Call API Library Reference*.

The **gc_GetConfigData( )** function returns the value previously set by **gc_SetConfigData( )** on the same CRN. If no previous setting occurred for that CRN, GCPV_DISABLED is returned.

Implementation Guidelines

The following guidelines apply when implementing runtime control of single or double hookflash:

- This feature is only available on Dialogic® DM3 Boards using the United States T1 FXS/LS Bidirectional protocol.
- The **GCPARM_CONSDROP_HKFLASH_OVERRIDE** parameter setting via **gc_SetConfigData( )** does not take effect until a **gc_DropCall( )** on the consultation call CRN is invoked. The application must invoke the **gc_DropCall( )** with the appropriate CRN for the parameter to take effect (that is, single or double hookflash sent).
- In asynchronous mode, the application must update its state machine to wait for a success event on the **gc_SetConfigData( )** before a **gc_DropCall( )** on the consultation call is invoked.
- The **GCPARM_CONSDROP_HKFLASH_OVERRIDE** parameter has no effect on a CRN other than the consultation call CRN resulting from a successful **gc_SetupTransfer( )**.
- The setting of this parameter, and therefore the behavior for a drop on a consultation call, is not retained for subsequent calls on the same channel, unless explicitly set on each call.

The following guidelines discuss the use of the **GCPARM_CONSDROP_HKFLASH_OVERRIDE** parameter with regard to the related parameters in the *pdk_us_ls_fxs_io.cdp* file:

- The related parameters in the *pdk_us_ls_fxs_io.cdp* file are **CDP_AllowDblHookflashOnConsultationDrop** and **CDP_BypassHookflashOnConsultationDrop**. Both are disabled by default; the default behavior is that a single hookflash is sent when dropping a consultation call.
  - When **CDP_AllowDblHookflashOnConsultationDrop** is enabled, a double hookflash is sent when dropping a consultation call.
  - When **CDP_BypassHookflashOnConsultationDrop** is enabled, no hookflash is sent when dropping a consultation call.

  Note: Within the CDP file, the **CDP_BypassHookflashOnConsultationDrop** setting takes precedence over **CDP_AllowDblHookflashOnConsultationDrop**. But when **GCPARM_CONSDROP_HKFLASH_OVERRIDE** is set via **gc_SetConfigData( ),** its setting takes precedence over both of these CDP file parameters for the consultation call with the specified CRN.
- When **GCPARM_CONSDROP_HKFLASH_OVERRIDE** is set, the values of the CDP file parameters are not affected. However, the **GCPARM_CONSDROP_HKFLASH_OVERRIDE** parameter *overrides* the values of **CDP_AllowDblHookflashOnConsultationDrop** and **CDP_BypassHookflashOnConsultationDrop** for the consultation call with the specified CRN.
- If not set, the **GCPARM_CONSDROP_HKFLASH_OVERRIDE** parameter has no default (GCPV_DISABLED). Whatever is set at configuration time with the **CDP_AllowDblHookflashOnConsultationDrop** and **CDP_BypassHookflashOnConsultationDrop** parameters in the *pdk_us_ls_fxs_io.cdp* file will apply.

1.2.2 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the Dialogic® Global Call API, see the following documents:

- *Dialogic® Global Call API Programming Guide*
- *Dialogic® Global Call API Library Reference*

For features specific to E1 and T1 technology, see:

- *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*

For information about the United States T1 FXS/LS Bidirectional protocol and CDP file, see:

- *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*

1.3 Analog Call Transfer Support on Dialogic® Springware Boards

With the Service Update, blind and supervised analog call transfers using the Dialogic® Global Call API are now supported on Dialogic® Springware Boards.

Support for analog call transfer is applicable only to standard POTS ("plain old telephone service") line products. Proprietary private branch exchanges (PBXs) or key telephone systems (KTSs) and their related boards, such as the Dialogic® D/42JCT-U and D/82JCT-U PBX Integration Boards, are excluded from this feature since PBXs and KTSs may provide proprietary protocols for call transfers.

Aside from the Dialogic® PBX Integration Boards, this feature can be used with all other analog Springware boards that are supported in System Release 6.0 PCI Windows.

The following sections discuss:

- Call Transfer Overview
- Using Global Call with Analog Springware Boards
- Configuring the CDP File
- Documentation

1.3.1 Call Transfer Overview

There are two types of call transfers:

Supervised transfers
: The person transferring the call stays on the line, announces the call, and consults with the party to whom the call is being transferred before the transfer is completed.

Blind transfers
: The call is sent without any consultation or announcement by the person transferring the call. Blind transfers are also known as one-step or unsupervised transfers.

Supervised transfers use the following Global Call API functions:

gc_SetupTransfer( )
: Initiates a supervised transfer.

gc_CompleteTransfer( )
: Completes a supervised transfer.

gc_SwapHold( )
: Switches between the consultation call and the call pending transfer.

Blind transfers use the following Global Call API function:

gc_BlindTransfer( )
: Initiates and completes an unsupervised (one-step) transfer.

For further information about call transfers, see the Call State Models chapter of the *Dialogic® Global Call API Programming Guide*.

1.3.2 Using Global Call with Analog Springware Boards

In order to use Global Call with analog Springware Boards:

- The Global Call Protocols package must be installed.
- The boards must be using the North American Analog Bidirectional PDK protocol, pdk_na_an_io.

The Global Call Protocols package is one of the features that can be selected when installing System Release 6.0 PCI Windows. If you have already installed System Release 6.0 PCI Windows without the Global Call Protocols, you can add it by using the Modify or Change option as explained in the *Dialogic® System Release 6.0 PCI for Windows® Software Installation Guide*.

With Springware Boards, the protocol is assigned when a Global Call device is opened with the **gc_OpenEx( )** function. For information about using PDK protocols, see the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*.

1.3.3 Configuring the CDP File

To support blind and supervised call transfers, the country dependent parameters file for the North American Analog Bidirectional PDK protocol, *pdk_na_an_io.cdp*, has the following new parameters that can be set by the user:

CDP_BtStartTimer

Description: For a **supervised** transfer, specifies the maximum time that the protocol will wait after issuing hookflash as a part of **gc_SetupTransfer( )** and before the application issues **gc_MakeCall( )**.

For a **blind** transfer, specifies the maximum time that the protocol will wait after issuing hookflash as a part of **gc_BlindTransfer( )** and before the protocol completes the digit dial. Since the call is made within Global Call, this parameter can be used as a bail-out timer to dial tone detection when **CDP_Detect_DialTone** (Outbound) is enabled and none is detected during the elapsed time.

Values: Time in milliseconds. Default is 8000 (8 seconds).

CDP_BlindXferTime

Description: Specifies the delay time between the third party ringing and the controller going on-hook, i.e., disconnecting; it can be used to guard against network latencies, ensuring that the end-to-end audio path has been established before transfer.

Values: Time in milliseconds. Default is 2000 (2 seconds).

Other changes to the *pdk_na_an_io.cdp* file include:

SYS_FEATURES parameter now includes feature_transfer and feature_hold:
All CHARSTRING_t SYS_FEATURES = “feature_inbound,feature_outbound, feature_DNIS,feature_ANI,feature_transfer,feature_hold”

New analog pulse signal:
All CAS_SIGNAL_ANALOG_PULSE_t CAS_HOOKFLASH = Hookflash

1.3.4 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about generic Dialogic® Global Call API features, see the following documents:

- *Dialogic® Global Call API Programming Guide*
- *Dialogic® Global Call API Library Reference*

For features specific to Analog technology, see:

- *Dialogic® Global Call Analog Technology Guide*

For information about PDK protocols and CDP files, see:

- *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*

1.4 Support for Windows Vista® Operating System

With the Service Update, Dialogic® System Release 6.0 PCI for Windows® supports the Windows Vista® operating system. The following versions of Windows Vista are supported:

- Windows Vista Business, 32-bit edition
- Windows Vista Enterprise, 32-bit edition
- Windows Vista Ultimate, 32-bit edition

Note: The Dialogic® Software installation will be aborted if a 64-bit edition is detected.

When running System Release 6.0 PCI Windows on Windows Vista, only the following Dialogic® Springware Boards are supported:

Dialogic® Boards Supported with Windows Vista® Operating Systems			
PCI Boards		PCI Express Boards	
D/4PCI	D/300JCT-E1	D/4PCIE-4F-W	D/480JCT-2T1-EW
D/4PCIU	D/320JCT	D/4PCIE-4S-W	D/600JCT-1E1-75-EW
D/4PCIUF	D/480JCT-1T1	D/41JCT-LS-EW	D/600JCT-1E1-120-EW
D/41JCT-LS	D/480JCT-2T1	D/120JCT-LS-EW	D/600JCT-2E1-75-EW
D/120JCT-LS	D/600JCT-1E1	D/240JCT-T1-EW	D/600JCT-2E1-120-EW
D/160JCT	D/600JCT-2E1	D/300JCT-E1-EW	VFX/41JCT-LS-EW
D/240JCT-T1	VFX/41JCT-LS	D/480JCT-1T1-EW	
Note: Dialogic® DM3 Boards, SS7 Boards, and PBX Integration Boards are not currently supported with Windows Vista. Support for additional boards is planned for future Service Update releases.			

The following sections highlight some of the differences that users will see when running System Release 6.0 PCI Windows on Windows Vista, as opposed to running on other Windows® operating systems.

- Separate Install Media
- Content Split into Different Locations
- Changes to Environment Variables
- Building Applications
- Consent Dialog
- UDD Must Be Set to “Run as Administrator”
- WinHlp32.exe Not Included in Windows Vista
- RTF Logging
- Physical Address Extension

Separate Install Media

When you download System Release 6.0 PCI Windows from the Dialogic Support website, note that there is a separate link for downloading the System Release version for Windows Vista.

Content Split into Different Locations

In order to meet User Account Control and File Virtualization security features of Windows Vista, those files that may require user modification of some sort have now been moved to a non-restricted target folder. It is strongly advised to keep them in this or other equivalent target location; otherwise, user modification of these files may be disallowed by the operating system.

When installing System Release 6.0 PCI Windows on Windows Vista, you will be prompted to enter two directory locations (rather than one) for storing Dialogic® System Release Software files:

Choose Program File Destination Location
This is the directory where non-user-modifiable files in the System Release Software will be installed. The default location is *C:\Program Files\Dialogic*.

Choose User-Modifiable File Destination Location
This is the directory where user-modifiable files in the System Release Software will be installed. User-modifiable files include configuration files such as .prm and .cdp files, and demo programs. The default location is *C:\ProgramData\Dialogic*.

Note: *C:\ProgramData* is a hidden directory.

Changes to Environment Variables

Because the Dialogic® System Release Software files are now installed under two directory locations, it was necessary to make changes to the associated environment variables. Three new variables are being introduced to enable internal components to locate the non-modifiable *cfg* and *data* directories, and some of the existing variables point to a different location for a Windows Vista install:

Environment Variable	Old Default Value	Windows Vista Default Value
INTEL_DIALOGIC_DIR	C:\Program Files\Dialogic	Unchanged
INTEL_DIALOGIC_BIN	C:\Program Files\Dialogic\bin	Unchanged
INTEL_DIALOGIC_CFG	C:\Program Files\Dialogic\cfg	C:\ProgramData\Dialogic\cfg
INTEL_DIALOGIC_FWL	C:\Program Files\Dialogic\data	C:\ProgramData\Dialogic\data
INTEL_DIALOGIC_INC	C:\Program Files\Dialogic\inc	Unchanged
INTEL_DIALOGIC_LIB	C:\Program Files\Dialogic\lib	Unchanged
INTEL_DIALOGIC_QSCRIPT	C:\Program Files\Dialogic\qscript	Unchanged
DIALOGIC_CFG_INTERNAL	N/A	C:\Program Files\Dialogic\cfg
DIALOGIC_FWL_BIN	N/A	C:\Program Files\Dialogic\data
DIALOGIC_USERDATA_DIR	N/A	C:\ProgramData\Dialogic

Building Applications

Starting with this release, applications must adhere to a minimum software development environment. Validation has been achieved with Microsoft® Visual C++® versions that are part of the Visual Studio® .NET 2003 and the Visual Studio 2005. In future Service Updates, the Visual Studio .NET 2003 development environment support is planned to be dropped; thus it is recommended that developers use the Visual Studio 2005 environment instead.

Furthermore, Microsoft has dropped support of Visual Studio .NET 2002 or Visual Studio .NET 2003 on Windows Vista, and has released the Visual Studio 2005 Service Pack 1 update for Windows Vista. Please refer to the following MSDN® pages for more information:

- http://msdn2.microsoft.com/en-us/vstudio/aa948853.aspx
- http://msdn2.microsoft.com/en-us/vstudio/aa948854.aspx

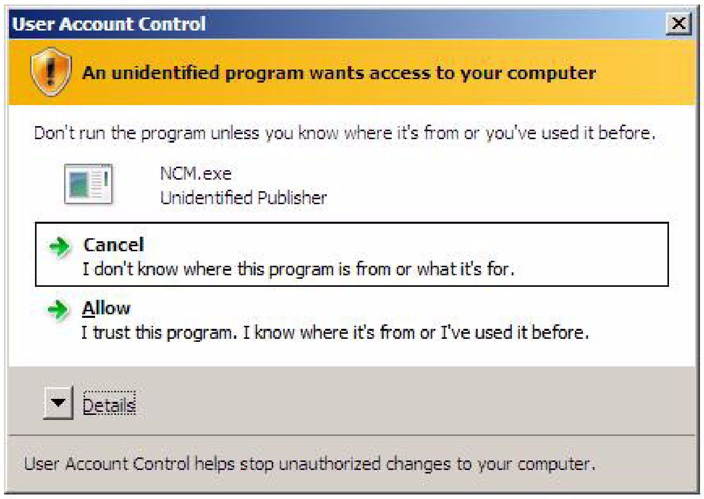
The following table elaborates on version numbers and restrictions. Any development environment prior to these is no longer supported.

Development Environment	Visual C++ Version	Dialogic Support
Visual Studio .NET 2003	Visual C++ Version 7.1	Restricted: Support is planned to be dropped in a future System Release 6.0 PCI Windows Service Update.
Visual Studio 2005	Visual C++ Version 8.0	No restrictions. Microsoft supported environment for Windows Vista.

Consent Dialog

In order to meet User Account Control restrictions in Windows Vista, most of the Dialogic® administration utilities have been adapted and embed a request for administration privileges from the user invoking them. This is also known as requiring “elevation” or requesting “administration tokens” from invoker. The user must belong to the administrator's group. The shield icon (shown below) will decorate the icon on any Dialogic® utility that requires elevation.

When the utility is invoked, it will request administration tokens, and if available, execution is granted. However, unless the user is the system’s (sole) administrator account (as opposed to belonging to the administrator group), a user consent dialog will still be displayed by the operating system, and the user must confirm before execution can be started. This is the way Windows Vista User Account Control works; this is not unique to Dialogic® Software under Windows Vista. An example of a consent dialog is:


User Account Control

An unidentified program wants access to your computer

Don't run the program unless you know where it's from or you've used it before.

NCM.exe
Unidentified Publisher

Cancel
I don't know where this program is from or what it's for.

Allow
I trust this program. I know where it's from or I've used it before.

Details

User Account Control helps stop unauthorized changes to your computer.

UDD Must Be Set to "Run as Administrator"

Dialogic® Diagnostics Software (UDD) works with Windows Vista; however, it lacks the embedded request for administration tokens from the invoker, so execution will fail due to this. In order to correct this problem, the user can manually set the appropriate rights. One way to do this is by right-clicking on the UDD application icon located in the INTEL_DIALOGIC_BIN directory, and in the Compatibility tab set it to "Run as Administrator."

WinHlp32.exe Not Included in Windows Vista

With the exception of the Installation Setup program, none of the Dialogic® GUI utilities, demos, etc., have appropriate functional help. These utilities still use Windows Help, which has been deprecated by Microsoft in favor of HTML help.

(Refer to the Microsoft document "The Windows Help (WinHlp32.exe) program is no longer included in Windows operating systems starting with Windows Vista" at http://support.microsoft.com/kb/917607/en-us.)

HTML help is not currently available for these Dialogic applications. As a workaround, Microsoft allows end users to install a package with the Windows Help program; users can download this package from:
http://www.microsoft.com/downloads/details.aspx?FamilyId=6EBCFAD9-D3F5-4365-8070-334CD175D4BB&displaylang=en

RTF Logging

With this release, Runtime Trace Facility (RTF) configuration and logging are only available for users who have administrator privileges. The RTF Manager will not run unless this condition is met.

In order for user applications to be able to generate RTF logs, the application must be run as an administrator. This can be accomplished either by right-clicking the application icon and selecting "Run As Administrator," or by applying a manifest to the application indicating that administrator rights are required.

Physical Address Extension

This release does not support systems running Windows with Physical Address Extension (PAE). The installation will warn when PAE is detected on the host machine. The user must disable PAE for proper Dialogic® System operation before any attempt to configure or start boards in the system.

1.5 Dialogic® DM3 Media Channel Reset Capability (Stuck Channel Recovery)

With the Service Update, whenever a media channel gets into a "stuck" state, there is a way to recover that channel without having to restart the application or redownload the board.

Note: A stuck channel is defined as a failure where the host application is unable to recover the channel and no further media operations are possible on that channel until the application is restarted or (in some cases) the board is redownloaded.

1.5.1 Feature Description

It has been observed, in rare occasions with high-density applications and high-load systems, that media channels have become stuck and no further processing would take place until the application is restarted or (in some cases) until the board is redownloaded.

This feature provides new API functions in the Dialogic® Voice library and in the Dialogic® Continuous Speech Processing (CSP) library that enable the application to recover from the stuck channel and return it to an idle and usable state.

Note: Not all stuck channels are recoverable. Also, not all errors are stuck channel errors. See Section 1.5.3, "Restrictions and Limitations", on page 53 for more information.

Supported Boards

All Dialogic® Media Span Boards support this media channel reset feature, namely Dialogic® DM/V, DM/V-A, DM/V-B, and DM/IP Boards.

New APIs

The two new API functions are:

- **dx_resetch( )** - Call this API to recover the media channel when the channel is stuck and in a recoverable state. If the channel is recovered, a TDX_RESET event is generated to the application, which enables the application to reuse the channel for more media functions. If the channel is not in a recoverable state, a TDX_RESETERR event is sent back to the application indicating that the specific channel is not recoverable.
- **ec_resetch( )** - Call this API to recover the CSP channel when the channel is stuck and in a recoverable state. If the channel is recovered, TDX_RESET and TEC_RESET events are generated to the application, which enables the application to reuse the channel for more media functions. If the channel is not in a recoverable state, TDX_RESETERR and TEC_RESETERR events are sent back to the application indicating that the specific channel is not recoverable. Note that the **ec_resetch( )** function resets both the voice and the CSP channels.

Function reference information is provided next.

dx_resetch( )

Name:	dx_resetch (chdev, mode)	
Inputs:	int chdev	• valid channel device handle
	int mode	• mode of operation
Returns:	0 if success -1 if failure	
Includes:	srllib.h dxxxlib.h	
Category:	I/O	
Mode:	asynchronous or synchronous	
Dialogic® Platform:	DM3	

Description

The **dx_resetch( )** function recovers a channel that is "stuck" (busy or hung) and in a recoverable state, and brings it to an idle and usable state. This function blocks all other functions from operating on the channel until the function completes.

Parameter	Description
chdev	Specifies the valid device handle obtained when the channel was opened using **dx_open( )**
mode	Specifies the mode of operation: • EV_ASYNC – asynchronous mode. The calling thread returns immediately so it can process media functionality on other channels. • EV_SYNC – synchronous mode. The calling thread waits until the channel is recovered or discovers that the channel is not in a recoverable state.

In synchronous mode, 0 is returned if the function completes successfully, and -1 is returned in case of error.

In asynchronous mode, the TDX_RESET event is generated to indicate that the channel was recovered and is in an idle and usable state. The TDX_RESETERR event is generated to indicate that the channel is not recoverable. Issuing any other media calls on this channel will result in an error.

Cautions

- The **dx_resetch( )** function is intended for use on channels that are stuck and not responding. Do **not** use it in place of **dx_stopch( )**. Use **dx_resetch( )** only if you do not receive an event within 30 seconds of when it's expected. Overuse of this function creates unnecessary overhead and may affect system performance.

■ Errors

If the function returns -1, use the Dialogic® Standard Runtime Library (SRL) Standard Attribute function **ATDV_LASTERR( )** to obtain the error code or use **ATDV_ERRMSGP( )** to obtain a descriptive error message. One of the following error codes may be returned:

EDX_BADPARM
: Invalid parameter

EDX_FWERROR
: Firmware error

EDX_NOERROR
: No error

■ Example

```
#include <srllib.h>
#include <dxxxlib.h>

main()
{
   int chdev, srlmode;
   /* Set SRL to run in polled mode. */
   srlmode = SR_POLLMODE;

   if (sr_setparm(SRL_DEVICE, SR_MODEID, (void *)&srlmode) == -1) {
   /* process error */
   }

   /* Open the channel using dx_open( ). Get channel device descriptor in
   * chdev.
   */

   if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
   /* process error */
   }

   /* continue processing */
   . .
   /* Force the channel to idle state. The I/O function that the channel
    * is executing will be terminated, and control passed to the handler
    * function previously enabled, using sr_enbhdlr(), for the
    * termination event corresponding to that I/O function.
    * In asynchronous mode, dx_stopch() returns immediately,
    * without waiting for the channel to go idle.
    */

   if ( dx_stopch(chdev, EV_ASYNC) == -1) {
   /* process error */
   }

   /* Wait for dx_stopch() to stop the channel and return the termination event
    * for the present media function.
    */

   /* After waiting for 30 secs if the termination event is not returned, issue a
    * dx_resetch() to reset the channel.
    */

   if (dx_resetch(chdev, EV_ASYNC) <0 )
   {
      /*process error */
   }

   /* Wait for TDX_RESET or TDX_RESETERR events */

}
```

See Also

- **ec_resetch( )** in the *Dialogic® Continuous Speech Processing API Library Reference*

ec_resetch( )

Name:	ec_resetch (chdev, mode)	
Inputs:	int chdev	• valid channel device handle
	int mode	• mode of operation
Returns:	0 if success -1 if failure	
Includes:	srllib.h eclib.h	
Category:	I/O	
Mode:	asynchronous or synchronous	
Dialogic® Platform:	DM3	

■ Description

The **ec_resetch( )** function recovers a channel that is "stuck" (busy or hung) and in a recoverable state, and brings it to an idle and usable state. This function blocks all other functions from operating on the channel until the function completes. This function recovers both the CSP channel and the voice channel.

Parameter	Description
chdev	Specifies the valid device handle obtained when the channel was opened using **dx_open( )**
mode	Specifies the mode of operation: • EV_ASYNC – asynchronous mode. The calling thread returns immediately so it can process media functionality on other channels. • EV_SYNC – synchronous mode. The calling thread waits until the channel is recovered or discovers that the channel is not in a recoverable state.

In synchronous mode, 0 is returned if the function completes successfully, and -1 is returned in case of error.

In asynchronous mode, the TDX_RESET and the TEC_RESET events are generated to indicate that the channel was recovered and is in an idle and usable state. The TDX_RESETERR and the TEC_RESETERR events are generated to indicate that the channel is not recoverable. Issuing any other media calls on this channel will result in an error.

■ Cautions

- The **ec_resetch( )** function is intended for use on channels that are stuck and not responding. Do **not** use it in place of **ec_stopch( )**. Use **ec_resetch( )** only if you do not receive an event within 30 seconds of when it's expected. Overuse of this function creates unnecessary overhead and may affect system performance.

■ Errors

If the function returns -1, use the Dialogic® Standard Runtime Library (SRL) Standard Attribute function **ATDV_LASTERR( )** to obtain the error code or use **ATDV_ERRMSGP( )** to obtain a descriptive error message. One of the following error codes may be returned:

EDX_BADPARM	Invalid parameter
EDX_FWERROR	Firmware error
EDX_NOERROR	No error

■ Example

```
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <eclib.h>
#include <errno.h> /* include in Linux applications only; exclude in Windows */

main()
{
   int chdev, srlmode;
   /* Set SRL to run in polled mode. */
   srlmode = SR_POLLMODE;
   if (sr_setparm(SRL_DEVICE, SR_MODEID, (void *)&srlmode) == -1) {
   /* process error */
   }

   /* Open the channel using dx_open( ). Get channel device descriptor
   * in chdev.
   */
   if ((chdev = dx_open("dxxxB1C1",0)) == -1) {
   /* process error */
   }
   /* continue processing */
   . .
   /* Force the channel to idle state. The I/O function that the channel
   * is executing will be terminated, and control passed to the handler
   * function previously enabled, using sr_enbhdlr(), for the
   * termination event corresponding to that I/O function.
   * In the asynchronous mode, ec_stopch() returns immediately,
   * without waiting for the channel to go idle.
   */
   if (ec_stopch(chdev, FULLDUPLEX, EV_ASYNC) == -1) {
   /* process error */
   }

   /* Wait for the termination events (TEC_STREAM and/or TDX_PLAY) */

   /* After waiting for 30 secs, if the channel is still in a busy state,
    * issue ec_resetch() to reset both the CSP channel and the voice channel.
    * When issued in asynchronous mode, it will return both (TEC_RESET/TEC_RESETERR)
    * and (TDX_RESET/TDX_RESETERR) events.
    */

   if (ec_resetch(chdev, EV_ASYNC)  == -1 ) {
      /* process error */
   }
   /* Wait for TEC_RESET/TEC_RESETERR and TDX_RESET/TDX_RESETERR */
}
```

■ See Also

- **dx_resetch( )** in the *Dialogic® Voice API Library Reference*

1.5.2 Implementation Guidelines

The following guidelines apply when implementing the media channel reset capability using the Dialogic® Voice API:

- It is recommended that you issue the function in asynchronous mode for more efficient processing. In synchronous mode, the calling thread is blocked until the function completes, which may take up to a minute in worst-case scenarios.
- The **dx_resetch( )** function is intended for use on channels that are stuck and not responding. Do **not** use it in place of **dx_stopch( )**. Use **dx_resetch( )** only if you do not receive an event within 30 seconds of when it's expected. Overuse of this function creates unnecessary overhead and may affect system performance.
- If you call **dx_resetch( )** immediately following **dx_stopch( )** without waiting at least 30 seconds for **dx_stopch( )** to complete, you will not receive events, such as TDX_PLAY and TDX_RECORD, even if the stop operation is successful and the channel was not stuck. Instead, you will only receive the TDX_RESET event if the channel recovery is successful or the TDX_RESETERR event if the channel is not recoverable.
- If you call **dx_resetch( )** without first using **dx_stopch( )** to stop the channel, the Voice library will internally call **dx_stopch( )** and wait 30 seconds for it to complete. If the internal stop channel is successful, you will receive the TDX_RESET event only. If the internal stop channel is unsuccessful, the Voice library will then call **dx_resetch( )**. Once a reset is attempted, you will receive the TDX_RESET event if the channel recovery is successful or the TDX_RESETERR event if the channel is not recoverable.
- Unrecoverable channels are written to a log file in the DebugAngel tool or the Runtime Trace Facility (RTF) tool. See the *Dialogic® System Software Diagnostics Guide* for more information on these tools.

The following guidelines apply when implementing the media channel reset capability using the Dialogic® Continuous Speech Processing (CSP) API:

- The guidelines described for **dx_resetch( )** and **dx_stopch( )** apply to the **ec_resetch( )** and **ec_stopch( )** functions in the CSP API.
- For CSP applications, it is recommended that you use **ec_resetch( )** since this function resets both the voice and the CSP channels. The **dx_resetch( )** function resets the voice channels only.

1.5.3 Restrictions and Limitations

The following restrictions and limitations apply to the media channel reset feature:

- This feature only addresses scenarios where the firmware and the host library have lost synchronization or an event has not been propagated. DSP crashes, catastrophic firmware failures (killtasks), or unsynchronized firmware state machines are **not** recoverable without redownload of the board.
- This feature only addresses channels that become stuck while performing play and record, tone generation, or FSK operations. It also addresses channels that become stuck during CSP play or record operations.

- This feature does **not** address reset of IP media channels on Dialogic® DM/IP Boards. It only addresses the reset of voice channels on DM/IP Boards.
- The reset may not succeed if CPU utilization on the host system is close to 100 percent. It is recommended that the CPU usage be at a reasonable level (less than 70 percent) before you attempt a channel reset.

1.5.4 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the Dialogic® Voice API, see the following documents:

- *Dialogic® Voice API Programming Guide*
- *Dialogic® Voice API Library Reference*

For more information about the Dialogic® Continuous Speech Processing (CSP) API, see the following documents:

- *Dialogic® Continuous Speech Processing API Programming Guide*
- *Dialogic® Continuous Speech Processing API Library Reference*

1.6 AMD Opteron Server Support

With the Service Update, Dialogic® System Release 6.0 PCI for Windows® has been validated for use with Advanced Micro Devices, Inc. (AMD) Opteron server processors.

1.7 Global DPD Enabled on Dialogic® Springware Boards

With the Service Update, Global Dial Pulse Detection (DPD) is now available by default via software. Previously, this feature had to be enabled from the factory or by ordering a separate GDPD enablement package to enable DPD on a board.

Global DPD is supported on Dialogic® Springware Boards, such as Dialogic® JCT Media Boards and Dialogic® D/4PCIU Media Boards. Global DPD is **not** supported on Dialogic® D/42JCT and D/82JCT PBX Integration Boards or on Dialogic® DM3 Media Boards.

For information about implementing Global DPD, see the *Dialogic® Voice API Programming Guide*.

1.8 Enhanced Special Information Tones on Dialogic® DM3 Boards Using Voice and Global Call APIs

With the Service Update, the user has the ability to detect new custom special information tones (SITs) on Dialogic® DM3 Boards using the Dialogic® Voice API and Dialogic® Global Call API. The new custom SITs are detected via the regular API events for detecting call progress analysis outcome, and in particular, SIT tone detection. In addition, create, query, and modify support of these new SIT tones and three existing default tones via the Voice API is now available.

1.8.1 Feature Description

Predictive dialing applications, which are widely used in call centers, need to detect a variety of SITs being used by Service Providers around the world. For this feature, the Voice and Global Call APIs each provide an additional 15 SITs with customizable SIT templates, which allow the user to detect a variety of nonstandard SITs used by Service Providers. When the board firmware detects an incoming SIT tone during call progress analysis, it tries to match it to one of the existing (default) templates. Tones that do not match the default templates will be matched against the custom SIT templates created by the user, and reported as such. If the SIT still does not fall into any of those two categories, custom or standard, it may still be collected and reported as undetected (SIT_ANY), and also reported back.

Voice API

For the Voice API, the 15 new custom tone templates (plus the existing default tone templates) are supported for detection and reporting by the **ATDX_CRTNID( )** function. (For a description of the **ATDX_CRTNID( )** function support on DM3 Boards, see Section 1.59, "Enhanced Special Information Tone Frequency Detection on Dialogic® DM3 Boards", on page 229.)

For this feature, full create, query, and modify support has been added for the new custom SITs and for three of the existing default SITs, via the **dx_createtone( )**, **dx_querytone( )**, and **dx_deletetone( )** functions, as follows:

```
#define TID_CUSTOM_SIT1
#define TID_CUSTOM_SIT2
#define TID_CUSTOM_SIT3
#define TID_CUSTOM_SIT4
#define TID_CUSTOM_SIT5
#define TID_CUSTOM_SIT6
#define TID_CUSTOM_SIT7
#define TID_CUSTOM_SIT8
#define TID_CUSTOM_SIT9
#define TID_CUSTOM_SIT10
#define TID_CUSTOM_SIT11
#define TID_CUSTOM_SIT12
#define TID_CUSTOM_SIT13
#define TID_CUSTOM_SIT14
#define TID_CUSTOM_SIT15
```

```
#define TID_SIT_NC_INTERLATA
#define TID_SIT_RO_INTERLATA
#define TID_SIT_IO
```

For more information on modifying tone definitions, see the *Dialogic® Voice API Programming Guide*.

Global Call API

For the Global Call API, 15 new custom SITs are allowed and are reported to the application via the GCEV_DISCONNECTED event once any one of them is detected via Global Call. The following table maps the custom SIT tone ID to the Global Call values:

Global Call Result Value	Tone ID	Description
GCRV_SIT_UNKNOWN (GCRV_RESULT | 0x70)	0x38F	Custom SIT tone 1 detected
	0x390	Custom SIT tone 2 detected
	0x391	Custom SIT tone 3 detected
	0x392	Custom SIT tone 4 detected
	0x393	Custom SIT tone 5 detected
	0x394	Custom SIT tone 6 detected
	0x395	Custom SIT tone 7 detected
	0x396	Custom SIT tone 8 detected
	0x397	Custom SIT tone 9 detected
	0x398	Custom SIT tone 10 detected
	0x399	Custom SIT tone 11 detected
	0x39A	Custom SIT tone 12 detected
	0x39B	Custom SIT tone 13 detected
	0x39C	Custom SIT tone 14 detected
	0x39D	Custom SIT tone 15 detected

In addition, four new default SITs can be detected via Global Call. The following table maps the Voice SITs to the new Global Call values:

Voice SIT	Global Call Result Value	Value	Global Call Error Code	Value	Description
TID_SIT_ANY	GCRV_SIT_UNKNOWN	(GCRV_RESULT | 0x70)	EGC_SIT_ UNKNOWN	0x162	Unknown SIT detected
TID_SIT_NC_INTERLATA	GCRV_NO_CIRCUIT_ INTERLATA	(GCRV_RESULT | 0x71)	EGC_NO_ CIRCUIT_ INTERLATA	0x163	No circuit interlata SIT detected

Voice SIT	Global Call Result Value	Value	Global Call Error Code	Value	Description
TID_SIT_RO_INTERLATA	GCRV_REORDER_ INTERLATA	(GCRV_RESULT | 0x72)	EGC_REORDER _INTERLATA	0x164	Reorder interlata SIT detected
TID_SIT_IO	GCRV_INEFFECTIVE_ OTHER	(GCRV_RESULT | 0x73)	EGC_INEFFECTI VE_OTHER	0x165	Ineffective other SIT detected

1.8.2 Supported Boards

The following boards support this feature:

- Dialogic® DMV-B Media Boards
- Dialogic® DMV300BTEPEQ, DMV600BTEPEQ, and DMV1200BTEPEQ Media Boards
- Dialogic® DISI Switching Boards
- Dialogic® DMV160LP Media Boards
- Dialogic® DMV and DMV-A Media Boards (ISDN or resource)

Note: DMV and DMV-A Media Boards running CAS, PDK (R2MF), and clear channel (ts16) do **not** support this feature. Dialogic® DM/IP, HDSI, VFN, DM3 Fax, and CPI Fax Boards do **not** support this feature. Refer to the table at the end of this section, PCD Files That Do Not Support Enhanced Special Information Tones Feature, for a list of PCD files that are **excluded** from this feature. If you attempt to use this feature with a board using one of these PCD files, an error code is returned.

1.8.3 Example

The **dx_createtone( )** function creates a new tone definition for a specific call progress tone. On successful completion of the function, the TONE_DATA structure is used to create a tone definition for the specified call progress tone.

Prior to creating a new tone definition with **dx_createtone( )**, use **dx_querytone( )** to get tone information for that tone, then use **dx_deletetone( )** to delete that tone. The custom SIT tone templates have empty on-board firmware definitions after board initialization.

The following is a code example for the TONE_DATA data structure using TID_CUSTOM_SIT1 tone ID. The TONE_DATA structure is defined in *dxxxlib.h*.

```
#include "srllib.h"
#include "dxxxlib.h"

main()
{
     int brdhdl; /* physical board device handle */
     .
     .
     .
     /* Open physical board */
     if ((brdhdl = dx_open("brdB1",0)) == -1) {
         printf("Cannot open board\n");
         /* Perform system error processing */
         exit(1);
     }
     /* Get the Tone Information for the TID_CUSTOM_SIT1 tone*/
     int result;
     TONE_DATA tonedata;
     if ((result = dx_querytone(brdhdl, TID_CUSTOM_SIT1, &tonedata, EV_ASYNC)) == -1) {
        printf("Cannot obtain tone information for TID_CUSTOM_SIT1 \n");
        /* Perform system error processing */
        exit(1);
     }

     /* Delete the current TID_CUSTOM_SIT1 call progress tone before creating a new definition*/
     if ((result = dx_deletetone(brdhdl, TID_CUSTOM_SIT1, EV_ASYNC)) == -1) {
        printf("Cannot delete the TID_CUSTOM_SIT1 tone\n");
        /* Perform system error processing */
        exit(1);
}

     /* Change call progress default CUSTOM SIT tone */

     tonedata.numofseg   = 3; /* triple segment tone */
     tonedata.tn_rep_cnt = 1;
     tonedata.toneseg[0].tn_dflag   = 0;
     tonedata.toneseg[0].tn1_min    = 874;
     tonedata.toneseg[0].tn1_max    = 955;
     tonedata.toneseg[0].tn2_min    = 0;
     tonedata.toneseg[0].tn2_max    = 0;
     tonedata.toneseg[0].tn_twinmin = 0;
     tonedata.toneseg[0].tn_twinmax = 0;
     tonedata.toneseg[0].tnon_min   = 15;
     tonedata.toneseg[0].tnon_max   = 30;
     tonedata.toneseg[0].tnoff_min  = 0;
     tonedata.toneseg[0].tnoff_max  = 5;

     tonedata.toneseg[1].tn_dflag   = 0;
     tonedata.toneseg[1].tn1_min    = 1310;
     tonedata.toneseg[1].tn1_max    = 1410;
     tonedata.toneseg[1].tn2_min    = 0;
     tonedata.toneseg[1].tn2_max    = 0;
     tonedata.toneseg[1].tn_twinmin = 0;
     tonedata.toneseg[1].tn_twinmax = 0;
     tonedata.toneseg[1].tnon_min   = 15;
     tonedata.toneseg[1].tnon_max   = 30;
     tonedata.toneseg[1].tnoff_min  = 0;
     tonedata.toneseg[1].tnoff_max  = 5;

     tonedata.toneseg[2].tn_dflag   = 0;
     tonedata.toneseg[2].tn1_min    = 1845;
     tonedata.toneseg[2].tn1_max    = 1950;
     tonedata.toneseg[2].tn2_min    = 0;
     tonedata.toneseg[2].tn2_max    = 0;
     tonedata.toneseg[2].tn_twinmin = 0;
     tonedata.toneseg[2].tn_twinmax = 0;
     tonedata.toneseg[2].tnon_min   = 0;
     tonedata.toneseg[2].tnon_max   = 0;
     tonedata.toneseg[2].tnoff_min  = 0;
     if ((result = dx_createtone(brdhdl, TID_CUSTOM_SIT1, &tonedata, EV_SYNC)) == -1) {
        printf("create tone for TID_CUSTOM_SIT1 failed\n");
        /* Perform system error processing */
        exit(1);
}
```

1.8.4 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the Dialogic® Voice API, see the following documents:

- *Dialogic® Voice API Programming Guide*
- *Dialogic® Voice API Library Reference*

For more information about the Dialogic® Global Call API, see the following documents:

- *Dialogic® Global Call API Programming Guide*
- *Dialogic® Global Call API Library Reference*

Note: The following table lists PCD files that are **excluded** from this feature. If you attempt to create, query, or delete any of the custom tones with a board using one of these PCD files, error code EDX_TNQUERYDELETE is returned. Detection of the custom tones will not work either. The Description column in the table reflects the same text displayed in the Assign Firmware File dialog box when using the procedure described in Section 4.4, Selecting a Configuration File Set, in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*. For most products, the file names of the configuration file set reflect the media load supported. If a media load number (ml*x*) is not present in the file name, no media load is supported for that configuration. See Section 2.4.3, Media Load Configuration File Sets, in the DM3 Configuration Guide for details on each of the board families.

PCD Files That Do Not Support Enhanced Special Information Tones Feature					
PCD File (Unsupported)	**Description**	**PCD File (Unsupported)**	**Description**	**PCD File (Unsupported)**	**Description**
4x2_cas.pcd	DMV480_4T1 (BV 48 channels PSTN 4 Trunks CAS protocol)	ipvs_evr_2cas_311.pcd	DM/IP481-2T1-100BT (DM/IPLINK-T1_NIC, 1 daughterboard, 48 channels)	ml1b_qsa_cas.pcd	DMV960_4T1 (BV+ADSI/FSK 96 channels PSTN 4 Trunks CAS protocol)
4x2_r2mf.pcd	DMV600_4E1 (BV 60 channels PSTN 4 Trunks R2MF protocol)	ipvs_evr_2cas_311c.pcd	DM/IP481-2T1-100BT (DM/IPLINK-T1_NIC, 1 daughterboard, 48 channels)	ml1b_qsa_r2mf.pcd	DMV1200_4E1 (BV+ADSI/FSK 120 channels PSTN 4 Trunks R2MF protocol)
4xt_cas.pcd	DMT960_4T1 (PSTN 4 Trunks CAS protocol)	ipvs_evr_2cas_ml11_311c.pcd	DM/IP481-2T1-100BT (DM/IPLINK-T1_NIC, 1 daughterboard, 48 channels)	ml1b_qsa_ts16.pcd	DMV1200_4E1 (BV+ADSI/FSK 120 channels PSTN 4 Trunks TS16 protocol)
4xt_r2mf.pcd	DMT1200_4E1 (PSTN 4 Trunks R2MF protocol)	ipvs_evr_2isdn_4ess_311.pcd	DM/IP481-2T1 (DM/IPLINK-T1_NIC, 1 daughterboard, 48 channels)	ml2_dsa_cas.pcd	DMV480A_2T1 (BV+CSP 48 channels PSTN 2 Trunks CAS protocol)
Note: List of PCD files that do not support enhanced special information tones is subject to change with each Service Update.					

PCD Files That Do Not Support Enhanced Special Information Tones Feature					
PCD File (Unsupported)	Description	PCD File (Unsupported)	Description	PCD File (Unsupported)	Description
at_hdsi.pcd	HDSI	ipvs_evr_2isdn_4ess_311c.pcd	DM/IP481-2T1-100BT (DM/IPLINK-2T1_NIC, 1 daughterboard, 48 channels)	ml2_dsa_r2mf.pcd	DMV600A_2E1 (BV+CSP 60 channels PSTN 2 Trunks R2MF protocol)
at_hdsi_48_play_rec.pcd	HDSI	ipvs_evr_2isdn_4ess_ml11_311c.pcd	DM/IP481-2T1-100BT (DM/IPLINK-2T1_NIC, 1 daughterboard, 48 channels)	ml2_dsa_ts16.pcd	DMV600A_2E1 (BV+CSP 60 channels PSTN 2 Trunks TS16 protocol)
at_hdsi_72_play_rec.pcd	HDSI	ipvs_evr_2isdn_5ess_311.pcd	DM/IP481-2T1 (DM/IPLINK-T1_NIC, 1 daughterboard, 48 channels)	ml2_qsa_cas.pcd	DMV960_4T1 (BV+CSP 96 channels PSTN 4 Trunks CAS protocol)
at_hdsi_96_play_rec.pcd	HDSI	ipvs_evr_2isdn_5ess_311c.pcd	DM/IP481-2T1-100BT (DM/IPLINK-2T1_NIC, 1 daughterboard, 48 channels)	ml2_qsa_r2mf.pcd	DMV1200_4E1 (BV+CSP 120 channels PSTN 4 Trunks R2MF protocol)
au_hdsi.pcd	HDSI	ipvs_evr_2isdn_5ess_ml11_311c.pcd	DM/IP481-2T1-100BT (DM/IPLINK-2T1_NIC, 1 daughterboard, 48 channels)	ml2_qsa_ts16.pcd	DMV1200_4E1 (BV+CSP 120 channels PSTN 4 Trunks TS16 protocol)
au_hdsi_48_play_rec.pcd	HDSI	ipvs_evr_2isdn_dms_311.pcd	DM/IP481-2T1 (DM/IPLINK-T1_NIC, 1 daughterboard, 48 channels)	ml5bc_dsa_cas.pcd	DMV480A_2T1 (BV+CSP+64EC+CSPtoCTBus 48 channels Fax 12 channels PSTN 2 Trunks CAS protocol)
au_hdsi_72_play_rec.pcd	HDSI	ipvs_evr_2isdn_dms_311c.pcd	DM/IP481-2T1-100BT (DM/IPLINK-2T1_NIC, 1 daughterboard, 48 channels)	ml5bc_dsa_r2mf.pcd	DMV600A_2E1 (BV+CSP+64EC+CSPtoCTBus 60 channels Fax 12 channels PSTN 2 Trunks R2MF protocol)
au_hdsi_96_play_rec.pcd	HDSI	ipvs_evr_2isdn_dms_ml11_311c.pcd	DM/IP481-2T1-100BT (DM/IPLINK-2T1_NIC, 1 daughterboard, 48 channels)	ml5bc_dsa_ts16.pcd	DMV600A_2E1 (BV+CSP+64EC+CSPtoCTBus 60 channels Fax 12 channels PSTN 2 Trunks TS16 protocol)
be_hdsi.pcd	HDSI	ipvs_evr_2isdn_net5_311.pcd	DM/IP601-2E1 (DM/IPLINK-E1_NIC, 1 daughterboard, 60 channels)	mn_4xt_cas.pcd	DMT960_4T1 (PSTN 4 Trunks CAS protocol)
Note: List of PCD files that do not support enhanced special information tones is subject to change with each Service Update.					

PCD Files That Do Not Support Enhanced Special Information Tones Feature					
PCD File (Unsupported)	**Description**	**PCD File (Unsupported)**	**Description**	**PCD File (Unsupported)**	**Description**
be_hdsi_48_play_rec.pcd	HDSI	ipvs_evr_2isdn_net5_311c.pcd	DM/IP601-2E1 (DM/IPLINK-E1_NIC, 1 daughterboard, 60 channels)	mn_4xt_r2mf.pcd	DMT1200_4E1 (PSTN 4 Trunks R2MF protocol)
be_hdsi_72_play_rec.pcd	HDSI	ipvs_evr_2isdn_net5_ml11_311c.pcd	DM/IP601-2E1-100BT (DM/IPLINK-E1_NIC, 1 daughterboard, 60 channels)	mn_4xt_ts16.pcd	DMT1200_4E1 (PSTN 4 Trunks TS16 protocol)
be_hdsi_96_play_rec.pcd	HDSI	ipvs_evr_2isdn_net5_ts16_311.pcd	DM/IP601-2E1 (DM/IPLINK-E1_NIC, 1 daughterboard, 60 channels)	mx_hdsi.pcd	HDSI
ch_hdsi.pcd	HDSI	ipvs_evr_2isdn_ni2_311.pcd	DM/IP481-2T1 (DM/IPLINK-T1_NIC, 1 daughterboard, 48 channels)	mx_hdsi_48_play_rec.pcd	HDSI
ch_hdsi_48_play_rec.pcd	HDSI	ipvs_evr_2isdn_ni2_311c.pcd	DM/IP481-2T1-100BT (DM/IPLINK-2T1_NIC, 1 daughterboard, 48 channels)	mx_hdsi_72_play_rec.pcd	HDSI
ch_hdsi_72_play_rec.pcd	HDSI	ipvs_evr_2isdn_ni2_ml11_311c.pcd	DM/IP481-2T1-100BT (DM/IPLINK-2T1_NIC, 1 daughterboard, 48 channels)	mx_hdsi_96_play_rec.pcd	HDSI
ch_hdsi_96_play_rec.pcd	HDSI	ipvs_evr_2isdn_ntt_311.pcd	DM/IP481-2T1 (DM/IPLINK-T1_NIC, 1 daughterboard, 48 channels)	my_hdsi.pcd	HDSI
cpi400bripcipmp.pcd	CPi/400 BRI-PCI Point to Multipoint	ipvs_evr_2isdn_ntt_311c.pcd	DM/IP481-2T1-100BT (DM/IPLINK-2T1_NIC, 1 daughterboard, 48 channels)	my_hdsi_48_play_rec.pcd	HDSI
cpi400bripcipp.pcd	CPi/400 BRI-PCI Point to Point	ipvs_evr_2isdn_ntt_ml11_311c.pcd	DM/IP481-2T1-100BT (DM/IPLINK-2T1_NIC, 1 daughterboard, 48 channels)	my_hdsi_72_play_rec.pcd	HDSI
de_hdsi.pcd	HDSI	ipvs_evr_2isdn_qsige1_311.pcd	DM/IP601-2E1 (DM/IPLINK-E1_NIC, 1 daughterboard, 60 channels)	my_hdsi_96_play_rec.pcd	HDSI
de_hdsi_48_play_rec.pcd	HDSI	ipvs_evr_2isdn_qsige1_311c.pcd	DM/IP601-2E1 (DM/IPLINK-E1_NIC, 1 daughterboard, 60 channels)	nl_hdsi.pcd	HDSI
Note: List of PCD files that do not support enhanced special information tones is subject to change with each Service Update.					

PCD Files That Do Not Support Enhanced Special Information Tones Feature					
PCD File (Unsupported)	Description	PCD File (Unsupported)	Description	PCD File (Unsupported)	Description
de_hdsi_72_play_rec.pcd	HDSI	ipvs_evr_2isdn_qsige1_ml11_311c.pcd	DM/IP601-2E1-100BT (DM/IPLINK-E1_NIC, 1 daughterboard, 60 channels)	nl_hdsi_48_play_rec.pcd	HDSI
de_hdsi_96_play_rec.pcd	HDSI	ipvs_evr_2isdn_qsigt1_311.pcd	DM/IP481-2T1 (DM/IPLINK-T1_NIC, 1 daughterboard, 48 channels)	nl_hdsi_72_play_rec.pcd	HDSI
dk_hdsi.pcd	HDSI	ipvs_evr_2isdn_qsigt1_311c.pcd	DM/IP481-2T1-100BT (DM/IPLINK-2T1_NIC, 1 daughterboard, 48 channels)	nl_hdsi_96_play_rec.pcd	HDSI
dk_hdsi_48_play_rec.pcd	HDSI	ipvs_evr_2isdn_qsigt1_ml11_311c.pcd	DM/IP481-2T1-100BT (DM/IPLINK-2T1_NIC, 1 daughterboard, 48 channels)	no_hdsi.pcd	HDSI
dk_hdsi_72_play_rec.pcd	HDSI	ipvs_evr_2r2mf_311.pcd	DM/IP601-2E1-100BT	no_hdsi_48_play_rec.pcd	HDSI
dk_hdsi_96_play_rec.pcd	HDSI	ipvs_evr_2r2mf_311c.pcd	DM/IP601-2E1-100BT (DM/IPLINK-E1_NIC)	no_hdsi_72_play_rec.pcd	HDSI
es_hdsi.pcd	HDSI	ipvs_evr_2r2mf_ml11_311c.pcd	DM/IP601-2E1-100BT (DM/IPLINK-E1_NIC, 1 daughterboard, 60 channels)	no_hdsi_96_play_rec.pcd	HDSI
es_hdsi_48_play_rec.pcd	HDSI	ipvs_evr_cas_311.pcd	DM/IP241-T1 (DM/IPLINK-T1_NIC)	nz_hdsi.pcd	HDSI
es_hdsi_72_play_rec.pcd	HDSI	ipvs_evr_cas_ml11_311.pcd	DM/IP241-T1 (DM/IPLINK-T1_NIC)	nz_hdsi_48_play_rec.pcd	HDSI
es_hdsi_96_play_rec.pcd	HDSI	ipvs_evr_isdn_4ess_311.pcd	DM/IP241-1T1-100BT (DM/IPLINK-T1_NIC, 1 daughterboard, 24 channels)	nz_hdsi_72_play_rec.pcd	HDSI
fax24.pcd	DM/F240-PCI - 24 Ch. Fax	ipvs_evr_isdn_4ess_ml11_311.pcd	DM/IP241-1T1-100BT (DM/IPLINK-T1_NIC, 1 daughterboard, 24 channels)	nz_hdsi_96_play_rec.pcd	HDSI
fax30.pcd	DM/F300-PCI - 30 Ch. Fax	ipvs_evr_isdn_5ess_311.pcd	DM/IP241-1T1-100BT (DM/IPLINK-T1_NIC, 1 daughterboard, 24 channels)	pt_hdsi.pcd	HDSI
fn_isdn_4ess.pcd	DM/F240-1T1-PCI - T1 w/24 Ch. Fax FN	ipvs_evr_isdn_5ess_ml11_311.pcd	DM/IP241-1T1-100BT (DM/IPLINK-T1_NIC, 1 daughterboard, 24 channels)	pt_hdsi_48_play_rec.pcd	HDSI
Note: List of PCD files that do not support enhanced special information tones is subject to change with each Service Update.					

PCD Files That Do Not Support Enhanced Special Information Tones Feature					
PCD File (Unsupported)	**Description**	**PCD File (Unsupported)**	**Description**	**PCD File (Unsupported)**	**Description**
fn_isdn_5ess.pcd	DM/F240-1T1-PCI - T1 w/24 Ch. Fax FN	ipvs_evr_isdn_dms_311.pcd	DM/IP241-1T1 (DM/IPLINK-T1_NIC, 1 daughterboard, 24 channels)	pt_hdsi_72_play_rec.pcd	HDSI
fn_isdn_dms.pcd	DM/F240-1T1-PCI - T1 w/24 Ch. Fax FN	ipvs_evr_isdn_dms_ml11_311.pcd	DM/IP241-1T1 (DM/IPLINK-T1_NIC, 1 daughterboard, 24 channels)	pt_hdsi_96_play_rec.pcd	HDSI
fn_isdn_net5.pcd	DM/F300-1E1-PCI - E1 w/30 Ch. Fax FN	ipvs_evr_isdn_net5_311.pcd	DM/IP301-1E1-100BT (DM/IPLINK-E1_NIC, 1 daughterboard, 30 channels)	qs_cas.pcd	DMV960_4T1 (BV 96 channels PSTN 4 Trunks CAS protocol)
fn_isdn_ni2.pcd	DM/F240-1T1-PCI - T1 w/24 Ch. Fax FN	ipvs_evr_isdn_net5_ml11_311.pcd	DM/IP301-1E1-100BT (DM/IPLINK-E1_NIC, 1 daughterboard, 30 channels)	qs_r2mf.pcd	DMV1200_4E1 (BV 120 channels PSTN 4 Trunks R2MF protocol)
fn_isdn_ntt.pcd	DM/F240-1T1-PCI - T1 w/24 Ch. Fax FN	ipvs_evr_isdn_ni2_311.pcd	DM/IP241-1T1-100BT (DM/IPLINK-T1_NIC, 1 daughterboard, 24 channels)	se_hdsi.pcd	HDSI
fn_isdn_qsige1.pcd	DM/F300-1E1-PCI - E1 w/30 Ch. Fax FN	ipvs_evr_isdn_ni2_ml11_311.pcd	DM/IP241-1T1-100BT (DM/IPLINK-T1_NIC, 1 daughterboard, 24 channels)	se_hdsi_48_play_rec.pcd	HDSI
fn_isdn_qsigt1.pcd	DM/F240-1T1-PCI - T1 w/23 Ch. Fax FN	ipvs_evr_isdn_ntt_311.pcd	DM/IP241-1T1-100BT (DM/IPLINK-T1_NIC, 1 daughterboard, 23 channels)	se_hdsi_72_play_rec.pcd	HDSI
fn_r2mf.pcd	DM/F300-1E1-PCI - E1 w/30 Fax Channels FN	ipvs_evr_isdn_ntt_ml11_311.pcd	DM/IP241-1T1-100BT (DM/IPLINK-T1_NIC, 1 daughterboard, 23 channels)	se_hdsi_96_play_rec.pcd	HDSI
fn_t1.pcd	DM/F240-1T1-PCI - T1 w/24 Ch. Fax FN	ipvs_evr_isdn_qsige1_311.pcd	DM/IP301-1E1-100BT (DM/IPLINK-E1_NIC, 1 daughterboard, 30 channels)	sg_hdsi.pcd	HDSI
fn3_isdn_4ess.pcd	DM/F240-1T1-PCI - T1 w/24 Ch. Fax FN3	ipvs_evr_isdn_qsige1_ml11_311.pcd	DM/IP301-1E1-100BT (DM/IPLINK-E1_NIC, 1 daughterboard, 30 channels)	sg_hdsi_48_play_rec.pcd	HDSI
fn3_isdn_5ess.pcd	DM/F240-1T1-PCI - T1 w/24 Ch. Fax FN3	ipvs_evr_isdn_qsigt1_311.pcd	DM/IP241-1T1-100BT (DM/IPLINK-T1_NIC, 1 daughterboard, 23 channels)	sg_hdsi_72_play_rec.pcd	HDSI
Note: List of PCD files that do not support enhanced special information tones is subject to change with each Service Update.					

PCD Files That Do Not Support Enhanced Special Information Tones Feature					
PCD File (Unsupported)	**Description**	**PCD File (Unsupported)**	**Description**	**PCD File (Unsupported)**	**Description**
fn3_isdn_dms.pcd	DM/F240-1T1-PCI - T1 w/24 Ch. Fax FN3	ipvs_evr_isdn_qsigt1_ml11_311.pcd	DM/IP241-1T1-100BT (DM/IPLINK-T1_NIC, 1 daughterboard, 23 channels)	sg_hdsi_96_play_rec.pcd	HDSI
fn3_isdn_net5.pcd	DM/F300-1E1-PCI - E1 w/30 Ch. Fax FN3	ipvs_evr_r_311.pcd	DM/IP601-2E1 (DM/IPLINK-E1_NIC, 1 daughterboard, 60 channels)	ul2_qsa_cas.pcd	DMV960_4T1 (BV+CSP 96 channels CONF+EC 15 channels Fax 4 channels PSTN 4 Trunks CAS protocol)
fn3_isdn_ni2.pcd	DM/F240-1T1-PCI - T1 w/24 Ch. Fax FN3	ipvs_evr_r_311_ml1a.pcd	DM/IP601-2E1 (DM/IPLINK-E1_NIC, 1 daughterboard, 60 channels)	ul1_qsa_*e1ISDNprot*.pcd	DMV1200_4E1 (BV 60 channels CONF+EC 60 channels Fax 8 channels PSTN 4 Trunks DPNSS protocol)
fn3_isdn_ntt.pcd	DM/F240-1T1-PCI - T1 w/24 Ch. Fax FN3	ipvs_evr_r_311c.pcd	DM/IP1200 (Resource Only board)	ul2_qsa_*t1ISDNprot*.pcd	DMV960_4T1 (BV+CSP 96 channels CONF+EC 15 channels Fax 4 channels PSTN 4 Trunks QSIGT1 protocol)
fn3_isdn_qsige1.pcd	DM/F300-1E1-PCI - E1 w/30 Ch. Fax FN3	ipvs_evr_r_ml11_311c.pcd	DM/IP1200 (Resource Only board)	us_hdsi.pcd	HDSI
fn3_isdn_qsigt1.pcd	DM/F240-1T1-PCI - T1 w/23 Ch. Fax FN3	ipvs_evr_r2mf_311.pcd	DM/IP301-E1 (DM/IPLINK-E1_NIC)	us_hdsi_48_play_rec.pcd	HDSI
fn3_r2mf.pcd	DM/F300-1E1-PCI - E1 w/30 Fax Channels FN3	ipvs_evr_r2mf_ml11_311.pcd	DM/IP301-E1 (DM/IPLINK-E1_NIC)	us_hdsi_72_play_rec.pcd	HDSI
fn3_t1.pcd	DM/F240-1T1-PCI - T1 w/24 Ch. Fax FN3	ipvs_evr_ts16_ml11_311.pcd	DM/IP301-1E1-100BT (DM/IPLINK-E1_NIC, 1 daughterboard, 30 channels)	us_hdsi_96_play_rec.pcd	HDSI
fr_hdsi.pcd	HDSI	it_hdsi.pcd	HDSI	vfn_isdn_4ess.pcd	DM/VF240-1T1-PCI - T1 w/24 Ch. Fax VFN
fr_hdsi_48_play_rec.pcd	HDSI	it_hdsi_48_play_rec.pcd	HDSI	vfn_isdn_5ess.pcd	DM/VF240-1T1-PCI - T1 w/24 Ch. Fax VFN
fr_hdsi_72_play_rec.pcd	HDSI	it_hdsi_72_play_rec.pcd	HDSI	vfn_isdn_dms.pcd	DM/VF240-1T1-PCI - T1 w/24 Ch. Fax VFN
fr_hdsi_96_play_rec.pcd	HDSI	it_hdsi_96_play_rec.pcd	HDSI	vfn_isdn_net5.pcd	DM/VF300-1E1-PCI - E1 w/30 Ch. Fax VFN
gb_hdsi.pcd	HDSI	jp_hdsi.pcd	HDSI	vfn_isdn_ni2.pcd	DM/VF240-1T1-PCI - T1 w/24 Ch. Fax VFN
Note: List of PCD files that do not support enhanced special information tones is subject to change with each Service Update.					

PCD Files That Do Not Support Enhanced Special Information Tones Feature					
PCD File (Unsupported)	**Description**	**PCD File (Unsupported)**	**Description**	**PCD File (Unsupported)**	**Description**
gb_hdsi_48_play_rec.pcd	HDSI	jp_hdsi_48_play_rec.pcd	HDSI	vfn_isdn_ntt.pcd	DM/VF240-1T1-PCI - T1 w/24 Ch. Fax VFN
gb_hdsi_72_play_rec.pcd	HDSI	jp_hdsi_72_play_rec.pcd	HDSI	vfn_isdn_qsige1.pcd	DM/VF300-1E1-PCI - E1 w/30 Ch. Fax VFN
gb_hdsi_96_play_rec.pcd	HDSI	jp_hdsi_96_play_rec.pcd	HDSI	vfn_isdn_qsigt1.pcd	DM/VF240-1T1-PCI - T1 w/23 Ch. Fax VFN
gdk_isdn_4ess.pcd	Cpi/2400CT-T1 ISDN 4ESS with 24 Fax Channels	lu_hdsi.pcd	HDSI	vfn_r2mf.pcd	DM/VF300-1E1-PCI - E1 w/30 Fax Channels VFN
gdk_isdn_5ess.pcd	Cpi/2400CT-T1 ISDN 5ESS with 24 Fax Channels	lu_hdsi_48_play_rec.pcd	HDSI	vfn_t1.pcd	DM/VF240-1T1-PCI - T1 w/24 Ch. Fax VFN
gdk_isdn_dms.pcd	Cpi/2400CT-T1 ISDN DMS with 24 Fax Channels	lu_hdsi_72_play_rec.pcd	HDSI	vfn3_isdn_4ess.pcd	DM/VF240-1T1-PCI - T1 w/24 Ch. Fax VFN3
gdk_isdn_net5.pcd	Cpi/3000CT-E1 ISDN NET5 with 30 Fax Channels	lu_hdsi_96_play_rec.pcd	HDSI	vfn3_isdn_5ess.pcd	DM/VF240-1T1-PCI - T1 w/24 Ch. Fax VFN3
gdk_isdn_ntt.pcd	Cpi/2400CT-T1 ISDN NTT with 24 Fax Channels	ml1_4x2_cas.pcd	DMV480_4T1 (BV 48 channels PSTN 4 Trunks CAS protocol)	vfn3_isdn_dms.pcd	DM/VF240-1T1-PCI - T1 w/24 Ch. Fax VFN3
gdk_t1_em.pcd	CPi/2400CT-T1 CAS E&M with 24 Fax Channels	ml1_4x2_r2mf.pcd	DMV600_4E1 (BV 60 channels PSTN 4 Trunks R2MF protocol)	vfn3_isdn_net5.pcd	DM/VF300-1E1-PCI - E1 w/30 Ch. Fax VFN3
gdk_t1_gs.pcd	CPi/2400CT-T1 with 24 Fax Channels	ml1_4x2_ts16.pcd	DMV600_4E1 (BV 60 channels PSTN 4 Trunks TS16 protocol)	vfn3_isdn_ni2.pcd	DM/VF240-1T1-PCI - T1 w/24 Ch. Fax VFN3
gdk_t1_ls.pcd	CPi/2400CT-T1 T1 CAS Loop Start with 24 Fax Channels	ml1_qs_cas.pcd	DMV960_4T1 (BV 96 channels PSTN 4 Trunks CAS protocol)	vfn3_isdn_ntt.pcd	DM/VF240-1T1-PCI - T1 w/24 Ch. Fax VFN3
hk_hdsi.pcd	HDSI	ml1_qs_r2mf.pcd	DMV1200_4E1 (BV 120 channels PSTN 4 Trunks R2MF protocol)	vfn3_isdn_qsige1.pcd	DM/VF300-1E1-PCI - E1 w/30 Ch. Fax VFN3
hk_hdsi_48_play_rec.pcd	HDSI	ml1_qs_ts16.pcd	DMV1200_4E1 (BV 120 channels PSTN 4 Trunks TS16 protocol)	vfn3_isdn_qsigt1.pcd	DM/VF240-1T1-PCI - T1 w/23 Ch. Fax VFN3
hk_hdsi_72_play_rec.pcd	HDSI	ml10_dsa_cas.pcd	DMV480A_2T1 (BV+CSP 48 channels CONF+EC 60 channels PSTN 2 Trunks CAS protocol)	vfn3_r2mf.pcd	DM/VF300-1E1-PCI - E1 w/30 Fax Channels VFN3
Note: List of PCD files that do not support enhanced special information tones is subject to change with each Service Update.					

PCD Files That Do Not Support Enhanced Special Information Tones Feature					
PCD File (Unsupported)	Description	PCD File (Unsupported)	Description	PCD File (Unsupported)	Description
hk_hdsi_96_play_rec.pcd	HDSI	ml10_dsa_r2mf.pcd	DMV600A_2E1 (BV+CSP 60 channels CONF+EC 60 channels PSTN 2 Trunks R2MF protocol)	vfn3_t1.pcd	DM/VF240-1T1-PCI - T1 w/24 Ch. Fax VFN3
ie_hdsi.pcd	HDSI	ml10_dsa_ts16.pcd	DMV600A_2E1 (BV+CSP 60 channels CONF+EC 60 channels PSTN 2 Trunks TS16 protocol)	za_hdsi.pcd	HDSI
ie_hdsi_48_play_rec.pcd	HDSI	ml1b_dsa_cas.pcd	DMV480A_2T1 (BV+ADSI/FSK 48 channels PSTN 2 Trunks CAS protocol)	za_hdsi_48_play_rec.pcd	HDSI
ie_hdsi_72_play_rec.pcd	HDSI	ml1b_dsa_r2mf.pcd	DMV600A_2E1 (BV+ADSI/FSK 60 channels PSTN 2 Trunks R2MF protocol)	za_hdsi_72_play_rec.pcd	HDSI
ie_hdsi_96_play_rec.pcd	HDSI	ml1b_dsa_ts16.pcd	DMV600A_2E1 (BV+ADSI/FSK 60 channels PSTN 2 Trunks TS16 protocol)	za_hdsi_96_play_rec.pcd	HDSI
Note: List of PCD files that do not support enhanced special information tones is subject to change with each Service Update.					

1.9 Troubleshooting Information for RTF Logs

To assist in troubleshooting, a table showing runtime and firmware errors that may appear in Dialogic® Runtime Trace Facility (RTF) logs is now available. You can get a description of errors and the suggested action to resolve the error. To access the table, use this link:

- Error Code Table

For runtime errors, the table provides the following information:

Internal error value
: The error code detected internally by the library. In some of the libraries, more than one internal error is mapped to an end user error. When contacting support about failures, this information will be helpful to the support engineer because it provides more specific information about why the error was generated. This number may appear in the RTF log (with the end user error value).

> ***Note:*** Sometimes the internal error value and end user error value are listed in the same trace entry. Sometimes the internal error value may appear as a separate entry.

End user error
: The name of the constant that is documented in the library API reference.

End user error value
: The numeric value of the constant that is documented in the library API reference. This is the value that will appear in the RTF log, which you can then search for in the table.

Description of the error
: A textual description of the error.

Action to be taken
: The suggested action to resolve the error.

For firmware errors, the table provides the following information:

Resource
: The firmware entity in which the error occurred. A resource is technically called a DM3 resource and is a software entity that provides a service to other DM3 resources. You can use the resource information to better narrow down what activity was occurring when the error occurred.

Loc hex
: The value that will appear in the RTF log (for example, 0x80000C), which you can then search for in the table.

Error class
: A classification of the firmware error in broad categories. You can use this column to understand the type of action to take for a particular type of error. For example, if an error is classified as a memory error, action can be taken that is specific to this type of error (such as a pool configuration change).

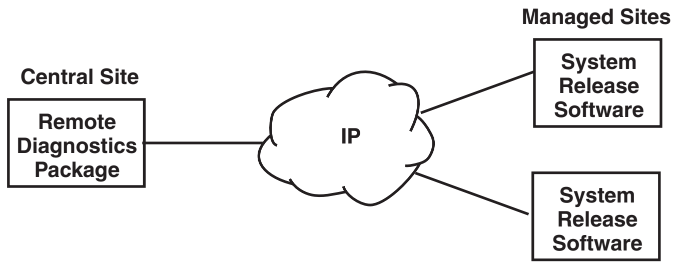
Error subclass
: Provides a bit more specialization with regard to the error class. Whenever possible, if a class could be subdivided into more specific classifications, it was done. The use of the error subclass is the same as that of the error class.

Action to be taken
: The suggested action to resolve the error.

1.10 Remote Diagnostics Package

A remote diagnostics package is now available that allows you to run Dialogic® diagnostics utilities remotely from a central site. The managed sites must have Dialogic®

System Release 6.0 PCI for Windows® installed. The central site does **not** need Dialogic® System Release 6.0 PCI for Windows® installed.



The remote diagnostics package is a *subset* of the system release software. It is designed for managing multiple remote sites from a central site, where the central site does not need the system software release or any Dialogic® boards installed. Instead, the remote diagnostics package must be installed at the central site. The diagnostics utilities in the remote diagnostics package are the same as the diagnostics utilities in Dialogic® System Release 6.0 PCI for Windows®.

1.10.1 Diagnostics Utilities

The remote diagnostics package includes the following utilities:

- Diagnostics Management Console (DMC)
- Runtime Trace Facility Manager and Server application (RTFManager, RTFServer)

For information about these utilities, see the *Dialogic® System Software Diagnostics Guide*.

1.10.2 Installing the Remote Diagnostics Package

The remote diagnostics package can be downloaded from the Dialogic support website.

Requirements at central site:

- SSH client
- IP connectivity to managed sites
- Java Runtime Environment (JRE) version 1.5 or later

Requirements at managed sites:

- SSH server
- IP connectivity to central site
- Dialogic® System Release 6.0 PCI for Windows® installed

1.11 New Parameter for Adjusting Silence Threshold on Dialogic® DM3 Boards

With the Service Update, the user has the ability to adjust the silence threshold parameter on Dialogic® DM3 Boards to a value above or below the default value of -43 dBm0 while using play and record functions like **dx_play( )**, **dx_record( )**, and **ec_reciottdata( )**. For instance, its adjustment affects the threshold for silence termination conditions in the R4 API TPT structure. It also affects silence detection via R4 unsolicited Standard Runtime Library (SRL) events.

The silence threshold is the level that defines whether the incoming data to a voice channel is recognized as silence or non-silence. The threshold is defined by the minimum energy level of a signal below which it is considered as silence. With this new feature, the user can statically adjust the silence threshold default value of -43 dBm0 via the DM3 firmware configuration file across all voice channels on a DM3 Board.

Configuration Example

To change the default value of the silence threshold, you must add a new parameter in the CONFIG file that was selected for your board. The parameter is **0x70B**, and must be added in the [sigDet] section of the CONFIG file. A value equal to the desired silence threshold, measured in dBm0, must be entered. For example:

```
[sigDet]

SetParm=0x70B, 0xffd3 ! SD_ParmMinEnergy in dBm0 (e.g. 0xffd3=-45, 0xffda=-38,
                        Default: 0xffd5=-43)
```

After the CONFIG file is saved, the changes take effect after downloading.

For further information about modifying DM3 CONFIG files, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

1.12 Support for PCI Express Boards - Dialogic® Station Interface Boards

With the Service Update, Dialogic® System Release 6.0 PCI for Windows® now supports the following Dialogic® Station Interface Boards in the PCI Express form factor:

Dialogic® DISI16-EW Switching Board
Provides connectivity for up to 16 station interfaces in a single, full-length PCI Express slot. Includes conferencing, voice play/record, tone detection and generation, and caller ID capabilities.

Dialogic® DISI24-EW Switching Board
Provides connectivity for up to 24 station interfaces in a single, full-length PCI Express slot. Includes conferencing, voice play/record, tone detection and generation, and caller ID capabilities.

Dialogic® DISI32-EW Switching Board

Provides connectivity for up to 32 station interfaces in a single, full-length PCI Express slot. Includes conferencing, voice play/record, tone detection and generation, and caller ID capabilities.

When configuring the system for the PCI Express form factor boards, use the same menu selections and configuration settings that are documented for the PCI version of the boards.

Note: When installing the Dialogic® DISI16-EW, DISI24-EW, and DISI32-EW Boards, be sure to refer to the Installation Guide (Dialogic® Quick Install Card) that is provided with each board for important information about power budgeting and guidelines for selecting the slot where a board can be installed.

1.13 Support for PCI Express Boards - Dialogic® DM/V-B Boards

Summary

With the Service Update, Dialogic® System Release 6.0 PCI for Windows® now supports the following Dialogic® DM/V-B Boards in the PCI Express form factor:

Dialogic® DMV300BTEPEQ Media Board

The DMV300BTEPEQ Board is a single span board with software selectable T1/E1.

- One digital network interface with 30+ channels of media processing.
- Support for universal media load with simultaneous voice, fax, and conferencing.
- Provides A-law/mu-law conversion and the ability to mix selected protocols on the board.
- PCI Express form factor.

Dialogic® DMV600BTEPEQ Media Board

The DMV600BTEPEQ Board is a dual span board with software selectable T1/E1 (per network interface).

- Two digital network interfaces with 60+ channels of media processing.
- Support for universal media load with simultaneous voice, fax, and conferencing.
- Provides A-law/mu-law conversion and the ability to mix selected protocols on the board.
- PCI Express form factor.

Dialogic® DMV1200BTEPEQ Media Board

The DMV1200BTEPEQ Board is a quad span board with software selectable T1/E1 (per network interface).

- Four digital network interfaces with 120+ channels of media processing.
- Support for universal media load with simultaneous voice, fax, and conferencing.
- Provides A-Law/Mu-Law conversion and the ability to mix selected protocols on the board.
- PCI Express form factor.

When configuring the system for the PCI Express form factor boards, use the same menu selections and configuration settings that are documented for the PCI version of the boards. Any differences are discussed below.

Note: When installing the Dialogic® DMV300BTEPEQ, DMV600BTEPEQ, and DMV1200BTEPEQ Boards, be sure to refer to the Installation Guide (Dialogic® Quick Install Card) that is provided with each board for important information about power budgeting and guidelines for selecting the slot where a board can be installed.

Features

The Dialogic® DMV300BTEPEQ, DMV600BTEPEQ, and DMV1200BTEPEQ Boards support the same features as the existing Dialogic® DM/V-B PCI Boards plus new media loads, lower latencies/increased performance, and first time support for a single span Dialogic® DM3 board.

Some of the features for these boards are listed below. Refer to the product data sheet, which is accessible at http://www.dialogic.com/products, for additional information about applications, configurations, features, and technical specifications.

- Software selectable T1/E1. Ability to mix T1 and E1 on each network interface.
- Ability to combine protocols on the same board. Protocols within a group can be mixed among network interfaces on the same board; however, protocols from different groups cannot be mixed on the same board.
  - Group 1: Mix any combination of 4ESS (T1), 5ESS (T1), NTT (T1), NI2 (T1), DMS (T1), QSIGT1 (T1), QSIGE1 (E1), NET5 (E1), T1CC (T1 Clear Channel), CAS (T1), E1CC (E1 Clear Channel), R2MF (E1) protocols on the same board.
  - Group 2: Mix any combination of DPNSS (E1) or DASS2 (E1) protocols on the same board.
- Ability to send alarm state to the network at all times from power-up to application start-up (i.e., trunk preconditioning).
- Universal load available (simultaneous voice + speech + fax + conferencing) on all Dialogic® DM/V-B Boards. All supported media loads are listed below.
- A-Law/Mu-Law conversion.

Note: Fixed routing configuration is not supported on Dialogic® DM/V-B Boards. Refer to the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* for more information about configuration, media loads, and mixing of protocols for the above features and board support.

Media Loads

The media loads supported by the Dialogic® DMV300BTEPEQ, DMV600BTEPEQ, and DMV1200BTEPEQ Boards are listed below.

ML	Media Loads/Features Supported											
	Voice Only								Fax	Conferencing Only		
	Basic Voice	Transaction Record	Enhanced Coders	TrueSpeech	Enhanced Echo Cancellation‡	CSP	CSP Streaming to CT Bus	FSK	Fax	Conferencing Parties	Conferencing - Tone Clamping	Conferencing - Echo Cancellation
Dialogic® DMV300BTEPEQ (Single Span Board)												
UL1	30	30	30	30	30	30	30	30	12	30	30	-
ML5	30	-	-	-	-	-	-	-	30	-	-	-
ML10	30	30	30	30	30	30	30	30	-	32	32	32
Dialogic® DMV600BTEPEQ (Dual Span Board)												
UL1	60	60	60	60	60	60	60	60	16	60	60	60
UL2	90	90	90	-	-	-	-	90	6	48	48	48
Dialogic® DMV1200BTEPEQ (Quad Span Board)												
UL1	120	120	120	120	120	120	120	120	12	30	30	-
UL2	120	120	-	-	-	-	-	120	12	120	120	-
UL3	120	120	120	-	-	-	-	120	8	36	36	36
ML2	150	150	150	150	-	150	-	-	-	-	-	-
ML5	120	-	-	-	-	-	-	-	30	-	-	-
ML5B	120	120	120	-	-	120	120	-	12	-	-	-
ML9B	-	-	-	-	-	-	-	-	-	160	160	160
ML9C	-	-	-	-	-		-	-	-	576	-	-
ML9D	-	-	-	-	-		-	-	-	270	270	-
ML10	120	120	120	-	-		120	120	-	54	54	54
ML10B	120	120	-	-	-		-	120	-	120	120	120

Notes: For more information about media loads, refer to the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide.*
Features within a resource group (headings marked as Voice Only, Fax, or Conferencing Only) are inclusive. Features across resource groups are additive. For example, on the Dialogic® DMV600BTEPEQ Board using UL1, there are 60 total voice resources, 16 fax resources, and 60 conferencing resources. This means that any combination of the listed voice resources (Voice Only subheadings marked as Basic Voice, Transaction Record, Enhanced Coders, TrueSpeech, Enhanced Echo Cancellation, CSP, CSP Streaming to CT Bus, and FSK) can be used up to a total of 60. For example, 30 Basic Voice plus 10 Enhanced Coders plus 10 TrueSpeech plus 10 CSP Streaming to CT Bus. In addition to these various voice resources, the UL1 media load can use 16 fax resources and 60 conferencing resources (with Tone Clamping and Echo Cancellation) simultaneously.
‡ Default configuration is EEC (enhanced EC, 32 ms) for CSP supported ML, unless otherwise indicated or set in the component named [0x2c] in the respective CONFIG file. You can only change it to a lower EC tail length, by changing the CSP parameter **0x2c03** accordingly in the respective CONFIG file. Conferencing EC, however, will always be 16 ms, regardless of the EC parameter setting.

1.14 Support for Dialogic® D/4PCI Voice Board

With the Service Update, the Dialogic® D/4PCI Voice Board that was supported in older system releases is now supported in Dialogic® System Release 6.0 PCI for Windows®. The D/4PCI Board has 4 voice channels (analog) and does not have CT Bus capabilities.

If present in the system, the D/4PCI Board will be detected and displayed in the Dialogic® Configuration Manager (DCM). Its default firmware file cannot be changed or configured.

For information about using DCM to configure and download Dialogic® Springware Boards, see the *Dialogic® Springware Architecture Products on Windows® Configuration Guide*.

1.15 New Parameter for Adjusting Silence Threshold on Dialogic® DM3 Boards

With the Service Update, the user has the ability to adjust the silence threshold parameter on Dialogic® DM3 Boards to a value above or below the default value of -43 dBm0 while using play and record functions like **dx_play( )**, **dx_record( )**, and **ec_reciottdata( )**.

The silence threshold is the level that defines whether the incoming data to a voice channel is recognized as silence or non-silence. The threshold is defined by the minimum energy level of a signal below which it is considered as silence. With this new feature, the user can statically adjust the silence threshold default value of -43 dBm0 via the DM3 firmware configuration file across all voice channels on a DM3 Board.

Configuration Example

To change the default value of the silence threshold, you must add a new parameter in the CONFIG file that was selected for your board. The parameter is **0x70B**, and must be added in the [sigDet] section of the CONFIG file. A value equal to the desired silence threshold, measured in dBm0, must be entered. For example:

```
[sigDet]

SetParm=0x70B, 0xffd3 ! SD_ParmMinEnergy in dBm0 (e.g. 0xffd3=-45, 0xffda=-38,
                        Default: 0xffd5=-43)
```

After the CONFIG file is saved, the changes take effect after downloading.

For further information about modifying DM3 CONFIG files, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

1.16 File Management Enhancements for ISDNtrace Tool

With the Service Update, the user can specify new command line options provided with the Dialogic® ISDNtrace tool to set the output log file size and to create multiple backup log files to be archived.

1.16.1 Feature Description

This feature enhances the existing ISDN tracing file management for boards configured with an ISDN load. Currently, all data is logged to a single file that can get too large during a session, and the batch operations can copy over files that might be needed. With this feature, the user can set command line options for size so that the single file is a manageable size, and also set options to create multiple log files when the file reaches the designated file size. In addition, the standard log file name format now conveniently shows the date and time the log was created. The user also has an option to disable logging to STDOUT to help manage trace output.

New Command Line Options

Currently, the ISDNtrace tool supports the following command line options as described in the *Dialogic® System Software Diagnostics Guide:*

```
syntax: isdntrace -b# [-f xxxx] [-d#]
```

-b<n>
: Logical ID of board (required). Use the listboards utility (Linux) or the Dialogic® Configuration Manager (DCM) (Windows®) to obtain the board's logical ID.

  Note: The listboards utility is described in the Administration Guide for the release and DCM is described in the Configuration Guides for the (Windows®) release.

-d<n>
: The D-channel number (trunk number) on the specified board. The default value is 1.

-f <file>
: Output log file name (required to save output in a file).

  Note: A space is used after the -f option but not after -b or -d options.

-h
: displays the same help information available in the ISDNtrace help menu screen. Note that this option does not show on the syntax above; however it is available.

For the ISDNtrace tool, new command line options have been added and the -f option enhanced to allow the user to manage log file(s) as follows:

```
syntax: isdntrace [-a#] -b# [-d#] [-f xxxx] [-m#] [-s]
```

-a<n>
: Log file array size, max=10, default=1, optional

-f <file>

Enable logging to file, optional

Note: A space is used after the -f option but not after -a, -b, -d, -m or -s options.

-m<n>

Max log file size (express in bytes; for example, 500,000 bytes is specified as -m500000), optional

Min=100 Kilobytes, max=100 Megabytes

Default=unlimited if log file array size=1, else 100 Megabytes

-s

Disable logging to STDOUT, optional

Details about these command line options follow:

-a<n>

This command line option allows the user to specify the maximum number of log files to maintain.

The user can specify a log file array size between 1 and 10. By default, the number of log files to be archived is 1. If the user specifies the -f command line option but does not specify this option (or specifies it with an array size of 1), then ISDNtrace creates a single log file that grows without bound (that is, no limit to the log file size).

If the user specifies this option with an array size greater than 1 (but less than or equal to 10), then ISDNtrace creates an initial log file at startup. When the log file reaches the maximum file size (either the default maximum log file size or the value specified via the -m command line option), the log file is closed and a new log file is created.

Whenever ISDNtrace attempts to open a new log file, it first checks to see if the current number of log files created is equal to the number of files specified for the log file array. If not, then the new log file is created. Otherwise, the oldest log file is deleted and a new log file is created to replace it.

It should be noted that any ISDNtrace log files that exist prior to running the ISDNtrace tool are not deleted or modified in any way. Due to the new log file naming convention (see -f option), all ISDNtrace log files have unique timestamps in their log file names and are not overwritten when ISDNtrace starts up.

-f <file>

This option existed in the previous versions of ISDNtrace. However, the processing associated with this option has been modified to include date and time information.

This command line option specifies the log file name of the log file into which the trace can be captured. If this option is not specified on the command line, then no trace output will be saved to a log file.

The naming of ISDNtrace log files has been modified to fit the following format:

```
<File>-MMDDYYYY-xxhyymzzs.log
```

where:

- MM - current month (01=Jan, 02=Feb, 03=Mar, … 12=Dec)
- DD - current day of the month
- YYYY - current year (e.g. 2006)
- xx - current hour in day (24 Hour Format, 00-23)

- yy - current minute in hour (00 - 59)
- zz - current second in minute (00 - 59)

In the description above, the log file name is what the user specified on the command line. If the user specifies a -f command line option as the last parameter on the command line and does not specify a log file name, then the default log file name of ISDNTRACE will be used.

Note: In order to get a default log file name, the -f option has to be used at the end of the command line.

For example, if the user started ISDNtrace specifying the -f command line option without a log file name on January 17, 2007 at 03:11:27 pm, the log file created would be:

```
isdntrace-01172007-15h11m27s.log
```

Alternatively, the user can specify the -f command line option with a log file name specified as in the following example:

```
isdntrace -b0 -f test
```

In this example, if ISDNtrace was started on January 17, 2007 at 03:11:27 pm, the resultant log file name would be:

```
test-01172007-15h11m27s.log
```

It should be noted that since the log file name created by ISDNtrace has a .log extension appended to it, if the user specifies a log file name with a .log extension already appended to it, the resultant log file name will have the date and time inserted between the root log file name and the extension. For example, if the user issued the following command line:

```
isdntrace -b0 -f 4ess_test.log
```

Then the resultant log file name would be:

```
4ess_test-01172007-15h11m27s.log
```

-m<n>

The -m command line option is used to specify the maximum log file size. By default, the maximum log file size is 100 Megabytes. The valid range that can be specified for the maximum log file size is from 100 Kilobytes up to 100 Megabytes.

The format of the file size is specified as a long integer value. For example, to specify a maximum log file size of 250,000 bytes, the following command line should be specified:

```
isdntrace -b0 -m250000 -f test.log
```

It should also be noted that the -m command line option will have no effect if the log file array size is 1, in which case the log file will be allowed to grow in size without limit.

-s

The -s command line option can be specified to prevent trace output to STDOUT. When ISDNtrace attempts to capture a large amount of trace information in a short amount of time, its processing can fall behind if trace output is displayed to STDOUT. This will result in “enqueue fail” failures and the loss of trace information as seen in the example below:

```
                                 Tue Jan 16 17:30:58 2007
                                 TX Frame: Time = 2428.372
                                 Command=1   SAPI=0x00
                                 TEI=0x00
                                 0x01 0xe6  Receive Ready
                                 Hex Dump:
                                 02 01 01 e6
Enqueue Failed

Tue Jan 16 17:30:58 2007
RX Frame: Time = 2428.372
Command=1   SAPI=0x00
TEI=0x00
0xe6 0xce  Information
PD=0x08 Dest=0 CR=0x1e2a
CALL DISCONNECT(0x45)
  1:          CAUSE(0x08)
  2:          IE Length(0x02)
  3:  1------- Extension Bit
      -00----- Coding Standard
      ---0---- Spare
      ----0010 Location
  4:  1------- Extension Bit
      -0010000 Cause Value
Hex Dump:
02 01 e6 ce 08 02 1e 2a 45 08
02 82 90
Enqueue Failed
```

In order to avoid loss of trace information and provide more robust performance of the ISDNtrace tool, the -s command line option should be specified to disable trace output to STDOUT whenever the capture of trace information for a large amount of calls is being performed, or “enqueue fail” failures occur.

1.16.2 Supported Boards

The following boards support this feature:

- Dialogic® DM3 Network Interface Boards

1.16.3 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the ISDNtrace tool, see the *Dialogic® System Software Diagnostics Guide.*

1.17 Support for Dialogic® DI/0408-LS-AR2 Board

With the Service Update, the Dialogic® DI/0408-LS-AR2 Board is now supported in Dual Processor systems.

1.18 Change in ipmedia.log Implementation

With the Service Update, the *ipmedia.log* file is no longer overwritten when the system is restarted.

The *ipmedia.log* file is generated whenever IP Media Services are run. If there is an existing log file when the system is restarted, it is saved and renamed *ipmedia.log.bak*. If *ipmedia.log.bak* already exists, it is overwritten (only one backup file is saved).

1.19 Adjusting Pre-Record Beep Tone Characteristics through the CONFIG File

With the Service Update, several Dialogic® Boards now support the ability to modify the pre-record beep tone characteristics. This new functionality is provided through the configuration file set. Changed values take effect at the time the firmware is downloaded to the board using the Dialogic® Configuration Manager (DCM) utility.

1.19.1 Supported Boards

The following boards support this feature:

- Dialogic® DI0408LSAR2 Switching Boards
- Dialogic® DM/V160LP Media Boards
- Dialogic® DM/V480A-2T1-PCI Media Boards
- Dialogic® DM/V600A-2E1-PCI Media Boards
- Dialogic® DM/V960A-4T1-PCI Media Boards
- Dialogic® DM/V1200A-4E1-PCI Media Boards
- Dialogic® DM/IP241-1T1-PCI-100BT IP Boards
- Dialogic® DM/IP301-1E1-PCI-100BT IP Boards
- Dialogic® DM/IP481-2T1-PCI-100BT IP Boards
- Dialogic® DM/IP601-2E1-PCI-100BT IP Boards

1.19.2 Feature Description

A beep tone is used in some applications to indicate the start of recording. This beep tone is enabled through the RM_TONE value in the **mode** parameter of various record functions (for example, **dx_reciottdata( )**) in the Voice API library. The characteristics of the pre-record beep tone were previously hardcoded and differed on Dialogic® Springware Boards versus Dialogic® DM3 Boards.

With the Service Update, you can modify the beep tone values, such as the amplitude, in the Tone Templates [tonegen] section of a particular media load CONFIG file. Default values are provided that are consistent with previous service updates and system releases to preserve backward compatibility.

Pre-Record Beep Tone Characteristics and Default Values

Two pre-record beep tones are defined:

- A custom customer tone, BEEP_DGSD, defined as 444 Hz for 400 ms, and BEEP_DLGC, defined as 1000 Hz for 400 ms.
- The traditional pre-record beep tone, BEEP_DLGC, defined as 1000 Hz. This corresponds to the beep tone definition on Springware Boards and is the default setting.

The [recorder] section of the CONFIG file includes the following parameter, which specifies the tone to be used in the application:

BeepSignalID (Pre-Record Beep Tone)

Number: 0x203

Description: The BeepSignalID parameter is the signal identifier of the beep tone preceding the recording.

Values:

- 0x21: 444 Hz tone for 400 ms
- 0x22: 1000 Hz tone for 400 ms (default)

The pre-record beep tone characteristics for the two beep tones, stored in the Tone Templates [tonegen] section of the CONFIG file, are described as follows:

Record Beep Tone	Characteristic	Default Value
BEEP_DGSD	Signal Id	33
	Label	(blank)
	Segment Count	1
	Segment Signal Type	2
	Segment Frequency 1 (Hz)	444
	Segment Amplitude 1 (.25 dbm)	-40
	Segment Frequency 2 (Hz)	0
	Segment Amplitude 2 (.25 dbm)	0
	Segment On Duration (125 microsecs)	3200
	Segment Off Duration (125 microsecs)	320
	Segment Reps	1
	Next Segment	65535

Record Beep Tone	Characteristic	Default Value
BEEP_DLGC	Signal Id	34
	Label	(blank)
	Segment Count	1
	Segment Signal Type	2
	Segment Frequency 1 (Hz)	1000
	Segment Amplitude 1 (.25 dbm)	-40
	Segment Frequency 2 (Hz)	0
	Segment Amplitude 2 (.25 dbm)	0
	Segment On Duration (125 microsecs)	3200
	Segment Off Duration (125 microsecs)	320
	Segment Reps	1
	Next Segment	65535

Media Loads Supported

The following media loads support the new functionality to modify pre-record beep tone parameter values:

- On DI0408LSAR2 Boards, all media loads support the new functionality.
- On DMV160LP Boards, all media loads support the new functionality.
- On DM/V480A-2T1-PCI Boards, Media Loads 1b and 10 support the new functionality.
- On DM/V600A-2E1-PCI Boards, Media Loads 1b and 10 support the new functionality.
- On DM/V960A-4T1-PCI Boards, Media Loads 1b and 5 support the new functionality.
- On DM/V1200A-4E1-PCI Boards, Media Load 1b and Universal Media Load 1 support the new functionality.
- On DM/IP Boards, Media Load 11 supports the new functionality.

1.19.3 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about configuration files, configuration parameters, and configuration procedures, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

1.20 Reduced Dial Tone Delay with MWI

The **ms_SetMsgWaitInd( )** function generates an FSK signal to illuminate the message waiting LED. When a user of a station goes off-hook before the **ms_SetMsgWaitInd( )**

function is complete, a slight delay occurs before hearing the dial tone. With the Service Update, this delay has been reduced by 12%.

1.21 Enhanced Diagnostics

The Service Update provides several new and enhanced diagnostics features. The following sections introduce some of the new features:

- PSTN Diagnostics (pstndiag)
- Status Monitor (statusmon)
- New Dialogic® Diagnostics Management Console
- New Runtime Trace Facility (RTF) Manager

Note: Java Runtime Environment (JRE) version 1.5 or later must be installed on your system in order to run the new diagnostics tools.

1.21.1 PSTN Diagnostics (pstndiag)

The PSTN Diagnostics tool (pstndiag) is a utility for diagnosing and troubleshooting call control issues on public switched telephone network (PSTN) connections.

The pstndiag tool has a graphical user interface (GUI). When you start the tool, a tree view of all installed Dialogic® DM3 Boards is displayed. The view can be expanded to show the lines (trunks) on each board and the channels on each line. At each level (board, line, channel), different diagnostics activities can be launched, for example:

- At the board level, you can display board configuration (board name, board number, number of lines, number of channels per line, and signaling type). You can also launch the statusmon tool. (The new statusmon tool is described in Section 1.21.2, “Status Monitor (statusmon)”, on page 82.)
- At the line level, you can launch the lineadmin tool to put lines in/out of service, generate transmit alarms, enable/disable various types of loopbacks, and report bipolar violations, consecutively errored seconds, frame errors, and other saturation alarms.
- At the channel level, you can launch the phone tool to perform call control operations. You can also trace all call related activity on a given channel and store it in a columnar format based on timestamp deltas.

Running the PSTN Diagnostics Tool

To run the **new** version of pstndiag, enter the command:

- `pstndiag -j`

(The previous version of the tool is still supported and can be run by entering the command `pstndiag` without the `-j`.)

The new version of pstndiag includes the following changes:

- Faster startup
- Changes in the board tree view
- Additional features in the lineadmin tool: enabling all supported loopback modes and counters for saturation alarms
- Configurable modes of operation for the phone tool: basic, advanced, and expert

Note: More detailed information about the new version of pstndiag is planned to be provided in the *Dialogic® System Software Diagnostics Guide*, which is scheduled to be updated soon.

1.21.2 Status Monitor (statusmon)

The Status Monitor tool (statusmon) is a utility for monitoring the current activity on all lines and channels on a Dialogic® DM3 Board. The primary use case is as a long-term monitoring tool.

The statusmon tool displays the following information:

- Alarm status (red, yellow, LOS)
- Channel state
- Call state

Running the Status Monitor Tool

The statusmon tool is typically launched from pstndiag, but it can also be run on its own. To run the **new** version of statusmon, enter the command:

- `run_statusmon.sh -board #`

where # is the logical board number of the board to monitor.

(The previous version of the tool is still supported and can be run by entering the command `statusmon board` or `statusmon board trunk channel`.)

The new version of statusmon includes the following changes:

- No line (trunk) or channel mode. However, these capabilities are supported via the pstndiag tool.

Note: More detailed information about the new version of statusmon is planned to be provided in the *Dialogic® System Software Diagnostics Guide*, which is scheduled to be updated soon.

1.21.3 New Dialogic® Diagnostics Management Console

The Service Update introduces the Dialogic® Diagnostics Management Console (DMC) version 1.0. This GUI tool provides a means of quickly launching Dialogic® diagnostic utilities and viewing various log files created with those utilities.

The DMC:

- Provides a single portal for launching diagnostic tools:
  - AppMon
  - Castrace
  - Isdntrace
  - Dlgsnapshot
  - Dm3post
  - Debugangel
  - Getver
  - its_sysinfo
  - Pdktrace
  - Pstndiag
  - RTF Manager
  - StatusMon
- Supports local and remote execution of tools. Diagnostic tools are launched remotely via the standard remote control methods provided with the operating system, such as SSH or Remote Desktop.
- Lists the diagnostic logs available both locally and remotely for viewing.
- Launches appropriate viewers for displaying logged data.

For more information about the DMC, refer to the *Dialogic® System Software Diagnostics Guide*. The DMC also has online help.

1.21.4 New Runtime Trace Facility (RTF) Manager

The Service Update introduces the RTF Manager, a new GUI for the Runtime Trace Facility (RTF) diagnostic tool. RTF Manager allows users to easily configure logging and tracing levels. Previously, users had to manually edit the RTF configuration file.

For more information about the RTF Manager, refer to the *Dialogic® System Software Diagnostics Guide*.

1.22 Support for PCI Express Boards - Dialogic® Springware Boards

With the Service Update, Dialogic® System Release 6.0 PCI for Windows® now supports the following PCI Express boards:

- Dialogic® D/42JCT-EW and Dialogic® D/82JCT-EW PBX Integration Boards
- Dialogic® D/240JCT-T1-EW and Dialogic® D/300JCT-E1-EW Media Boards
- Dialogic® D/480JCT-EW and Dialogic® D/600JCT-EW Media Boards
- Dialogic® D/4PCIE-4S-W and Dialogic® D/4PCIE-4F-W Media Boards

- Dialogic® D/41JCT-LS-EW and Dialogic® VFX/41JCT-LS-EW Media Boards
- Dialogic® D/120JCT-LS-EW Media Board

When configuring the system for the PCI Express form factor boards, use the same menu selections and configuration settings that are documented for the PCI version of the boards. Any differences are discussed below.

Dialogic® D/42JCT-EW and Dialogic® D/82JCT-EW PBX Integration Boards

The Dialogic® D/42JCT-EW and Dialogic® D/82JCT-EW PBX Integration Boards offer advanced digital connectivity to many of today's most popular private branch exchanges (PBXs) for unified and Internet-ready call, voice, and fax processing in small- to medium-sized enterprises. The D/42JCT-EW Board is a 4-port voice processing board in a full-length PCI Express form factor. The D/82JCT-EW Board is an 8-port voice processing board in a full-length PCI Express form factor.

Note: When installing the D/42JCT-EW and D/82JCT-EW Boards, be sure to refer to the Installation Guide (Dialogic® Quick Install Card) that is provided with each board for important information about power budgeting and guidelines for selecting the slot where a board can be installed.

Dialogic® D/240JCT-T1-EW and Dialogic® D/300JCT-E1-EW Media Boards

The Dialogic® D/240JCT-T1-EW Media Board is a 24-channel voice and T1 network interface board in a full-length PCI Express form factor.

The Dialogic® D/300JCT-E1-EW Media Board is a 30-channel voice and E1 network interface board in a full-length PCI Express form factor. The board is available in a 75-Ohm version and a 120-Ohm version.

Note: When installing the D/240JCT-T1-EW and D/300JCT-E1-EW Boards, be sure to refer to the Installation Guide (Dialogic® Quick Install Card) that is provided with each board for important information about power budgeting and guidelines for selecting the slot where a board can be installed.

Dialogic® D/480JCT-EW and Dialogic® D/600JCT-EW Media Boards

The Dialogic® D/480JCT-EW and Dialogic® D/600JCT-EW PCI Express form factor boards include the following models:

Dialogic® D/480JCT-1T1-EW Media Board
Provides up to 24 channels of combined media resources and a single T1 network interface in a single, full-length PCI Express slot.

Dialogic® D/480JCT-2T1-EW Media Board
Provides up to 48 channels of combined media resources and two T1 network interfaces in a single, full-length PCI Express slot.

Dialogic® D/600JCT-1E1-75-EW Media Board
Provides up to 30 channels of combined media resources and a single 75-ohm E1 network interface in a single, full-length PCI Express slot.

Dialogic® D/600JCT-1E1-120-EW Media Board
Provides up to 30 channels of combined media resources and a single 120-ohm E1 network interface in a single, full-length PCI Express slot.

Dialogic® D/600JCT-2E1-75-EW Media Board
Provides up to 60 channels of combined media resources and two, 75-ohm E1 network interfaces in a single, full-length PCI Express slot.

Dialogic® D/600JCT-2E1-120-EW Media Board
Provides up to 60 channels of combined media resources and two, 120-ohm E1 network interfaces in a single, full-length PCI Express slot.

Notes:1. When installing the D/480JCT-EW and D/600JCT-EW PCI Express Boards, be sure to refer to the Installation Guide (Dialogic® Quick Install Card) that is provided with each board for important information about power budgeting and guidelines for selecting the slot where a board can be installed.

2. The D/480JCT-EW and D/600JCT-EW PCI Express Boards can be used with any System Release 6.0 PCI Windows Service Update release; it is not necessary to upgrade to a particular Service Update.

Dialogic® D/4PCIE-4S-W and Dialogic® D/4PCIE-4F-W Media Boards

The Dialogic® D/4PCIE-4S-W and Dialogic® D/4PCIE-4F-W Media Boards are combined media analog boards with four ports of voice, fax, and speech in a half-length PCI Express form factor. The D/4PCIE-4S-W Board has four ports of voice and speech, and the D/4PCIE-4F-W Board has four ports of voice and fax.

Dialogic® D/41JCT-LS-EW and Dialogic® VFX/41JCT-LS-EW Media Boards

The Dialogic® D/41JCT-LS-EW and Dialogic® VFX/41JCT-LS-EW Media Boards are combined media analog boards with H.100 connectivity and four ports of voice, fax, and speech in a full-length PCI Express form factor. The D/41JCT-LS-EW Board supports basic fax, and the VFX/41JCT-LS-EW Board supports enhanced fax.

Dialogic® D/120JCT-LS-EW Media Board

The Dialogic® D/120JCT-LS-EW Media Board is a 12-port analog telecom board in a full-length PCI Express form factor. The D/120JCT-LS-EW Board supports voice, fax, and software-based speech recognition processing in a single PCI Express slot, and provides 12 analog telephone interface circuits for direct connection to analog loop start lines.

Notes:1. When installing the D/120JCT-LS-EW Board, be sure to refer to the Installation Guide (Dialogic® Quick Install Card) that is provided with each board for important information about power budgeting and guidelines for selecting the slot where a board can be installed.

2. The D/120JCT-LS-EW Board can be used with any System Release 6.0 PCI Windows Service Update release; it is not necessary to upgrade to a particular Service Update.

1.23 PDK Trace Supports CAS/R2MF/Tone Tracing

With the Service Update, the Dialogic® DM3 PDK Protocol Trace (PDK Trace) tool has new functionality to log CAS, R2MF, and tone-on/tone-off information on supported boards. Formerly, the DM3 PDK Trace tool only logged SDL state transitions. Also, the tool for converting the binary output has been enhanced so the R2MF/CAS/tone-on/tone-off output, like the SDL data, can be converted into a readable format.

Note: See the *Dialogic® System Software Diagnostics Guide* for more information about the DM3 PDK Protocol Trace tool.

1.23.1 Feature Description

The functionality is enabled by using PDK Trace with a new command line option, `-e|E`. This enhanced option enables R2MF tone exchanges (when using R2MF protocol), CAS signaling changes, and tone-on/tone-off event tracing on supported boards. As with other PDK options, the new command option produces a default binary log file, *pdktrace.log*, which can be converted into readable files by contacting Dialogic technical support. The converted log is sent back to the user to interpret (see Sample Output Logs).

Enhanced tracing is not affected when a mixed ISDN/CAS configuration is used. As part of the processing done for tracing, the protocol type (PDK or ISDN) of the given trunk is queried. If it is a PDK trunk, then CAS, R2MF, and tone-on/tone-off events are traced. If the trunk is ISDN, tone-on/tone-off events are traced.

Note: For the Dialogic® DM/V-B Boards, **gc_Open( )** must be called **prior** to starting enhanced tracing on a channel. If tracing is started prior to **gc_Open( )** being called, PDK protocol tracing will function, but no CAS/ R2MF/TONE events will be detected.

New Option

Note: See the *Dialogic® System Software Diagnostics Guide* for a complete description of all the options and instructions for using the PDK Trace tool.

`-e`

This option enables the R2MF, CAS, and tone-on/tone-off event tracing on supported boards.

Example: `pdktrace -b0 -i -e` / *basic protocol tracing and enhanced tracing enabled for channel 1 on trunk 1, where 0 is the logical board ID of the destination board */

1.23.2 Supported Boards

The following boards support this feature:

- Dialogic® DM/V1200BTEP Media Boards
- Dialogic® DM/V600BTEP Media Boards
- Dialogic® DM/V3600BP Media Boards

1.23.3 Sample Output Logs

The following are examples and explanations of the converted output:

R2MF Outbound

File	Line	Level	Ser	Bo...	C...	SDL	Call	TxRx	Data
N/A	1	DEBUG	N/A	3	11	RSMF	Un	TX:1000 RX:1011	R2MF tone (OUTBOUND): Trunk=3 Chan=11 Tone=7

This event is logged when an R2MF tone has been transmitted by a channel that is being traced. This tone would be considered a “backward” tone if the channel is the called party, or a “forward” tone if the channel is the calling party. The tone value given represents one of the 15 possible R2MF tone numbers in the forward or backward tone set. The tone numbers (1-15) represent tone pair frequencies defined in various ITU standards.

R2MF Inbound

File	Line	Level	Ser	Bo...	C...	SDL	Call	TxRx	Data
N/A	1	DEBUG	N/A	3	11	RSMF	Un	TX:1000 RX:1011	R2MF tone (INBOUND): Trunk=3 Chan=11 Tone=5

This event is logged when an R2MF tone has been received by a channel that is being traced. This tone would be considered a “backward” tone if the channel is the calling party, or a “forward” tone if the channel is the called party. The tone value given represents one of the 15 possible R2MF tone numbers in the forward or backward tone set. The tone numbers (1-15) represent tone pair frequencies defined in various ITU standards.

CAS RX

File	Line	Level	Ser	Bo...	C...	SDL	Call	TxRx	Data
N/A	1	DEBUG	N/A	3	11	CAS	Un	TX:0000 RX:1001	CAS Transition (RX) PreTransition Code: 0x1 PostTransition Code: 0x1

This event is logged when a new ABCD bit pattern is detected on the channel being traced. Any change in any of the ABCD bits will cause a new CAS RX event to be generated. The “PreTransition Code” is the value of the ABCD bits (bits 0-3) and the bit mask (bits 4-7) **prior** to the change. The “PostTransitionCode” represents the ABCD bits and the bit mask of the newly detected signaling pattern. The bit mask is used to denote which of the ABCD bits are of significance for the pattern being transmitted.

CAS TX

File	Line	Level	Ser	Bo...	C...	SDL	Call	TxRx	Data
N/A	1	DEBUG	N/A	3	11	CAS	Un	TX:1001 RX:1001	CAS Transition (RX) PreTransition Code: 0x3d PostTransition Code: 0xd

This event is logged when a new ABCD bit pattern is transmitted on the channel being traced. Any change in any of the ABCD bits will cause a new CAS TX event to be generated. The “PreTransition Code” is the value of the ABCD bits (bits 0-3) and the bit mask (bits 4-7) **prior** to the change. The “PostTransitionCode” represents the ABCD bits and the bit mask of the newly transmitted signaling pattern. The bit mask is used to denote which of the ABCD bits are of significance for the pattern being transmitted.

Tone-on

File	Line	Level	Ser	Bo...	C...	SDL	Call	TxRx	Data
N/A	1	DEBUG	N/A	2	5	TONE_ ON	Un	TX:1011 RX:1000	TONEON EVENT: ID=14bcc Freq1=1737 Freq2= 1499 OnTime =8 OffTime = 4

This event is logged when a new tone has been received and matches a defined tone template. The Event ID represents the tone ID of the tone template that was matched. Freq1 represents the first frequency of a dual tone. Freq2 represents the second frequency in the dual tone. These frequencies are given in Hz. If the tone is a single frequency tone, then Freq2 would be 0. Ontime represents the amount of time (in microseconds) that the tone was present. Offtime represents the minimum amount of time (in microseconds) that the tone was not present.

Notes:1. Tone-on/tone-off events come in pairs.

2. Ontime and Offtime values are defined in the [sigdet] section of the tone template in the CONFIG/FCD file, as well as via the Voice (dx) API.

Tone-off

File	Line	Level	Ser	Bo...	C...	SDL	Call	TxRx	Data
N/A	1	DEBUG	N/A	2	5	TONE_ OFF	Un	TX:1011 RX:1000	TONEOFF EVENT: ID=14bcc

This event is logged when a tone matching a tone template is no longer present. The Event ID represents the tone ID of the tone template that was previously matched and is no longer present.

1.23.4 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about PDK Trace, see the *Dialogic® System Software Diagnostics Guide*.

Note: The online bookshelf has not been updated for this feature, so the new option is not currently documented in the *Dialogic® System Software Diagnostics Guide*.

1.24 Ability to Lower or Disable White Noise

With the Service Update, the user can lower the white noise gain or disable the injection of white noise entirely by adding a new parameter to the CONFIG file.

1.24.1 Feature Description

Users can lower the white noise gain or disable the injection of the white noise entirely when the white noise produces a distracting “hiss” noise during conference calls. The user can control the amount of white noise by adding the parameter, **0x2c22**, to the CONFIG file and uncommenting out the desired option.

Number: 0x2c22

Description: Add the parameter to the CONFIG file to disable the injection of white noise entirely or to set a value that reduces the level of the white noise.

Values:

- 0 (disables white noise gain completely). Use this setting if white noise is not desired.

  Note: When white noise is disabled, the user will have no “noise” to indicate that the application is still working; therefore, the user may want the 0xfff setting instead.
- 0xfff (sets white noise gain to a very low value). Use this setting if some small level of noise is desired so that there is not complete silence.
- 0x4285fc (sets the white noise gain to the default of -43 dB). This value does not have to be set by the user; it is the default value used if the parameter is omitted from the CONFIG file.

Example:

The following is an example for disabling white noise:

```
[0x2c]
SetParm=0x2c22, 0                 ! Disables white noise gain
!SetParm=0x2c22, 0xfff            ! Sets the white noise gain to a very low value
!SetParm=0x2c22, 0x4285fc         ! Sets the white noise gain to the default of -43 dB
```

1.24.2 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about setting parameters in the CONFIG file, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Note: The online bookshelf has not been updated for this feature, so the new option is not currently documented in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

1.25 Optional Use of Sharing of Timeslot (SOT) Algorithm

The Sharing of Timeslot (SOT) algorithm for Dialogic® DM3 Boards maximizes the efficiency of the internal timeslots used for external transmit reference, allowing a full 120 channel density for such features as continuous speech processing and transaction record. The SOT algorithm is enabled by default, regardless of whether continuous speech processing or transaction record functionality is needed. Its use places certain constraints on an application for performing listen/unlisten functions in a specific sequence.

For increased flexibility in application design, it is now possible to disable the SOT algorithm by adding a new parameter, **QKERNEL_DISABLE_TIMESLOT_SHARING**, to the board's CONFIG file.

Note: The SOT algorithm is now supported on the Dialogic® DM/V600-4E1 Board with media load ml1_4x2_r2mf.

For more detailed information about the SOT algorithm, guidelines for enabling or disabling the algorithm, and supported boards and media loads, see the technical note titled "Disabling the Sharing of Timeslot (SOT) Algorithm via DM3 config file change" on the Dialogic website at
http://www.dialogic.com/support/helpweb/dxall/tnotes/legacy/2000/tn104.htm

1.26 New FSK Transmit and Receive Signal Level Parameters

With the Service Update, there are new transmit and receive FSK parameters that let you change the volume level of the FSK modem signals sent and received by the board using the parameter control in the configuration/FCD files for Dialogic® DM3 Boards.

1.26.1 Feature Description

The FSK transmit (**FM_ParmFSKTxSignalLevel**) and receive (**FM_ParmFSKRxSignalLevel**) signal level parameters are described below.

FSK Transmit and Receive Signal Level

Number: 0x2a (0x2a04 for transmit) (0x2a00 for receive)

Description: Two-way Frequency Shift Keying (FSK) and ETSI FSK allow the exchange of small amounts of data between a telephone and the server using FSK as the transport layer. The two-way FSK functionality allows products to transmit and receive half-duplex FSK Bell 202 1200 bps data over the Public Switched Telephone Network (PSTN). ETSI FSK functionality is based on the specification ETSI 201 912.

The Transmit and Receive Signal Level parameters allow you to adjust the signal level of both the transmit and receive FSK signal levels.

Values:

- -50 to -5 dbm for FSK transmit signal level, -14 dbm (default)
- -60 to -5 dbm for FSK receive signal level, -46 dbm (default)

Guidelines for FSK Transmit: To set the signal level of the FSK transmit signal to other than the default value of -14 dbm, you must edit the applicable CONFIG file.

Example: To set the FSK transmit signal level to a value of -20 dbm, you need to add a new section [0x2a] at the end of the CONFIG file and include the FSK Transmit Signal Level parameter in that section as follows (shown in bold):

```
[0x2a]
SetParm=0x2a04,-20    !FM_ParmFSKTxSignalLevel
```

Guidelines for FSK Receive: To set the signal level of the FSK receive signal to other than the default value of -46 dbm, you need to edit the CONFIG file by adding the FSK Receive Signal parameter to the new [0x2a] section.

Example: To set the receive signal level to a value of -15 dbm, add the line shown in bold to the new section you created for the FSK Transmit Signal Level parameter:

```
[0x2a]
SetParm=0x2a04,-20    !FM_ParmFSKTxSignalLevel
SetParm=0x2a00,-15    !FM_ParmFSKRxSignalLevel
```

1.26.2 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about configuration files, configuration parameters, and configuration procedures, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Note: The online bookshelf has not been updated for this feature, so these new parameters are not currently documented in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

1.27 Support for Reporting Billing Type

With this Service Update, for Dialogic® DM3 Boards, there is now a way for the application to know which billing type (for a call on PDK R2 protocol) was received when the lines are available for call establishment. B tones are sent to indicate whether the line is available or not, and also to indicate the type of billing for the call (for example, CHARGE, NO CHARGE, or CHARGE WITH CLEARING FROM INBOUND).

This feature is already supported on Dialogic® Springware Boards; however, CHARGE WITH CLEARING FROM INBOUND is a new billing type that is also supported on Springware Boards now.

1.27.1 Feature Description

The user is notified of the billing type for a successful call establishment. The **gc_GetCallInfo( )** function with info_id equal to CALLINFOTYPE is used to retrieve the billing type. The following mappings are implemented:

Group B Tone	Billing Type String Returned
GrpB - line free, charge	“CHARGE”
GrpB - line free, no charge	“NO CHARGE”
GrpB - line free, charge with clearing from inbound only	“CHARGE WITH CLEARING FROM INBOUND”

For B tones indicating unavailability of the line (call establishment failure), the following mappings are used for assigning cause values to the GCEV_DISCONNECT event:

Group B Tone	GC Cause Value	Description
GrpB - User Busy	GCRV_BUSY	“Line is busy”
GrpB - Network Congestion	GCRV_CONGESTION	“Congestion”
GrpB - Normal Clearing	GCRV_NORMAL	“Normal Clearing”
GrpB - UnAssigned Number	For DM3 Boards: GCRV_UNALLOCATED For Springware Boards: GCRV_NOT_INSERVICE	For DM3 Boards: “Number not allocated” For Springware Boards: “Number not in service”
GrpB - SIT	For DM3 Boards: GCRV_SIT_UNKNOWN For Springware Boards: GCRV_CEPT	For DM3 Boards: “Unknown SIT detected” For Springware Boards: “Operator intercept”
GrpB - Rejected	GCRV_REJECT	“Call Rejected”

Note: If the billing type is not supported on a protocol, then **gc_GetCallInfo(CALLINFOTYPE)** returns “UNKNOWN BILLING”.

1.27.2 Supported Boards

DM3

The following Dialogic® DM3 Boards support this feature:

- Dialogic® DM/V-A Media Boards
- Dialogic® DM/V-B Media Boards

Springware

The following Dialogic® Springware Boards support this feature:

- Dialogic® D/300JCT-E1 Media Boards
- Dialogic® D/600JCT-1E1 Media Boards
- Dialogic® D/600JCT-2E1 Media Boards

1.27.3 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the Dialogic® Global Call API, see the following documents:

- *Dialogic® Global Call API Library Reference*
- *Dialogic® Global Call API Programming Guide*

For features specific to E1 (R2) technology, see the following documents:

- *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*
- *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*

1.28 Runtime Control of Double Answer for R2MF

With this Service Update, a connection method called double answer is now supported for rejecting collect calls on a call-by-call basis.

1.28.1 Feature Description

Currently, double answer signaling is statically enabled or disabled by setting the **CDP_DOUBLE_ANSWER_FLAG** parameter in the CDP file. However, this setting applies to all the calls on the channels and cannot be controlled on a call-by-call basis.

With this new feature, double answer can be triggered on a call-by-call basis by issuing **gc_AnswerCall( )** with the number of rings ORed with a new define, **GC_DBL_ANSWER** (0x100).

Notes: ***1.*** The double answer feature must be disabled (disabled by default) in the CDP file. If the double answer feature is enabled by setting the CDP_DOUBLE_ANSWER_FLAG parameter in the CDP file, then there will be no application control of this feature on a call-by-call basis (this feature will always be triggered).

2. If **gc_AnswerCall( )** is issued with the number of rings ORed with **GC_DBL_ANSWER** on a protocol that does not support double answer functionality, there will be no error reported as there is no range checking being done in the PDK protocols for the number of rings. The expected behavior is that while the inbound side is busy generating the ring back tone (>= 256 rings), the remote side will time out and the call will eventually get dropped.

1.28.2 Supported Boards

DM3

The following Dialogic® DM3 Boards support this feature:

- Dialogic® DM/V-A Media Boards
- Dialogic® DM/V-B Media Boards

Springware

The following Dialogic® Springware Boards support this feature:

- Dialogic® D/300JCT-E1 Media Boards
- Dialogic® D/600JCT-1E1 Media Boards

- Dialogic® D/600JCT-2E1 Media Boards

1.28.3 Example Code

```
#include <stdio.h>
#include <srllib.h>
#include <gclib.h>
#include <gcerr.h>
/*
* Assume the following has been done:
* 1. Opened line devices for each time slot on DTIB1.
* 2. Wait for a call using gc_WaitCall()
* 3. An event has arrived and has been converted to a metaevent
* using gc_GetMetaEvent() or gc_GetMetaEventEx() (Windows)
* 4. The event is determined to be a GCEV_OFFERED event
*/

int answer_call(int num_rings, int dbl_answ_flag)
{
       CRN crn; /* call reference number */
       GC_INFO gc_error_info; /* GlobalCall error information data */
       int rings = 0;

       /*
       * Do the following:
       * 1. Get the CRN from the metaevent
       * 2. Proceed to answer the call as shown below
       */
       crn = metaevent.crn;

       /*
       * Answer the incoming call. Check the dbl_answ_flag to determine
       * if double answer should be triggered or not
       */
       if (dbl_answ_flag)
           rings = num_rings | GC_DBL_ANSWER;
       else
           rings = num_rings;

       if (gc_AnswerCall(crn, rings, EV_ASYNC) != GC_SUCCESS) {
           /* process error return as shown */
           gc_ErrorInfo( &gc_error_info );
           printf ("Error: gc_AnswerCall() on device handle: 0x%lx, GC ErrorValue: 0x%hx - %s,
               CCLibID: %i - %s, CC ErrorValue: 0x%lx - %s\n",
               metaevent.evtdev, gc_error_info.gcValue, gc_error_info.gcMsg,
               gc_error_info.ccLibId, gc_error_info.ccLibName,
               gc_error_info.ccValue, gc_error_info.ccMsg);
           return (gc_error_info.gcValue);
       }

       /*
       * gc_AnswerCall() terminates with GCEV_ANSWERED event
       */
       return (0);
}
```

1.28.4 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the Dialogic® Global Call API, see the following documents:

- *Dialogic® Global Call API Library Reference*
- *Dialogic® Global Call API Programming Guide*

For features specific to E1 (R2) technology, see the following documents:

- *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*
- *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*

1.29 Enhanced ISDN Trace Functionality for DPNSS Tracing

With the Service Update, the Dialogic® ISDNtrace tool now provides functionality to support DPNSS tracing.

1.29.1 Feature Description

This feature enhances the existing ISDN tracing functionality so that you can capture D-channel signaling information into an ASCII text readable form. The DPNSS tracing uses the same command line options already available in the ISDNtrace tool.

1.29.2 Sample DPNSS Trace Output

The following DPNSS sample trace shows ISRM(C) and NAM messages:

```
PROTOCOL TYPE : PRI DPNSS B-End
TRACE START TIME (MM/DD/YYYY) : 2/25/2003, 14:27:52.52

                   Time Stamp : 2/25/2003, 14:28:24.252
                   TRANSMIT
                   Timeslot 01
                   UI(COMMAND) Sequence 0(0x03)
                   Initial Service Request Msg-Complete(0x00)
                     Service Indicator Code
                     1:  0------- Extension Bit
                         -001---- Type of Information
                         ----0000 Speech/Data Rate
                      Selection Field
                         *1#*50*8080808#*58*aziz#132838

Time Stamp : 2/25/2003, 14:28:24.252
RECEIVE
Timeslot 01
UI(RESPONSE) Sequence 0(0x03)


                   Time Stamp : 2/25/2003, 14:28:26.702
                   TRANSMIT
                   Timeslot 31
                   UI(RESPONSE) Sequence 0(0x03)
```

```
Time Stamp : 2/25/2003, 14:28:26.702
RECEIVE
Timeslot 31
UI(COMMAND) Sequence 0(0x03)
Number Acknowledge Msg(0x09)
Indication Field
   *128A*32216070#*6#*50*32205505#
```

1.29.3 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For further information about the ISDNtrace tool, see the *Dialogic® System Software Diagnostics Guide*.

1.30 Notification of Layer 1 Alarm Events on SS7 Boards

With the Service Update, the support for alarm notification has been added for Dialogic® SS7 Boards. By adding support for alarm notification, applications are able to better determine which devices are available for making and receiving calls, or enabling/disabling voice activity.

For further information about this feature, see the *Dialogic® Global Call SS7 Technology Guide*.

1.31 Global Call Support for Time Slots on Dialogic® SS7 Boards Running in DTI Mode

With the Service Update, Dialogic® Global Call Software works with Dialogic® SS7 Boards that include trunks not configured for SS7 signalling (DTI mode); i.e., all the time slots on these trunks operate in clear channel mode.

For further information about this feature, see the *Dialogic® Global Call SS7 Technology Guide*.

1.32 Time Stamp for Tone-On/Off Events

With the Service Update, a new time stamp has been added to the existing DE_TONEON and DE_TONEOFF events. A new TN_TIMESTAMP structure has been added to the device header file *dxxxlib.h*. This time stamp is used to associate, or group, certain tones in order to detect a particular country tone made up of two or more defined tone templates.

1.32.1 Feature Description

To test the various tones from various countries, the Tone-On/Off Call Status Transition (CST) event data have been modified to add a time stamp structure to the end of the TN_INFO structure. The CST event data are obtained by calling **sr_getdatalen( )** and **sr_getevtdatap( )**. A new structure, TN_TIMESTAMP, is in the device header file, *dxxxlib.h*. If the event is for Tone-On, then the time stamp represents the tone-on time, and if the event is for Tone-Off, then it represents the tone-off time.

The Tone-On/Tone-Off messages are extended to add the “start time” and the “stop time,” respectively. These time stamps are used by the customer application to calculate the Tone-On/Tone-Off duration (cadence).

1.32.2 Supported Boards

The following boards support this feature:

- Dialogic® DM/V2400A Media Boards

1.32.3 Structure

TN_TIMESTAMP is as follows:

```
// Tone ON/OFF time stamp
typedef struct {

    unsigned long  tn_TimeStamp;   /* Time stamp for tone on/off event. The time stamp is in
                                     milliseconds from when the firmware was downloaded on the
                                     board. There is no co-relation to the system time. It wraps
                                     around every ~149 hours. */
} TN_TIMESTAMP;
```

Scenario

When a Tone-On CST event is received, the application gets the CST event data with the **sr_getdatalen( )** and **sr_getevtdatap( )** functions, as usual. The application then applies the TN_TIMESTAMP structure to the event data and obtains the time stamp of the tone-on event or tone-off event. The TN_TIMESTAMP structure is appended to the end of the TN_INFO structure. The CST event data comprises the DX_CST, TN_INFO, and TN_TIMESTAMP structures.

Sample

The following is an example for Tone-On. Tone-Off is done the same way.

```
DX_CST *datap;
TN_INFO *tonep;
TN_TIMESTAMP *tsp;
long timestamp; // time stamp in ms units
```

```
switch(sr_getevttype(ehandle))
{
     case TDX_CST:
         datap = (DX_CST *) sr_getevtdatap(ehandle);

         if (datap->cst_event == DE_TONEON)
         {
             tonep = (TN_INFO*)(datap+1); // tone structure starts at end of CST structure
             tsp = (TN_TIMESTAMP*)(tonep+1); // time stamp structure starts at end of
                                                                 TN_INFO structure.
             timestamp = tsp->tn_TimeStamp; // get the time stamp
}

break;
.
.
```

1.32.4 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the Dialogic® Standard Runtime Library and Voice APIs, see the following documents:

- *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: The online bookshelf has not been updated for this feature, so the manuals above do not contain information relating to this feature.

1.33 New Fax Parameter for Modem Receive Level

With the Service Update, a new fax parameter, **FC_MDM_RX_LVL**, has been added to allow setting of the fax modem receiver sensitivity from -43 dBm to -47 dBm. This parameter is supported on Dialogic® Springware Fax Boards only.

The **FC_MDM_RX_LVL** parameter is set with the **fx_setparm( )** function and can be retrieved with the **fx_getparm( )** function. Valid settings are:

- 0 = -43 dBm
- 1 = -44 dBm
- 2 = -45 dBm
- 3 = -46 dBm (default)
- 4 = -47 dBm

For example:

```
int parmValue = 3;

// set parameter
fx_setparm(DeviceHandle, FC_MDM_RX_LVL, (void*)&parmValue);

// get current setting
fx_getparm(DeviceHandle, FC_MDM_RX_LVL, (void*)&parmValue);
```

For further information about the **fx_setparm( )** and **fx_getparm( )** functions, see the *Dialogic® Fax Software Reference*.

1.34 Ability to Send and Receive DPNSS End to End Messages

With the Service Update, the user has the ability to send and receive the entire raw Digital Private Network Signalling System (DPNSS) end to end message (EEM) using API control on Dialogic® DM3 Boards. A generic mechanism enables the user to add DPNSS supplementary services (like Single/Dual channel transfer services, Call Diversion, and Call Waiting) without needing outside support for those services first. This feature is only supported on ISDN DPNSS loads.

1.34.1 Feature Description

This feature enables the application to:

- Enable GCEV_EXTENSION through **gc_SetConfigData( )** (for enabling the event).
- Send raw DPNSS EEM through **gc_SndMsg( )** with a new message type (for sending the event).
- Receive raw DPNSS EEM through GCEV_EXTENSION event on DM3 Boards (for receiving the event).

The user has the ability to send and receive raw EEM frames. The user can extract the content of the EEM message and take the appropriate action when he/she receives any of the messages. The API is allowed in any intermediate call state. A majority of DPNSS supplementary services can be supported and the user does not need to request outside support for every new service that is being planned for the future.

EEM frames are of two types:

- EEM(I) - an end to end message (incomplete)
- EEM(C) - an end to end message (complete)

An EEM(C) is typically used, but if the size of the message exceeds 45 bytes in length, it can be split up into multiple EEM(I) messages, with a final piece of the message completed by an EEM(C).

Note: The application tracks the receipt of the various EEM(I) frames and reassembles them together to form the entire final EEM(C) message.

New Parameters

The feature is implemented using the Dialogic® Global Call **gc_SetConfigData( )** function and the GCTGT_CCLIB_CHAN parameter set. New extension event IDs define the receive raw DPNSS EEM through the GCEV_EXTENSION event. This unsolicited event can be enabled or disabled through **gc_SetConfigData( )**. The ISDN *dm3cc_param.h* header file is updated with the following:

- New extension ID: DM3CC_EXT_EVT_RAWEEM.
- New bit mask: EXTENSIONEVT_RAWEEM - Use to enable or disable the GCEV_EXTENSION event for DPNSS Raw EEM.
- New set ID: CCSET_RAWEEM.
- New PARMID: CCPARM_RAWEEM_DATA.

The ISDN *isdndef.h* header file is updated with SndMsg_RawEEM for the application to send raw EEM through **gc_SndMsg( )**.

Generated Events

GCEV_EXTENSION

The **gc_SetConfigData( )** function is issued to enable this functionality and the following notification event is generated for the application:

EXTENSIONEVT_RAWEEM
: Use to enable or disable the GCEV_EXTENSION event for DPNSS raw EEM.

The **gc_SndMsg( )** function is issued and the following notification events may be generated for the application:

CCSET_RAWEEM
: Receives raw EEM.

GCEV_TASKFAIL
: Indicates failure, for example, in case the information element (IE) has state change information in that the raw data contains an invalid IE or the raw data is 45 bytes.

Error Codes

The following success code is generated by the **gc_SetConfigData( )** function:

GC_SUCCESS
: Success. The signal type change has been implemented.

The following error code is generated by the **gc_SndMsg( )** function:

GCEV_TASKFAIL
: Task failed. The firmware will return Std_MsgError if the call state is not transferring because there is an invalid IE (call state changing), or the raw data is 45 bytes.

1.34.2 Enabling/Disabling GCEV_Extension Event

For the **gc_SetConfigData( )** function, the bit mask (EXTENSIONEVT_RAWEEM) is saved for use later during GCEV_EXTENSION event generation. The **gc_SetConfigData( )** is set on a channel basis and has the target type set as GCTGT_CCLIB_CHAN. The general procedure is to call the **gc_SetConfigData( )** function to implement the change as follows:

```
gc_SetConfigData(GCTGT_CCLIB_CHAN.EXTENSIONEVT_RAWEEM)
```

1.34.3 Successfully Sending and Receiving Raw DPNSS EEM

The **gc_SndMsg(MsgType, GC_IE_BLK)** is used to send the raw DPNSS EEM like other DPNSS Supplementary Services (for example, Intrusion(SndMsg_Intrude), Diversion(SndMsg_Divert), NSI (SndMsg_NSI), etc.). The general procedures are as follows:

- To send an End to End Complete message, call the **gc_SndMsg( )** function with the msg_type parameter set to SndMsg_RawEEM. The first byte of the data portion (i.e., ie_Blk.data[0]) must contain 0x22 to indicate that it is an EEM(C) message. To receive the message, enable the GCEV_EXTENSION event.
- To send an End to End Incomplete message, call the **gc_SndMsg( )** function, with the msg_type parameter set to SndMsg_RawEEM. The first byte of the data portion (i.e., ie_Blk.data[0]) must contain 0x23 to indicate that it is an EEM(I) message. To receive the message, enable the GCEV_EXTENSION event.

The message is successfully received if no GCEV_TASKFAIL event is received at the user application.

Notes: ***1.*** The first byte in the GC_IE_BLK is the spec defined Message ID for an EEM(I) or EEM(C) message.

2. Certain supplementary information strings that may affect the firmware call state are not allowed in the raw EEM payload. Specifically not allowed are the HOLD-REQ string or 60B, and the RECON string or 61. If either of these strings is present, the application will receive a GCEV_TASKFAIL event.

3. The total length of the raw EEM payload allowed is 45 bytes: 1 byte specifies the EEM type, which is EEM(C) or EEM(I), and 44 bytes are allowed for supplementary information strings encoded using the Backus Naur format and conforming to the DPNSS standard BTNR 188.

1.34.4 Sample Code

The following are samples of code for sending raw EEM and receiving raw EEM.

To Send Raw EEM

```
int send_message(CRN crn)
{
 int      gc_err;          /* GlobalCall Error Code */
    int      cclibid;         /* Call Control library ID */
    long     cclib_err;       /* Call Control Error Code */
    char     *msg;            /* Error Message */
    LINEDEV  ldev;            /* Line device */
 char      str[MAX_STRING_SIZE];

 GC_IE_BLK gcIEBlk;
 IE_BLK ie_Blk;

 memset((unsigned char *)&ie_Blk, 0, sizeof(IE_BLK));

 gcIEBlk.gclib = NULL;
 gcIEBlk.cclib = &ie_Blk;
 ie_Blk.length =  7;       //length of the raw DPNSS EEM data
  /* EEM(C) = 0x22, EEM(I) = 0x23 */
 ie_Blk.data[0] = 0x22;   // raw DPNSS EEM data
 ie_Blk.data[1] = '*';   // raw DPNSS EEM data
 ie_Blk.data[2] = '1';   // raw DPNSS EEM data
 ie_Blk.data[3] = '1';   // raw DPNSS EEM data
 ie_Blk.data[4] = '0';   // raw DPNSS EEM data
 ie_Blk.data[5] = 'B';   // raw DPNSS EEM data
 ie_Blk.data[6] = '#';   // raw DPNSS EEM data

      if(gc_CRN2LineDev(crn, &ldev) != GC_SUCCESS) {
        gc_ErrorValue(&gc_err, &cclibid, &cclib_err);
        gc_ResultMsg(cclibid, cclib_err, &msg);
  sprintf(str, "Error on Device handle : 0x%lx ",ldev);
  printandlog(0, GC_APICALL, NULL, str, 0);
        return(cclib_err);
    }

    if(gc_SndMsg(ldev, crn, SndMsg_RawEEM, &gcIEBlk) != GC_SUCCESS) {
        gc_ErrorValue(&gc_err, &cclibid, &cclib_err);
        gc_ResultMsg(cclibid, cclib_err, &msg);
        sprintf(str, "Error on Device handle : 0x%lx ",ldev);
  printandlog(0, GC_APICALL, NULL, str, 0);
        return(cclib_err);
    }
    return 0 ;
}
```

To Enable the GCEV_EXTENSION Event to Receive Raw EEM Events

```
int EnableRawEEMInformation(int DeviceHdl)
{
 GC_PARM_BLKP pParmBlock = NULL;
 long requestID;
 char      str[MAX_STRING_SIZE];

 int iRetCode = gc_util_insert_parm_val(&pParmBlock, CCSET_EXTENSIONEVT_MSK,
  GCACT_ADDMSK, sizeof(long), EXTENSIONEVT_RAWEEM);

 int rc = gc_SetConfigData(GCTGT_CCLIB_CHAN, DeviceHdl, pParmBlock,0,
  GCUPDATE_IMMEDIATE, &requestID, EV_ASYNC);

 if(rc != GC_SUCCESS) {
  sprintf(str, "failed to set evt mask");
  printandlog(0, GC_APICALL, NULL, str, 0);
  return GC_ERROR;
 } else {
  sprintf(str, "gc_SetConfigData() called - Raw EEM event  reception enabled");
  printandlog(0, GC_APICALL, NULL, str, 0);
 }

 gc_util_delete_parm_blk(pParmBlock);
 return 0;
}
```

To Receive Raw EEM and Extract Raw DPNSS Data

```
void process_event(void)
{ ....
  ....
  ....
 switch (evttype)
 {
  case GCEV_EXTENSION:
   ExtractDPNSSInfo(pline, &metaevent);
   break;

 }
}

void ExtractDPNSSInfo(struct channel *pline,METAEVENT *metaeventp)
{
 GC_PARM_BLKP gcParmBlkp = NULL;
 GC_PARM_DATAP t_gcParmDatap = NULL;
 EXTENSIONEVTBLK *ext_evtblkp = NULL;

 GC_IE_BLK * t_gcIEBlk = NULL;
 IE_BLK * ie_blk = NULL;
 char rawData[100];
 char str[MAX_STRING_SIZE];
 int i=0;

 ext_evtblkp = (EXTENSIONEVTBLK *)metaeventp->extevtdatap;
 gcParmBlkp = &ext_evtblkp->parmblk;

 sprintf(str, "Received GCEV_EXTENSION event with ExtID = 0x%x",ext_evtblkp->ext_id);
 printandlog(0, GC_APICALL, NULL, str, 0);
 while (t_gcParmDatap = gc_util_next_parm(gcParmBlkp, t_gcParmDatap))
 {
 switch (t_gcParmDatap->set_ID)
  {
   case CCSET_RAWEEM:
    switch(t_gcParmDatap->parm_ID)
    {
     case CCPARM_RAWEEM_DATA:
         t_gcIEBlk =  (GC_IE_BLK *)t_gcParmDatap->value_buf;
      ie_blk =  t_gcIEBlk -> cclib;
      memcpy(rawData, ie_blk->data,ie_blk->length);

      sprintf(str, "RAWEEM_DATA : length = %d\n", ie_blk->length);
      printandlog(0, GC_APICALL, NULL, str, 0);
      memset(str, 0, MAX_STRING_SIZE);

      for (i=0; i < ie_blk->length; i++)
      {
       if((i!=0) && (isascii(rawData[i]))) {
        printf(str, "%c ", rawData[i]);

        fprintf(port[0].log_fp, "%c ", rawData[i]);
       }

       else {
        printf(str, "%02X ", rawData[i]);
        fprintf(port[0].log_fp, "%02X ", rawData[i]);
       }
      }
      printf("\n");
      fprintf(port[0].log_fp, "\n ");
     break;
     default:
      sprintf(str, "Unknown PARM ID");
      printandlog(0, GC_APICALL, NULL, str, 0);
     break;
    }
    break;
   default:
    sprintf(str, "Unknown SET ID");
    printandlog(0, GC_APICALL, NULL, str, 0);
   break;
  }
 }

}
```

1.34.5 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the Dialogic® Global Call API, see the following documents:

- *Dialogic® Global Call API Programming Guide*
- *Dialogic® Global Call API Library Reference*

For features specific to ISDN technology, see:

- *Dialogic® Global Call ISDN Technology Guide*

Note: The online bookshelf has not been updated for this feature, so the *Dialogic® Global Call ISDN Technology Guide* does not currently include information about sending and receiving raw DPNSS end to end messages.

1.35 Enhancements to the Configuration Process

With the Service Update, enhancements have been made to simplify the configuration process:

- PDK Configuration Property Sheet
- Automatic FCD File Generation

1.35.1 PDK Configuration Property Sheet

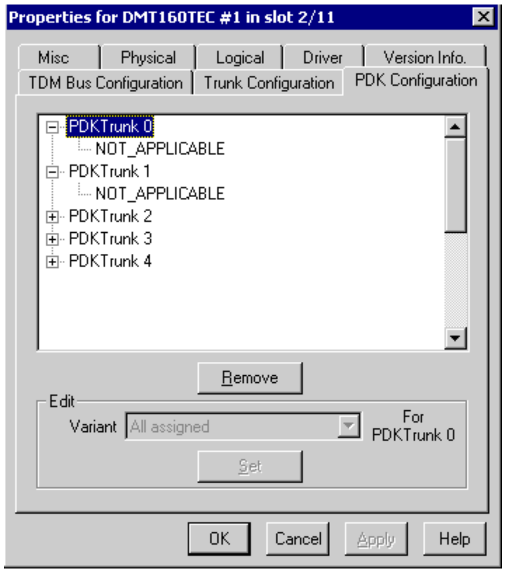
With the Service Update, a new PDK Configuration property sheet in the Dialogic® Configuration Manager (DCM) allows you to choose country dependent parameter (CDP) files for T1 trunks that use the CAS protocol or for E1 trunks that use the R2MF protocol. For each trunk selected, a list of applicable CDP file variants is presented, allowing you to assign a specific CDP file to that trunk.

The new PDK Configuration property sheet replaces the “Downloading the Protocol and CDP File on a Windows System” procedure documented in the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*. With this new feature, it is no longer necessary to set up the *pdk.cfg* file to download the protocol and CDP file.

Note: This procedure only applies to boards having network interfaces, and to trunks that are configured for the CAS or R2MF protocols.

1. From the DCM main window, highlight the board you wish to configure and choose **Configure Device** from the **Device** drop-down menu. The property sheets for this board will appear.

2. Select the **PDK Configuration** property sheet.



PDKTrunk 0
NOT_APPLICABLE
PDKTrunk 1
NOT_APPLICABLE
PDKTrunk 2
PDKTrunk 3
PDKTrunk 4

Variant: All assigned
For PDKTrunk 0

3. If all of the trunks on the board have been configured for either the CAS or R2MF protocol, and you wish to assign the same country dependent parameter (CDP) variant

file (other than the default value) to all trunks on the board, highlight PDKTrunk 0. Otherwise proceed to step 4.

3a. From the Variant drop-down menu, select a CDP variant file by highlighting the file and clicking the **Set** button.

Note: If you wish to remove a previously assigned CDP variant file, highlight the variant under that trunk in the window and click the **Remove** button.

3b. Repeat this step for each additional CDP variant file you wish to assign to all of the trunks on this board.

Note: When multiple CDP file variants are assigned to a trunk, an application can dynamically change variants on that trunk. In this case, for a given trunk, it is the last variant in the list that is taken as the default. For example, if one looks at a given trunk in the DCM/PDK Configuration window and sees a list:

```
PDKTrunk#
            pdk_ar_r2_io.cdp
            pdk_cn_r2_io.cdp
            pdk_be_r2_io.cdp
```

Then, the “pdk_be_r2_io.cdp” will be taken as the default.

3c. Click the **Apply** button and then click the **OK** button to return to the DCM main window.

4. If not all trunks on the board have been configured for CAS or R2MF, or if you wish to assign different CDP variant files to individual trunks on a trunk-by-trunk basis:

4a. Highlight the trunk to which you wish to assign a CDP variant file.

4b. Choose a CDP variant file from the Variant drop-down list and click the **Set** button.

Note: If you wish to remove a previously assigned CDP variant file, highlight the variant under that trunk in the window and click the **Remove** button.

4c. Repeat steps 4a and 4b for each trunk on the board that you wish to assign CDP variant files.

4d. Click the **Apply** button and then click the **OK** button to return to the DCM main window.

See the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for information about configuring the parameters contained in an individual CDP file.

The protocol package is included with the system software.

1.35.2 Automatic FCD File Generation

With the Service Update, the fcdgen utility is no longer required to generate the FCD file. When you download a PCD file and its corresponding CONFIG file to a board, the FCD file is automatically generated and also downloaded to the board. The FCD file is also copied into the data directory.

With this enhancement to the configuration process, it is no longer necessary to use the fcdgen utility to generate a modified FCD file. When you modify a CONFIG file, the modified FCD file is automatically created when the PCD file and CONFIG file are downloaded to the board.

1.35.3 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For detailed information about configuring DM3 boards, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Note: The online bookshelf has not been updated for these features, so the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* does not currently include information about the PDK Configuration property sheet or about automatic FCD file generation.

1.36 New Option for dm3post Utility

With the Service Update, the dm3post diagnostic utility now provides an option to run POST on a chassis level. By using the chassis option (-c), dm3post will retrieve the results of the last run POST for all Dialogic® DM3 Boards in the chassis. By using the chassis option with the reset (-r) option, you can run POST on all DM3 Boards in the system.

When using the chassis option, it is not necessary to provide the bus and slot numbers. Any option other than the reset option will be ignored when using the chassis option. In addition to output on the screen, more detailed output is logged to a log file, dm3post.log, by default.

For more information about the dm3post utility, see the *Dialogic® System Software Diagnostics Guide*.

1.37 New OAMIPC Mechanism Replaces CORBA

With the Service Update, a new OAMIPC mechanism replaces CORBA and CORBA will no longer be used. This mechanism changes the binary size of the oam binaries. The *ooc* directory under the *dialogic* directory will be removed if you are doing an upgrade install, or the *ooc* directory will not be installed in case of a new installation.

As part of the new OAMIPC, the TCP Port List for new installation and upgrade installation has changed. For new installations, you will see the Welcome screen with the message "Exclusive access to TCP ports 12001, 12004-5 for the loopback interface, and port 12002 for all network interfaces is required so ensure that these ports are available on your system."

If you are performing an upgrade installation, you will not see the message, but you will still have to check that these TCP ports are available before you perform the upgrade.

Refer to the “Checking TCP Port Availability” section in the *Dialogic® System Release 6.0 PCI for Windows Software Installation Guide*.

1.38 Support for Mixed ISDN and Clear Channel on Additional Dialogic® DM3 Boards

With the Service Update, the ability to mix ISDN (Net5) and clear channel on the same board on a trunk by trunk basis is now supported on the following boards:

- Dialogic® DM/IP601-2E1-PCI IP Boards
- Dialogic® DM/N1200-4E1-PCI Digital Telephony Interface Boards
- Dialogic® DM/V600-4E1-PCI and DM/V1200-4E1-PCI Voice Boards

With the Service Update, a clear channel media load is now supported on the following boards:

- Dialogic® DM/IP301-1E1-PCI IP Boards

1.38.1 Feature Description

Predefined sets of features for Dialogic® Boards are provided in media loads. A media load consists of a configuration file set (PCD, FCD, and CONFIG files) and the associated firmware that is downloaded to the board. The new PCD/FCD/CONFIG files for supporting mixed ISDN and clear channel are:

Board	Media Load	Filename
DM/IP601-2E1-PCI	ML2	ipvs_evr_2isdn_net5_ts16_311.*
DM/N1200-4E1-PCI	N/A	4x0_isdn_net5_ts16.*
DM/V600-4E1-PCI	ML1	ml1_4x2_net5_ts16.*
DM/V1200-4E1-PCI	ML1	ml1_qs_net5_ts16.*
DM/IP301-1E1-PCI	ML11	ipvs_evr_ts16_ml11_311.*

Note: DM/IP301-1E1-PCI supports clear channel media load only.

For a description of the features provided in ML1, ML2, and ML11, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*. Note that ML2 supports only the basic coders on the PSTN side for the DM/IP601-2E1-PCI Board.

Trunks that are configured for clear channel will include 31 bearer channels. The application should be aware that there will be 31 devices detected on these trunks.

No additional voice channels will be provided to accommodate the additional bearer channel(s). For example, when in clear channel mode on a DM/IP301-1E1-PCI Board, the PSTN side of the board will have 31 bearer channels and no signaling channel, but only 30 voice channels.

When in clear channel mode, the time slot 16 mapping is as follows:

- NI TS1-NI TS15 map to dtiB1T-dtiB1T15
- NI TS17-NI TS31 map to dtiB1T16-dtiB1T30
- NI TS16 maps to dtiB1T31

Note: The application should expect a GCEV_UNBLOCKED event when **gc_open( )** is called on trunks set for clear channel.

1.38.2 Configuring the Software

In the CONFIG file, the **Signaling Type** parameter (0x1602) allows you to configure a trunk for ISDN or clear channel. All trunks are set to Net5 by default. To switch a trunk to clear channel, the **SignalingType** parameter should be changed to 6 (Clear) in the [lineAdmin] section for that trunk in the CONFIG file. All CONFIG file parameters are described in detail in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

The new configuration files can be selected by using the Dialogic® Configuration Manager (DCM). This procedure, which must be performed before the boards are started, is also described in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

1.38.3 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For detailed information about configuring Dialogic® DM3 Boards, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Note: The online bookshelf has not been updated for this feature, so the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* does not currently include information about these new configuration files.

1.39 Detection of Unsupported Boards

With the Service Update, if an unsupported board (i.e., a board that is not supported in Dialogic® System Release 6.0 PCI for Windows®) is detected by the software, an error message about the unsupported board will appear in a log file in *dialogic/log* directory with the following filename *rtf*.txt* (for example, *rtflog-10072005-14h47m25.639s.txt*). System Release 6.0 PCI Windows will not prevent you from installing an unsupported board. However, the Dialogic® Configuration Manager (DCM) will not show any unsupported boards.

1.40 PBX Integration Support for Nortel BCM

The Service Update adds support for Nortel Business Communications Manager (BCM) when using the Dialogic® D/82JCT-U Board. The level of support is comparable to that provided for the Nortel M7324 telephone as described in the *Dialogic® PBX Integration Board User's Guide* with the following notable exceptions:

- The information in Section 4.5.1, "Nortel Norstar Programming Requirements" does not apply when using the Nortel BCM. See the Nortel BCM Programming Requirements section below for equivalent information.
- The Message Waiting Indication (MWI) and Calling Party ID (CPID) features, which are supported for the Nortel M7324 telephone, are not supported for the Nortel BCM. Accordingly, the behavior of the **dx_dial( )** and **d42_gtcallid( )** functions are as follows:
  - If **dx_dial( )** is called to set MWI, it does not return an error code. The MWI functionality simply does not take place.
  - If **d42_gtcallid( )** is called, the function returns -1. If the **ATDV_LASTERR( )** function is called after **d42_gtcallid( ),** it returns ED42_UNSUPPORTED.

Nortel BCM Programming Requirements

When using a Dialogic® PBX Integration Board with the Nortel Business Communication Manager (BCM), there are specific switch programming requirements. The user must ensure that these features are set correctly (and assigned to the right keys) so that the PBX Integration Board and the APIs provided by Dialogic function correctly. The following instructions demonstrate how to configure a single extension. Additional extensions can be configured in a similar manner.

Note: This section applies to the configuration of Nortel BCM, version 3.7. Different versions of the Nortel BCM software may not function exactly as described here.

Configuration of the Nortel BCM begins by logging in to the Unified Manager. After login, click the **Wizards** menu option to open the **Setup and Management Wizards** menu. Next, click the **Add Users** menu option. Following this selection, the user is presented with a series of configuration menus. The following tables and text identify the configuration menus in the order in which they are presented. Menu options shown in **bold** text indicate parameters that should be changed from their default values.

Add Users Menu	
DN Type:	Set DNs
Set Model:	**M7324**
Choose one or more DNs:	***<select DNs to program>***
Use settings:	Defined in This Wizard

The next table enables the user to define individual names for each extension. If desired, enter a name next to the corresponding extension. Otherwise, leave the name field blank.

The next table prompts the user about the installation of CallPilot Voice Messaging. Refer to Nortel BCM documentation for the proper configuration of the CallPilot voice mail system.

Line Access Menu	
Prime Line:	I/C
Intercom Keys:	**1**

Capabilities Menu	
DND on busy:	N
Handsfree:	**Standard**
HF answerback:	**N**
Pickup group:	None
Page zone:	**None**
Paging:	**N**
Direct dial:	Set 1
Priority Call:	N
Auto hold:	**N**
Aux ringer:	N
Allow redirect:	N
Redirect ring:	Y

Call Forward Menu	
Forward no answer to:	***<configure for extension immediately following the extension being configured>***
Forward on busy to:	***<configure for extension immediately following the extension being configured>***

Hotline Menu	
Type:	None

User Preferences Menu	
Set Model:	M7324
Call Log Options:	*<refer to Nortel BCM documentation>*
Dialing Options:	*<refer to Nortel BCM documentation>*
Language:	English
Contrast:	*<choose desired contrast>*
Ring Type:	*<choose desired ring type>*
Perform Button Programming:	**Yes**

Selecting “Yes” in the **Perform Button Programming** field causes a button programming menu to be displayed. Memory keys 00, 01, and 03 must be programmed as follows:

Button Programming Menu	
Button	**Functionality**
Memory Button 00	Handsfree/Mute
Memory Button 01	Intercom
Memory Button 03	Transfer

Memory buttons 00 and 01 are automatically programmed if the **Handsfree** option (in the Capabilities menu) is set to “Standard” and the **Intercom Keys** option (in the Line Access Menu) is set to “1” respectively.

To program button 03, select **Feature** in the drop down list corresponding to button 03 in the button programming menu. Another drop down list is displayed. Select feature **70 (transfer)** from the new drop down list.

Click **Next** after memory button programming is complete. A settings summary page is displayed. Click **Apply** to program the switch according to the new configuration.

1.41 Enhancements to Runtime Trace Facility (RTF) Logging

The Service Update provides enhancements to Runtime Trace Facility (RTF) logging. The RTF tool provides a mechanism for tracing the execution path of Dialogic® runtime libraries. The trace information can be captured in a log file or sent to a system-specific debug stream (e.g., debug console on Windows®). The resulting log file/debug stream output helps troubleshoot runtime issues for applications that are built with Dialogic® software.

For detailed information about RTF logging, see the *Dialogic® System Software Diagnostic Guide*.

1.42 Dynamically Retrieving and Modifying Selected Protocol Parameters When Using Dialogic® DM3 Boards

With the Service Update, the ability to dynamically retrieve or modify certain protocol-specific parameter values stored by the Dialogic® DM3 Board firmware is provided. The boards that support this feature are:

- Dialogic® DM/V-A Media Boards
- Dialogic® DM/V-B Media Boards

1.42.1 Feature Description

This feature allows a user to dynamically (at runtime) retrieve and/or modify the following parameter values:

- Protocol ID
- CAS signal definitions
- CDP variable values
- Line type (E1_CRC, D4, ESF) and coding (B8ZS, HDB3, AMI) for a trunk
- Protocol for a trunk

Some typical use cases for this feature are as follows:

- When a new system is configured and then provisioned by a new carrier, protocol parameters, such as wink settings, need to be tweaked before a call can be placed using the new switch. Currently, the configuration file has to be manually edited for every change and the firmware re-downloaded for the changes to take effect. The provision of this API to perform these changes at runtime alleviates the need to manually edit configuration files and subsequently re-download the firmware.
- When using ISDN protocols, the ability to dynamically determine the protocol running on a particular span is important in determining whether features such as Two-B Call Transfer (TBCT) or Overlapped Sending can be supported.

This feature is implemented using the Dialogic® Global Call Run Time Configuration Management (RTCM) facility, which uses the Global Call **gc_GetConfigData( )**, **gc_SetConfigData( )**, and **gc_QueryConfigData( )** functions. For general information on the operation of the Global Call RTCM facility, see the *Dialogic® Global Call API Programming Guide*. Details on how to dynamically configure the parameter types mentioned above are provided in the following sections.

1.42.1.1 Prerequisites for Feature Use

Creating a dm3enum.cfg File

Before this feature can be used, it must be enabled. To enable the feature, a new configuration file must be created. The name of the configuration file is *dm3enum.cfg* and it must be stored in the *Dialogic\cfg* directory. The *dm3enum.cfg* file determines the boards on which this dynamic protocol configuration feature is to be enabled.

The syntax of the commands that can be included in a *dm3enum.cfg* file are:

board <n>
Specifies a logical board (<n>) on which this feature is to be enabled.

board (startBd endBd)
Specifies a range of boards on which this feature is to be enabled.

Notes:1. The “board” command word can be abbreviated to “b”.

2. The startBd value must be less than the endBd value.

Some examples of commands that can be included in a *dm3enum.cfg* file are:

```
board 0
b 1
board (1 3)
b (1 3)
```

The feature generates a number of log files in the *Dialogic\log* directory:

Dm3enumreate.log
: Log file for application that starts dynamic protocol configuration enablement

Protocol_enmurate_board_#.log
: Log file of actual dynamic protocol configuration enablement process for boards that use non-PDK protocols

Pdkenumerate.log_board#
: Log file of actual dynamic protocol configuration enablement process for boards that use PDK protocols

Enabling Protocol Configuration

To enable protocol configuration for PDK protocols, after running the `pdkmanagersetup add` command, run the following command:

```
pdkmanagerregsetup enumerate
```

Then,

- Download the board firmware.
- Run the `devmapdump` utility to check that the protocol information has been loaded (search for CDP_ and CAS_ parameters). If this is not the case, run the `dm3enumerate` utility to load the protocol information manually after each firmware download.

To enable protocol configuration for non-PDK protocols, run the `dm3enumerate` utility to load protocol name information.

1.42.1.2 Retrieving a Protocol ID

DM3 protocol names have the format “lb#pv#:Variant_Name”, where:

- lb# is the logical board ID on a physical DM3 Board
- pv# is the protocol variant ID

Some examples are:

- “lb1pv1:pdk_us_mf_io” - A PDK protocol that is the first protocol variant on logical board 1
- “lb2pv1:isdn_net5” - An ISDN Net5 protocol that is the first protocol variant on logical board 2

- “lb0pv5:analog_loop_fxs”- An analog protocol that is the fifth protocol variant on logical board 0

Note: All characters in protocol names are lowercase.

The protocol ID is assigned by Global Call and the user must obtain the protocol ID prior to accessing any protocol-related data.

The protocol name and ID for a DM3 Board can be obtained by calling the **gc_GetConfigData( )** function on an opened time slot device handle with the following parameter values:

- **target_type** = GCTGT_GCLIB_CHAN
- **target_id** = the line device handle
- **target_datap** = GC_PARM_BLKP parameter pointer, as constructed by the utility function **gc_util_insert_parm_val( )** for protocol ID and **gc_util_insert_parm_ref( )** for protocol name.
- **time_out** = time interval (in seconds) during which data must be retrieved. If the interval is exceeded, the retrieve request is ignored. This parameter is supported in synchronous mode only, and it is ignored when set to 0.
- **request_idp** = pointer to the location for storing the request ID
- **mode** = EV_ASYNC for asynchronous execution or EV_SYNC for synchronous execution. EV_SYNC mode is recommended.

Note: Only time slot objects support the retrieval of the protocol ID and name.

See Section 1.42.1.5, “Sample Code for Getting and Setting CAS Signal Definitions and CDP Variable Values”, on page 120, specifically the **ObtainProtocolIDAndName( )** function, for example code that demonstrates how to retrieve the protocol ID and name.

If the protocol name is known, the protocol ID can be obtained by calling the **gc_QueryConfigData( )** function with the following parameter values:

- **target_type** = GCTGT_GCLIB_SYSTEM
- **target_id** = GC_LIB
- **source_datap** = GC_PARM parameter pointer for storing the protocol name (input)
- **query_id** = Query ID, in this case, GCQUERY_PROTOCOL_NAME_TO_ID
- **response_datap** = GC_PARM parameter pointer for storing the protocol ID (output)

1.42.1.3 Retrieving or Modifying CAS Signal Definitions

This feature enables the user to dynamically retrieve or modify CAS signal definitions. Before the CAS signal definition can be retrieved or modified, the {set ID:parm ID} pair that identifies the signal in the firmware must be retrieved. The datatype of the corresponding parameter value must also be retrieved. The following sections describe the operations relating to CAS signal definitions that can be performed.

Obtaining the {Set ID:Parm ID} Pair for a CAS Signal

Each CAS parameter in a DM3 PDK protocol has a unique {set ID:parm ID} pair, in which the **set ID** represents the component that contains the parameter and **parm ID** represents an internal ID within that component. The set ID is one of a predefined set of values in the *dm3cc_parm.h* file, and the parm ID is assigned by the DM3 firmware at download time. For example, the CAS_ANSWER parameter (which defines a CAS signal) is contained in the CAS component identified by the PRSET_CAS_SIGNAL set ID with the parm ID being assigned internally by the firmware.

Before dynamically retrieving or modifying the value of a CAS parameter in the DM3 firmware, the user must call the **gc_QueryConfigData( )** function to obtain the {set ID:parm ID} pair of the CAS parameter using the parameter name obtained from the CDP file.

The **gc_QueryConfigData( )** function is called with the following parameter values:

- **target_type** = GCTGT_PROTOCOL_SYSTEM
- **target_id** = PDK Protocol ID
- **source_datap** = GC_PARM parameter pointer for storing input CAS parameter name
- **query_id** = Query ID, in this case, GCQUERY_PARM_NAME_TO_ID
- **response_datap** = GC_PARM parameter pointer for storing output {set ID:parm ID} and value type

See Section 1.42.1.5, “Sample Code for Getting and Setting CAS Signal Definitions and CDP Variable Values”, on page 120, specifically the **QueryParmID( )** function, for example code that demonstrates how to retrieve the {set ID:parm ID} pair for a CAS signal.

Note: Obtaining the {set ID:parm ID} pair is a prerequisite to retrieving the definition of a CAS signal or redefining a CAS signal.

Retrieving a CAS Signal Definition

The **gc_GetConfigData( )** function can be used to retrieve the value of CAS parameters in the DM3 firmware. Function parameter values to use in this context are:

- **target_type** = GCTGT_PROTOCOL_SYSTEM
- **target_id** = PDK Protocol ID
- **target_datap** = GC_PARM_BLKP parameter pointer, as constructed by the **gc_util_insert_parm_ref( )** utility function for CAS signal
- **time_out** = time interval (in seconds) during which parameter value must be retrieved. If the interval is exceeded, the retrieve request is ignored. This parameter is supported in synchronous mode only, and it is ignored when set to 0.
- **request_idp** = pointer to the location for storing the request ID, output from Global Call
- **mode** = EV_ASYNC for asynchronous execution or EV_SYNC for synchronous execution. EV_ASYNC mode is recommended.

See Section 1.42.1.5, “Sample Code for Getting and Setting CAS Signal Definitions and CDP Variable Values”, on page 120, specifically the **GetCASSignalDef( )** function, for example code that demonstrates how to retrieve the definition of a CAS signal, in this case the CAS_WINKREV signal.

Setting a CAS Signal Definition

The **gc_SetConfigData( )** function with the following parameter values can be used to set a new definition for a CAS signal in the DM3 firmware:

- **target_type** = GCTGT_PROTOCOL_SYSTEM
- **target_id** = PDK Protocol ID
- **target_datap** = GC_PARM_BLKP parameter pointer, as constructed by the utility function **gc_util_insert_parm_ref( )** for the CAS signal
- **time_out** = time interval (in seconds) during which the parameter value must be updated. If the interval is exceeded, the update request is ignored. This parameter is supported in synchronous mode only, and it is ignored when set to 0.
- **update_cond** = ignored for DM3 PDK protocols
- **request_idp** = pointer to the location for storing the request ID, output from Global Call
- **mode** = EV_ASYNC for asynchronous execution or EV_SYNC for synchronous execution. EV_ASYNC mode is recommended.

See Section 1.42.1.5, “Sample Code for Getting and Setting CAS Signal Definitions and CDP Variable Values”, on page 120, specifically the **SetCASSignalDef( )** function, for example code that demonstrates how to change the definition of a CAS signal, in this case the CAS_WINKREV signal.

1.42.1.4 Retrieving or Modifying CDP Variable Values

This feature enables the user to dynamically retrieve or modify parameter values defined in DM3 PDK protocol country dependent parameter (CDP) files. Before the CDP variable value can be retrieved or modified, the {set ID:parm ID} pair that identifies the CDP variable in the firmware must be retrieved. The datatype of the corresponding CDP variable value must also be retrieved. The following sections describe the operations relating to CDP variable values that can be performed.

Obtaining the {Set ID:Parm ID} Pair for a CDP Variable

Each CDP variable in a DM3 PDK protocol has a unique {set ID:parm ID} pair, in which the **set ID** represents the component that contains the parameter and **parm ID** represents an internal ID within that component. The set ID is one of a predefined set of values in the *dm3cc_parm.h* file, and the parm ID is assigned by the DM3 firmware at download time.

Before dynamically retrieving or modifying the value of a CDP variable in the DM3 firmware, the user must call the **gc_QueryConfigData( )** function to obtain the {set ID:parm ID} pair of the CDP variable using the parameter name obtained from the CDP file.

The **gc_QueryConfigData( )** function is called with the following parameter values:

- **target_type** = GCTGT_PROTOCOL_SYSTEM
- **target_id** = PDK Protocol ID
- **source_datap** = GC_PARM parameter pointer for storing the input CDP variable name
- **query_id** = Query ID, in this case, GCQUERY_PARM_NAME_TO_ID
- **response_datap** = GC_PARM parameter pointer for storing the output {set ID:parm ID} and value type

Note: Obtaining the {set ID:parm ID} pair is a prerequisite to retrieving or changing the value of a CDP variable.

See Section 1.42.1.5, “Sample Code for Getting and Setting CAS Signal Definitions and CDP Variable Values”, on page 120, specifically the **QueryParmID( )** function, for example code that demonstrates how to retrieve the {set ID:parm ID} pair for a CDP variable.

Getting the Current Values of Multiple CDP Variables

The **gc_GetConfigData( )** function can be used to retrieve the value of a CDP variable in the DM3 firmware. Function parameter values to use in this context are:

- **target_type** = GCTGT_PROTOCOL_SYSTEM
- **target_id** = PDK Protocol ID
- **target_datap** = GC_PARM_BLKP parameter pointer, as constructed by the utility function **gc_util_insert_parm_val( )** for CDP integer value and **gc_util_insert_parm_ref( )** for CDP string value.
- **time_out** = time interval (in seconds) during which the parameter value must be retrieved. If the interval is exceeded, the retrieve request is ignored. This parameter is supported in synchronous mode only, and it is ignored when set to 0.
- **request_idp** = pointer to the location for storing the request ID, output from Global Call
- **mode** = EV_ASYNC for asynchronous execution or EV_SYNC for synchronous execution. EV_ASYNC mode is recommended.

See Section 1.42.1.5, “Sample Code for Getting and Setting CAS Signal Definitions and CDP Variable Values”, on page 120, specifically the **GetCDPVarParms( )** function, for example code that demonstrates how to get the current values of multiple CDP variables.

Setting New Values for Multiple CDP Variables

The **gc_SetConfigData( )** function can be used to set new values for multiple CDP variables in the DM3 firmware. Function parameter values to use in this context are:

- **target_type** = GCTGT_PROTOCOL_SYSTEM
- **target_id** = PDK Protocol ID

- **target_datap** = GC_PARM_BLKP parameter pointer, as constructed by the utility function **gc_util_insert_parm_val( )** for the CDP integer value and **gc_util_insert_parm_ref( )** for the CDP string value.
- **time_out** = time interval (in seconds) during which the parameter value must be updated. If the interval is exceeded, the update request is ignored. This parameter is supported in synchronous mode only, and it is ignored when set to 0.
- **update_cond** = ignored for DM3 PDK protocol parameters
- **request_idp** = pointer to the location for storing the request ID, output from Global Call
- **mode** = EV_ASYNC for asynchronous execution or EV_SYNC for synchronous execution. EV_ASYNC mode is recommended.

See Section 1.42.1.5, "Sample Code for Getting and Setting CAS Signal Definitions and CDP Variable Values", on page 120, specifically the **SetCDPVarParms( )** function, for example code that demonstrates how to set new values of multiple CDP variables.

1.42.1.5 Sample Code for Getting and Setting CAS Signal Definitions and CDP Variable Values

```
/* Dialogic Header Files */
#include <gcip.h>
#include <gclib.h>
#include <gcisdn.h>
#include <srllib.h>
#include <dm3cc_parm.h>

int ObtainProtocolIDAndName(LINEDEV a_GCLineDevH, char *a_pProtName, long *a_pProtID)
{
    GC_PARM_BLK * t_pParmBlk = NULL;
    GC_PARM_DATA * t_pParmData = NULL;
    char * t_ProtName[20];
    long t_RequestID = 0;
    int t_result;

    /* Reserve the space for protocol ID */
    *a_pProtID = 0;
    gc_util_insert_parm_val(&t_pParmBlk, GCSET_PROTOCOL, GCPARM_PROTOCOL_ID, sizeof(long),
                            *a_pProtID);
    /* Reserve the space for protocol Name */

    gc_util_insert_parm_ref(&t_pParmBlk, GCSET_PROTOCOL, GCPARM_PROTOCOL_NAME,
                            sizeof(t_ProtName), t_ProtName);
    /* Since the protocol information has already been stored in GC library during gc_OpenEx(),
        it is recommended to call gc_GetConfigData() in SYNC mode */
    t_result = gc_GetConfigData(GCTGT_GCLIB_CHAN, a_GCLineDevH, t_pParmBlk, 0, & t_RequestID,
                                EV_SYNC);
    if (t_result)
    {
        /* Process the error */
        gc_util_delete_parm_blk(t_pParmBlk);
        return t_result;
    }
    /* Obtain the protocol ID */
    t_pParmData = gc_util_find_parm(t_pParmBlk, GCSET_PROTOCOL, GCPARM_PROTOCOL_ID);
    if (NULL != t_pParmData)
    {
        memcpy(a_pProtID, t_pParmData->value_buf, t_pParmData->value_size);
    }
    /* Obtain the protocol Name */
    t_pParmData = gc_util_find_parm(t_pParmBlk, GCSET_PROTOCOL, GCPARM_PROTOCOL_NAME);
    if (NULL != t_pParmData)
    {
        strcpy(a_pProtName, (const char*)t_pParmData->value_buf);
    }
    printf("ObtainProtocolIDAndName(linedev:%d, protocol_id:%d, protocol_name:%s)",
            a_GCLineDevH, *a_pProtID, a_pProtName);
    gc_util_delete_parm_blk(t_pParmBlk);
    return t_result;
}

int QueryParmID(long a_PDKProtocolID, char *a_pParmName, unsigned short * a_pSetID,
                unsigned short * a_pParmID, unsigned char * a_pValType)
{
    GC_PARM t_SourceData;
    GC_PARM t_RespData;
    GC_PARM_ID t_ParmIDBlk;
    int t_result = 0;

    /* Pass the CDP name, which is defined in CDP file, e.g., "CAS_WINKRCV" or "CDP_ANI_ENABLED"
       in pdk_us_mf_io.cdp */
    t_SourceData.paddress = a_pParmName;
    memset(&t_ParmIDBlk, '0', sizeof(GC_PARM_ID));
    t_RespData.pstruct = & t_ParmIDBlk;
    t_result = gc_QueryConfigData(GCTGT_PROTOCOL_SYSTEM, a_PDKProtocolID, &t_SourceData,
                                  GCQUERY_PARM_NAME_TO_ID, &t_RespData);
    if (t_result)
    {
        /* Process the error */
        *a_pSetID = 0;
        *a_pParmID = 0;
        *a_pValType = 0;
        printf("gc_QueryConfigData(parm:%s) failed on protocol:%d", a_pParmName,
                a_PDKProtocolID);
    }
    else
    {
        *a_pSetID = t_ParmIDBlk.set_ID;
        *a_pParmID =  t_ParmIDBlk.parm_ID;
        *a_pValType = t_ParmIDBlk.value_type;
        printf("gc_QueryConfigData(parm:%s) succeed with  {setID:0x%x, parmID:0x%x, valType:%d}
               on protocol:%d",
               a_pParmName, *a_pSetID, *a_pParmID, *a_pValType, a_PDKProtocolID);
    }
    return t_result;
}

int SetCASSignalDef(long a_PDKProtocolID)
{
    GC_PARM_BLK * t_pParmBlk = NULL;
    unsigned short t_SetID;
    unsigned short t_ParmID;
    unsigned char t_ValType;
    long t_RequestID = 0;
    int t_result = 0;
    GC_CASPROT_TRANS t_CasTrans;
    GC_CASPROT_PULSE t_CasPulse = {"00xx", "11xx", 50, 62, 0, 80, 20, 250, 300};
    GC_CASPROT_TRAIN t_CasTrain;
    /* Find the {setID, parmID, DataType} of CAS_WINKRCV for pdk_us_mf_io */
    t_result = QueryParmID(a_PDKProtocolID, "CAS_WINKRCV", &t_SetID, &t_ParmID, &t_ValType);
    if (t_result)
    {
        /* Process the error */
        return t_result;
    }
    /* Insert new definition for CAS signals, dependent on the signal type */
    switch (t_ValType)
    {
        case GC_VALUE_CAS_TRANS:
            gc_util_insert_parm_ref(&t_pParmBlk, t_SetID, t_ParmID, sizeof(GC_CASPROT_TRANS),
                                    &t_CasTrans);
        break;
        case GC_VALUE_CAS_PULSE:
            gc_util_insert_parm_ref(&t_pParmBlk, t_SetID, t_ParmID, sizeof(GC_CASPROT_PULSE),
                                    &t_CasPulse);
        break;
        case GC_VALUE_CAS_TRAIN:
            gc_util_insert_parm_ref(&t_pParmBlk, t_SetID, t_ParmID, sizeof(GC_CASPROT_TRAIN),
                                    &t_CasTrain);
        break;
        default:
        /* Process the error here */
        return -1;
        break;
    }
    /* Set the CAS_WINKRCV with new value */
    t_result = gc_SetConfigData(GCTGT_PROTOCOL_SYSTEM, a_PDKProtocolID, t_pParmBlk, 0,
                            GCUPDATE_IMMEDIATE, &t_RequestID, EV_ASYNC);
    if (t_result)
    {
        /* Process the error */
        gc_util_delete_parm_blk(t_pParmBlk);
        return t_result;
    }
    gc_util_delete_parm_blk(t_pParmBlk);
    return t_result;
}

int GetCASSignalDef(long a_PDKProtocolID)
{
    GC_PARM_BLK * t_pParmBlk = NULL;
    unsigned short t_SetID;
    unsigned short t_ParmID;
    unsigned char t_ValType;
    long t_RequestID = 0;
    int t_result = 0;
    GC_CASPROT_TRANS t_CasTrans;
    GC_CASPROT_PULSE t_CasPulse;
    GC_CASPROT_TRAIN t_CasTrain;
    /* Find the {setID, parmID, dataType} of CAS_WINKRCV for pdk_us_mf_io */
    t_result = QueryParmID(a_PDKProtocolID, "CAS_WINKRCV", &t_SetID, &t_ParmID, &t_ValType);
    if (t_result)
    {
        /* Process the error */
        return t_result;
    }
    /* Insert memory space for storing definition for CAS signals, dependent on the signal type
    */
    switch (t_ValType)
    {
        case GC_VALUE_CAS_TRANS:
            memset( &t_CasPulse, 0, sizeof(GC_CASPROT_TRANS) );
            gc_util_insert_parm_ref(&t_pParmBlk, t_SetID, t_ParmID, sizeof(GC_CASPROT_TRANS),
                                    &t_CasTrans);
        break;
        case GC_VALUE_CAS_PULSE:
            memset( &t_CasPulse, 0, sizeof(GC_CASPROT_PULSE) );
            gc_util_insert_parm_ref(&t_pParmBlk, t_SetID, t_ParmID, sizeof(GC_CASPROT_PULSE),
                                    &t_CasPulse);
        break;
        case GC_VALUE_CAS_TRAIN:
            memset( &t_CasPulse, 0, sizeof(GC_CASPROT_TRAIN) );
            gc_util_insert_parm_ref(&t_pParmBlk, t_SetID, t_ParmID, sizeof(GC_CASPROT_TRAIN),
                                &t_CasTrain);
        break;
        default:
            /* Process the error here */
            return -1;
        break;
    }
    /* Get the CAS_WINKRCV with new value */
    t_result = gc_GetConfigData(GCTGT_PROTOCOL_SYSTEM, a_PDKProtocolID, t_pParmBlk, 0,
                                &t_RequestID, EV_ASYNC);
    if (t_result)
    {
        /* Process the error */
        gc_util_delete_parm_blk(t_pParmBlk);
        return t_result;
    }
    gc_util_delete_parm_blk(t_pParmBlk);
    return t_result;
}



typedef struct {
    char        name[50];
    int         type;
    void *      valuep;
} CDP_PARM;

int GetCDPVarParms(long a_PDKProtocolID, int a_NumParms, CDP_PARM * a_CDPVarParms, long *
a_pRequestID)
{
    GC_PARM_BLK * t_pParmBlk = NULL;
    unsigned short t_SetID;
    unsigned short t_ParmID;
    unsigned char t_ValType = 0;
    int t_result = 0;
    int index1 = 0;

    if (!a_PDKProtocolID)
    {
        /* Process the error */
        return -1;
    }
    if (!a_CDPVarParms)
    {
        /* Process the error */
        return -1;
    }
    /* Support retrieving multiple CDP variables in a single gc_GetConfigData() function call */
    for (index1 = 0; index1 < a_NumParms; index1 ++)
    {
        /* Find the {setID, parmID, valueType} of each CDP variable by its name: e.g.,
           "CDP_ANI_ENABLED" in pdk_ar_r2_io.cdp */
        t_result = QueryParmID(a_PDKProtocolID, a_CDPVarParms[index1].name, &t_SetID, &t_ParmID,
                               &t_ValType);
        if (t_result)
        {
            /* Process the error */
            gc_util_delete_parm_blk(t_pParmBlk);
            return t_result;
        }
        if (t_SetID != PRSET_TSC_VARIABLE)
        {
            /* Not a CDP variable parameter */
            gc_util_delete_parm_blk(t_pParmBlk);
            return -1;
        }
        /* Insert new definition for CDP variable signals, dependent on the value data type */
        switch (t_ValType)
        {
            case GC_VALUE_SHORT:
                gc_util_insert_parm_val(&t_pParmBlk, t_SetID, t_ParmID, sizeof(unsigned short),
                                        0);
            break;
            case GC_VALUE_STRING:
                gc_util_insert_parm_ref(&t_pParmBlk, t_SetID, t_ParmID, 30, "");
            break;
            case GC_VALUE_ULONG:
                gc_util_insert_parm_val(&t_pParmBlk, t_SetID, t_ParmID, sizeof(unsigned long),
                                        0);
            break;
            case GC_VALUE_UCHAR:
                gc_util_insert_parm_val(&t_pParmBlk, t_SetID, t_ParmID, sizeof(unsigned char),
                                        0);
            break;
            default:
                /* Process the error here */
                printf("!!!!Invalid value type for protocolID:%d to CDP variable(name:%s,
                        set_id:0x%x, parm_id:0x%x, valtype:%d)",
                        a_PDKProtocolID, a_CDPVarParms[index1].name, t_SetID, t_ParmID,
                        t_ValType);
                gc_util_delete_parm_blk(t_pParmBlk);
                return -1;
            break;
        }
    }
    /* Get the values of multiple CDP variables */
    *a_pRequestID = 0;
    t_result = gc_GetConfigData(GCTGT_PROTOCOL_SYSTEM, a_PDKProtocolID, t_pParmBlk, 0,
                               a_pRequestID, EV_ASYNC);
    if (t_result)
    {
        /* Process the error */
        printf("gc_GetConfigData(protocol_id:%d) failed on setting CDP parameters()",
                a_PDKProtocolID);
        *a_pRequestID = 0;
    }
    else
    {
        printf("gc_GetConfigData(protocol_id:%d, req_id:0x%x) succeed on setting CDP
               parameters",
               a_PDKProtocolID, *a_pRequestID);
    }
    gc_util_delete_parm_blk(t_pParmBlk);
    return t_result;
}

int SetCDPVarParms(long a_PDKProtocolID, int a_NumParms, CDP_PARM * a_CDPVarParms, long *
a_pRequestID)
{
    GC_PARM_BLK * t_pParmBlk = NULL;
    unsigned short t_SetID;
    unsigned short t_ParmID;
    unsigned char t_ValType = 0;
    int t_result = 0;
    int t_IntVal = 0;
    unsigned long t_ULongVal = 0;
    unsigned char t_UCharVal = 0;
    unsigned char t_StrSize = 0;
    int index1 = 0;

    if (!a_PDKProtocolID)
    {
        /* Process the error */
        return -1;
    }
    if (!a_CDPVarParms)
    {
        /* Process the error */
        return -1;
    }
    /* Support setting multiple CDP variables in a single gc_SetConfigData() function call */
    for (index1 = 0; index1 < a_NumParms; index1 ++)
    {
        /* Find the {setID, parmID, valueType} of each CDP variable by its name: e.g.,
           "CDP_ANI_ENABLED" in pdk_ar_r2_io.cdp */
        t_result = QueryParmID(a_PDKProtocolID, a_CDPVarParms[index1].name, &t_SetID, &t_ParmID,
                               &t_ValType);
        if (t_result)
        {
            /* Process the error */
            gc_util_delete_parm_blk(t_pParmBlk);
            return t_result;
        }
        if (t_SetID != PRSET_TSC_VARIABLE)
        {
            /* Not a CDP variable parameter */
            gc_util_delete_parm_blk(t_pParmBlk);
            return -1;
        }
        /* Insert new definition for CDP variable signals, dependent on the value data type */
        switch (t_ValType)
        {
            case GC_VALUE_INT:
                t_IntVal = *((int*)a_CDPVarParms[index1].valuep);
                gc_util_insert_parm_val(&t_pParmBlk, t_SetID, t_ParmID, sizeof(int), t_IntVal);
                printf("Set Integer Value:%d (0x%x) to parmID:0x%x",
                            t_IntVal, t_IntVal, t_ParmID);
            break;
            case GC_VALUE_STRING:
                t_StrSize = strlen((char *)a_CDPVarParms[index1].valuep) + 1;
                gc_util_insert_parm_ref(&t_pParmBlk, t_SetID, t_ParmID, t_StrSize, (char *)
                                        a_CDPVarParms[index1].valuep);
                printf("Set String Value:%s to parmID:0x%x",
                            (char *) a_CDPVarParms[index1].valuep, t_ParmID);
            break;
            case GC_VALUE_ULONG:
                t_ULongVal = *((unsigned long *)a_CDPVarParms[index1].valuep);
                gc_util_insert_parm_val(&t_pParmBlk, t_SetID, t_ParmID, sizeof(unsigned long),
                                        t_ULongVal);
                printf("Set Long Value:%d (0x%x) to parmID:0x%x",
                            t_ULongVal, t_ULongVal, t_ParmID);
            break;
            case GC_VALUE_UCHAR:
                t_UCharVal = *((unsigned char *)a_CDPVarParms[index1].valuep);
                gc_util_insert_parm_val(&t_pParmBlk, t_SetID, t_ParmID, sizeof(unsigned char),
                                        t_UCharVal);
                printf("Set Char Value:%d(0x%x) to parmID:0x%x",
                            t_UCharVal, t_UCharVal, t_ParmID);
            break;
            default:
                /* Process the error here */
                printf("!!!!Invalid value type for protocolID:%d to CDP variable(name:%s,
                        set_id:0x%x, parm_id:0x%x, valtype:%d)",
                        a_PDKProtocolID, a_CDPVarParms[index1].name, t_SetID, t_ParmID,
                        t_ValType);
                gc_util_delete_parm_blk(t_pParmBlk);
                return -1;
            break;
        }
    }
```

```
    /* Set the CDP parameters with new values */
    *a_pRequestID = 0;
     t_result  = gc_SetConfigData(GCTGT_PROTOCOL_SYSTEM, a_PDKProtocolID, t_pParmBlk, 0,
                              GCUPDATE_IMMEDIATE, a_pRequestID, EV_ASYNC);
    if (t_result)
    {
        /* Process the error */
        printf("gc_SetConfigData(protocol_id:%d) failed on setting CDP parameters()",
                a_PDKProtocolID);
        *a_pRequestID = 0;
    }
    else
    {
        printf("gc_SetConfigData(protocol_id:%d, req_id:0x%x) succeed on setting CDP
                parameters",
            a_PDKProtocolID, *a_pRequestID);
    }
    gc_util_delete_parm_blk(t_pParmBlk);
    return t_result;
}


int ProcessRTCMEvent(unsigned long a_GCEvent, unsigned long a_ReqID, GC_PARM_BLK * a_pParmBlk)
{
    GC_CASPROT_TRANS * t_pCasTrans = NULL;
    GC_CASPROT_PULSE * t_pCasPulse = NULL;
    GC_CASPROT_TRAIN * t_pCasTrain = NULL;
    unsigned char t_UCharVal = 0;
    unsigned short t_UShortVal = 0;
    unsigned long t_ULongVal = 0;
    char * t_StringVal = NULL;
    int t_StrLen = 0;

    /* Obtain the first parameter */
    GC_PARM_DATA * t_pParmData = gc_util_next_parm(a_pParmBlk, NULL);
    while (t_pParmData)
    {
        if (t_pParmData->set_ID == PRSET_CAS_SIGNAL)
        {
            /* This is a CAS signal */
            if (t_pParmData->value_size == sizeof(GC_CASPROT_TRANS) )
            {
                t_pCasTrans = (GC_CASPROT_TRANS *) &t_pParmData->value_buf;
                printf("Obtain CAS Trans signal definition on parmID:0x%x (%s, %s, %d, %d, %d,
                        %d)",
                    t_pParmData->parm_ID, t_pCasTrans->PreTransCode, t_pCasTrans->PostTransCode,
                    t_pCasTrans->PreTransInterval, t_pCasTrans->PostTransInterval,
                    t_pCasTrans->PreTransIntervalNom, t_pCasTrans->PostTransIntervalNom);
            }
            else if (t_pParmData->value_size == sizeof(GC_CASPROT_PULSE) )
            {
                t_pCasPulse = (GC_CASPROT_PULSE *) &t_pParmData->value_buf;
                printf("Obtain CAS Pulse signal definition on parmID:0x%x (%s, %s, %d, %d, %d,
                        %d, %d, %d, %d) ",
                    t_pParmData->parm_ID, t_pCasPulse->OffPulseCode, t_pCasPulse->OnPulseCode,
                    t_pCasPulse->PrePulseInterval, t_pCasPulse->PostPulseInterval,
                    t_pCasPulse->PrePulseIntervalNom, t_pCasPulse->PostPulseIntervalNom,
                    t_pCasPulse->PulseIntervalMin, t_pCasPulse->PulseIntervalNom,
                    t_pCasPulse->PulseIntervalMax);
            }
            else if (t_pParmData->value_size == sizeof(GC_CASPROT_TRAIN) )
            {
                t_pCasTrain = (GC_CASPROT_TRAIN *) &t_pParmData->value_buf;
                printf("Obtain CAS Train signal definition on parmID:0x%x (%s, %s, %d, %d, %d,
                        %d, %d, %d, %d) ",
                    t_pParmData->parm_ID, t_pCasTrain->OffPulseCode, t_pCasTrain->OnPulseCode,
                    t_pCasTrain->PreTrainInterval, t_pCasTrain->PostTrainInterval,
                    t_pCasTrain->PreTrainIntervalNom, t_pCasTrain->PostTrainIntervalNom,
                    t_pCasTrain->PulseIntervalMin, t_pCasTrain->PulseIntervalNom,
                    t_pCasTrain->PulseIntervalMax);
            }
            else
            {
                printf("Error! Incorrect value_size =%d for {setID:0x%x, parmID:0x%x}",
                        t_pParmData->value_size, t_pParmData->set_ID, t_pParmData->parm_ID);
            }
        }
        else if (t_pParmData->set_ID == PRSET_TSC_VARIABLE)
        {
            /* This is a TSC Variable */
            switch (t_pParmData->value_size)
            {
                case 1:
                    /* Unisgned char data */
                    memcpy(&t_UCharVal, &t_pParmData->value_buf,t_pParmData->value_size);
                    printf("Obtain TSC unsigned char value:%d(0x%x) of parmID:0x%x\n",
                                        t_UCharVal, t_UCharVal, t_pParmData->parm_ID);
                    break;
                case 2:
                    /* Unisgned short data */
                    memcpy(&t_UShortVal, &t_pParmData->value_buf,t_pParmData->value_size);
                    printf("Obtain TSC unsigned short value:%d(0x%x) of parmID:0x%x\n",
                        t_UShortVal, t_UShortVal, t_pParmData->parm_ID);
                    break;
                case 4:
                    /* Unisgned long data */
                    memcpy(&t_ULongVal, &t_pParmData->value_buf,t_pParmData->value_size);
                    printf("Obtain TSC integer value:%d(0x%x) of parmID:0x%x",
                            t_ULongVal, t_ULongVal,  t_pParmData->parm_ID);
                    break;
                default:
                {
                    t_StringVal = (char*) t_pParmData->value_buf;
                    t_StrLen = strlen(t_StringVal);
                    if ( t_pParmData->value_size > t_StrLen)
                    {
                        /* String data */
                        printf("Obtain TSC string value:%s(first char: 0x%x) of
                                parmID:0x%x",t_StringVal, t_StringVal[0], t_pParmData->parm_ID);
                    }
                    else
                    {
                        /* Unsupported value size */
                        printf("Unsupported value size:%d for TSC variable parmID:0x%x",
                                t_pParmData->value_size, t_pParmData->parm_ID);
                    }
                }
                break;
            }
        }
        else
        {
            /* Unsupported set ID */
            printf("Unsupported set_id:0x%x with (parmID:0x%x, value_size:%d) ",
                    t_pParmData->set_ID, t_pParmData->parm_ID, t_pParmData->value_size);
        }
        /* Obtain next parameter */
        t_pParmData = gc_util_next_parm(a_pParmBlk, t_pParmData);
    }
    return 0;
}

struct channel
{
   LINEDEV          LineDev;             /* GlobalCall line device handle */
   char             DevName[50];
   long             ProtocolID;
} port[120];

void process_event()
{
    METAEVENT           metaevent;
    int                 evttype;
    GC_RTCM_EVTDATA *   t_pRtcmEvt = NULL;
    int                 t_Result = 0;
    int                 index = 0;
    struct channel      *pline = NULL;
    char t_ProtocolName[30];
    int t_NumParms = 0;
    int t_RequesID = 0;
    CDP_PARM t_CDPVarParms[3] = {
        {"CDP_IN_WinkStart", GC_VALUE_INT, 0},
        {"CDP_OUT_WinkStart", GC_VALUE_INT, 0},
        {"CDP_OUT_Send_Alerting_After_Dialing", GC_VALUE_INT, 0}
    };


    /* Populate the metaEvent structure */
    if(gc_GetMetaEvent(&metaevent) != GC_SUCCESS)
    {
        printf("gc_GetMetaEvent() failed \n");
       /* Process error */
    }
    /* process GlobalCall events */
    if ((metaevent.flags & GCME_GC_EVENT) == 0)
    {
        printf("Received a non-GC Event 0x%lx\n", metaevent.evttype);
        return;
    }
    evttype = metaevent.evttype;
    if (metaevent.usrattr)
    {
        pline = (struct channel *) metaevent.usrattr;
    }
    switch (evttype)
    {
        case GCEV_UNBLOCKED:
        {
            int t_IntVal = 1;
            t_Result = ObtainProtocolIDAndName(pline->LineDev, t_ProtocolName,
                       &pline->ProtocolID);
            if (t_Result)
            {
                /* Error processs */
            }
            t_NumParms = 3;
            t_CDPVarParms[0].valuep = &t_IntVal;
            t_CDPVarParms[1].valuep = &t_IntVal;
            t_CDPVarParms[2].valuep = &t_IntVal;
            /* Setting new values to CDP variables */
            t_Result = SetCDPVarParms(pline->ProtocolID, t_NumParms, t_CDPVarParms,
                                      &t_RequesID);
            if (t_Result)
            {
                /* Processs error */
            }
        }
            break;
        case GCEV_GETCONFIGDATA:
            t_pRtcmEvt = (GC_RTCM_EVTDATA *) metaevent.evtdatap;
            if (! t_pRtcmEvt || !t_pRtcmEvt->retrieved_parmblkp)
            {
                break;
            }
            printf("Received GCEV_GETCONFIGDATA EVENT on target_type=%d, target_id=0x%x,
                    rquest_id=0x%x",
                    t_pRtcmEvt->target_type, t_pRtcmEvt->target_id, t_pRtcmEvt->request_ID);
            ProcessRTCMEvent(evttype, t_pRtcmEvt->request_ID, t_pRtcmEvt->retrieved_parmblkp);
            break;                                  /* RETURN POINT!!!!! */
        break;
        case GCEV_SETCONFIGDATA:
            t_pRtcmEvt = (GC_RTCM_EVTDATA *) metaevent.evtdatap;
            if (! t_pRtcmEvt)
            {
                break;
            }
            printf("Received GCEV_SETCONFIGDATA EVENT on target_type=%d, target_id=0x%x,
                   rquest_id=0x%x",
                   t_pRtcmEvt->target_type, t_pRtcmEvt->target_id, t_pRtcmEvt->request_ID);
            t_NumParms = 3;
            /* Retrieving existing values from CDP variables */
            t_Result = GetCDPVarParms(t_pRtcmEvt->target_id, t_NumParms, t_CDPVarParms,
                                      &t_RequesID);
            if (t_Result)
            {
                /* Processs error */
            }
        break;
        case GCEV_GETCONFIGDATA_FAIL:
            t_pRtcmEvt = (GC_RTCM_EVTDATA *) metaevent.evtdatap;
            if (! t_pRtcmEvt)
            {
                break;
            }
            printf("Received GCEV_GETCONFIGDATA EVENT_FAIL on target_type=%d, target_id=0x%x,
                   rquest_id=0x%x",
                   t_pRtcmEvt->target_type, t_pRtcmEvt->target_id, t_pRtcmEvt->request_ID);
            break;
        case GCEV_SETCONFIGDATA_FAIL:
            t_pRtcmEvt = (GC_RTCM_EVTDATA *) metaevent.evtdatap;
            if (! t_pRtcmEvt)
            {
                break;
            }
            printf("Received GCEV_SETCONFIGDATA_FAIL EVENT on target_type=%d, target_id=0x%x,
                    rquest_id=0x%x",
                    t_pRtcmEvt->target_type, t_pRtcmEvt->target_id, t_pRtcmEvt->request_ID);
            break;
        default:
            break;
    }
}
```

1.42.1.6 Dynamically Configuring a Trunk

This feature enables the user to perform the following dynamic configuration operations at runtime:

- Setting the Line Type and Coding for a Trunk
- Specifying the Protocol for a Trunk

Note: The **gc_SetConfigData( )** function can be used on a board device to perform these operations. However, it is the application's responsibility to handle all active calls on the trunk, and terminate them if necessary. In addition, the **gc_ResetLineDev( )** function may be issued on all channels (time slots) prior to issuing **gc_SetConfigData( )** to prevent incoming calls. If there are any active calls present at the time the **gc_ResetLineDev( )** or **gc_SetConfigData( )** function is issued, they are gracefully terminated internally. The application does not receive GCEV_DISCONNECTED events when calls are terminated in this manner.

Setting the Line Type and Coding for a Trunk

The **gc_SetConfigData( )** function can be used on the board device to reconfigure the line type for the trunk. The **gc_SetConfigData( )** function uses a GC_PARM_BLK structure that contains the configuration information. The GC_PARM_BLK is populated using the **gc_util_insert_parm_val( )** function.

To configure the *line type*, use the **gc_util_insert_parm_val( )** function with the following parameter values:

- **parm_blkpp** = pointer to the address of a valid GC_PARM_BLK structure where the parameter and value are to be inserted
- **setID** = CCSET_LINE_CONFIG
- **parmID** = CCPARM_LINE_TYPE
- **data_size** = sizeof(int)
- **data** = One of the following values:
  - Enum_LineType_dsx1_D4 - D4 framing type, Superframe (SF)
  - Enum_LineType_dsx1_ESF - Extended Superframe (ESF)
  - Enum_LineType_dsx1_E1 - E1 standard framing
  - Enum_LineType_dsx1_E1_CRC - E1 standard framing and CRC-4

To configure the *coding type*, use the **gc_util_insert_parm_val( )** function with the following parameter values:

- **parm_blkpp** = pointer to the address of a valid GC_PARM_BLK structure where the parameter and value are to be inserted
- **setID** = CCSET_LINE_CONFIG
- **parmID** = CCPARM_CODING_TYPE
- **data_size** = sizeof(int)
- **data** = One of the following values:
  - Enum_CodingType_AMI - Alternate Mark Inversion
  - Enum_CodingType_B8ZS - Modified AMI used on T1 lines
  - Enum_CodingType_HDB3 - High Density Bipolar of Order 3 used on E1 lines

Once the GC_PARM_BLK has been populated with the desired values, the **gc_SetConfigData( )** function can be issued to perform the configuration. The parameter values for the **gc_SetConfigData( )** function are as follows:

- **target_type** = GCTGT_CCLIB_NETIF
- **target_id** = the trunk line device handle, as obtained from **gc_OpenEx( )** with a **devicename** string of ":N_dtiBx:P..".

- **target_datap** = GC_PARM_BLKP parameter pointer, as constructed by the utility function **gc_util_insert_parm_val( )**
- **time_out** = time interval (in seconds) during which the target object must be updated with the data. If the interval is exceeded, the update request is ignored. This parameter is supported in synchronous mode only, and it is ignored when set to 0.
- **update_cond** = GCUPDATE_IMMEDIATE
- **request_idp** = pointer to the location for storing the request ID
- **mode** = EV_ASYNC for asynchronous execution or EV_SYNC for synchronous execution

The application receives one of the following events:

- GCEV_SETCONFIGDATA to indicate that the request to dynamically change the line type and/or coding has been successfully initiated.
- GCEV_SETCONFIGDATA_FAIL to indicate that the request to dynamically change the line type and/or coding failed. More information is available from the GC_RTCM_EVTDATA structure associated with the event.

The following code example shows how to dynamically configure a T1 trunk to operate with the Extended Superframe (ESF) line type and the B8ZS coding type.

```
GC_PARM_BLKP ParmBlkp = NULL;
long id;

/* configure Line Type = Extended Superframe for a T1 trunk */
gc_util_insert_parm_val(&ParmBlkp, CCSET_LINE_CONFIG, CCPARM_LINE_TYPE, sizeof(int),
                        Enum_LineType_dsx1_ESF);

/* configure Coding Type = B8ZS for a T1 trunk */
gc_util_insert_parm_val(&ParmBlkp, CCSET_LINE_CONFIG, CCPARM_CODING_TYPE, sizeof(int),
                        Enum_CodingType_B8ZS);

gc_SetConfigData(GCTGT_CCLIB_NETIF, bdev, ParmBlkp, 0, GCUPDATE_IMMEDIATE, &id, EV_ASYNC);
                 gc_util_delete_parm_blk(ParmBlkp);

if (sr_waitevt(-1) >= 0)
{
    METAEVENT meta;
    gc_GetMetaEvent(&meta);
    switch(sr_getevttype())
    {
        case GCEV_SETCONFIGDATA:
            printf("Received event GCEV_SETCONFIGDATA(ReqID=%d) on device %s
                   \n",((GC_RTCM_EVTDATA *)(meta.evtdatap))->request_ID,
                   ATDV_NAMEP(sr_getevtdev()));
            break;
        case GCEV_SETCONFIGDATA_FAIL:
            printf("Received event GCEV_SETCONFIGDATA_FAIL(ReqID=%d) on device
                   %s, Error=%s\n",((GC_RTCM_EVTDATA *)(meta.evtdatap))->request_ID,
                   ATDV_NAMEP(sr_getevtdev()),
            ((GC_RTCM_EVTDATA *)(meta.evtdatap))->additional_msg);
            break;
        default:
            printf("Received event 0x%x on device %s\n", sr_getevttype(),
                   ATDV_NAMEP(sr_getevtdev()));
            break;
    }
}
```

Specifying the Protocol for a Trunk

The protocol used by a trunk can be dynamically configured after devices have been opened using the **gc_SetConfigData( )** function. All channels on the affected trunk inherit the newly selected protocol.

The **gc_SetConfigData( )** function uses a GC_PARM_BLK structure that contains the configuration information. The GC_PARM_BLK is populated using the **gc_util_insert_parm_ref( )** function.

To configure the *protocol*, use the **gc_util_insert_parm_ref( )** function with the following parameter values:

- **parm_blkpp** = pointer to the address of a valid GC_PARM_BLK structure where the parameter and value are to be inserted
- **setID** = GCSET_PROTOCOL
- **parmID** = GCPARM_PROTOCOL_NAME
- **data_size** = strlen("<protocol_name>"), for example, strlen("4ESS")
- **data** = "<protocol_name>", for example, "4ESS" (a null-terminated string). For ISDN protocols, the protocol name must be one of the supported protocols listed in the CONFIG file that corresponds to the PCD/FCD file that is downloaded. Only protocols of the same line type can be selected, that is, if the trunk is of line type E1, then only a protocol variant that is valid for E1 can be selected.

Once the GC_PARM_BLK has been populated with the desired values, the **gc_SetConfigData( )** function can be issued to perform the configuration. The parameter values for the **gc_SetConfigData( )** function are as follows:

- **target_type** = GCTGT_CCLIB_NETIF
- **target_id** = the trunk line device handle, as obtained from **gc_OpenEx( )** with a **devicename** string of ":N_dtiBx:P...".
- **target_datap** = GC_PARM_BLKP parameter pointer, as constructed by the utility function **gc_util_insert_parm_ref( )**
- **time_out** = time interval (in seconds) during which the target object must be updated with the data. If the interval is exceeded, the update request is ignored. This parameter is supported in synchronous mode only, and it is ignored when set to 0.
- **update_cond** = GCUPDATE_IMMEDIATE
- **request_idp** = pointer to the location for storing the request ID
- **mode** = EV_ASYNC for asynchronous execution or EV_SYNC for synchronous execution

The application receives one of the following events:

- GCEV_SETCONFIGDATA to indicate that the request to dynamically change the protocol has been successfully initiated.
- GCEV_SETCONFIGDATA_FAIL to indicate that the request to change the protocol has failed. More information is available from the GC_RTCM_EVTDATA structure associated with the event.

The following code example shows how to dynamically configure a T1 trunk to operate with the 4ESS protocol.

```
static int MAX_PROTOCOL_LEN=20;
GC_PARM_BLKP ParmBlkp = NULL;
long id;
char protocol_name[]="4ESS";

gc_util_insert_parm_ref(&ParmBlkp, GCSET_PROTOCOL, GCPARM_PROTOCOL_NAME,
strlen(protocol_name)+1, protocol_name);

gc_SetConfigData(GCTGT_CCLIB_NETIF, bdev, ParmBlkp, 0, GCUPDATE_IMMEDIATE, &id, EV_ASYNC);
gc_util_delete_parm_blk(ParmBlkp);

if (sr_waitevt(-1) >= 0)
{
    METAEVENT meta;
    gc_GetMetaEvent(&meta);

    switch(sr_getevttype())
    {
        case GCEV_SETCONFIGDATA:
            printf("Received event GCEV_SETCONFIGDATA(ReqID=%d) on device %s
                    \n",((GC_RTCM_EVTDATA *)(meta.evtdatap))->request_ID,
                    ATDV_NAMEP(sr_getevtdev()));
            break;
        case GCEV_SETCONFIGDATA_FAIL:
            printf("Received event GCEV_SETCONFIGDATA_FAIL(ReqID=%d) on device
                    %s, Error=%s\n",((GC_RTCM_EVTDATA *)(meta.evtdatap))->request_ID,
                    ATDV_NAMEP(sr_getevtdev()),
                    ((GC_RTCM_EVTDATA *)(meta.evtdatap))->additional_msg);
            break;
        default:
            printf("Received event 0x%x on device %s\n", sr_getevttype(),
                    ATDV_NAMEP(sr_getevtdev()));
            break;
    }
}
```

1.42.2 Extended and New Data Structures

Extension of GC_RTCM_EVTDATA

Two new data fields (target_type and target_id) are appended to the GC_RTCM_EVTDATA data structure defined in the *gclib.h* file. The GC_RTCM_EVTDATA structure is generally associated with Global Call RTCM events (namely, GCEV_SETCONFIGDATA, GCEV_SETCONFIGDATA_FAIL, GCEV_GETCONFIGDATA, and GCEV_GETCONFIGDATA_FAIL).

The following shows the extended GC_RTCM_EVTDATA data structure with the new fields shown in **bold** text:

```
typedef struct{
    long            request_ID;          /* The RTCM request ID */
    int             gc_result;           /* GC result value for this event */
    int             cclib_result;        /* CCLib result value for this event */
    int             cclib_ID;            /* CCLib ID for the result */
    char *          additional_msg;      /* Additional message for this event */
    GC_PARM_BLKP    retrieved_parmblkp;  /* Retrieved GC_PARM_BLK -- */
```

```
                                        /* used for gc_GetConfigData() in */
                                        /* asynchronous mode */
    int             target_type;        /* Target type */
    long            target_id;          /* Target ID */
} GC_RTCM_EVTDATA, *GC_RTCM_EVTDATAP;
```

The addition of the target_type and target_id fields enables applications to easily identify the DM3 protocol object associated with an event. The change is backward-compatible with usage in current applications. Note that the line_dev and crn accessed by the evtdatap pointer in the METAEVENT structure are zero for DM3 protocol target objects.

New Data Structures for CAS Signals

New data structures are defined in the *gclib.h* file that are used by the **gc_SetConfigData( )** and **gc_GetConfigData( )** functions to retrieve/modify the CAS signal definitions associated with a PDK protocol.

As a convenience that enables the user to enter a new CAS signal definition and retrieve the current CAS signal definition, the fields in these data structures strictly follow the same sequence as the CAS signal definitions in the PDK CDP file. Since CAS signal defines in the CDP file apply to both DM3 and Springware Boards, some time parameters may not be supported on DM3 Boards. Also, ASCII characters are used to represent signal bit codes in the data structures. For example, “11xx” represents signal bits 11xx (where x represents “don't care”). All time parameters have units in milliseconds with a resolution of 4 milliseconds.

The following define for the size of the CAS signal bits string is common to all three structures following:

```
#define     GCVAL_CAS_CODE_SIZE          0x5     /* The size of CAS Signal code in string */
```

CAS Transition Signal

```
/* Data structure for CAS Transition signal */
typedef struct {
    char            PreTransCode[GCVAL_CAS_CODE_SIZE];   /* ABCD pre-transition code */
    char            PostTransCode[GCVAL_CAS_CODE_SIZE];  /* ABCD post-transition code */
    unsigned short  PreTransInterval;                    /* The minimum time for the duration
                                                            of the pre-transition (in msec)*/
    unsigned short  PostTransInterval;                   /* The minimum time for the duration
                                                            of the post-transition (in msec)*/
    unsigned short  PreTransIntervalNom;                 /* The nominal time for the duration
                                                            of the pre-transition (in msec).
                                                            Ignored in DM3: always 0 */
    unsigned short  PostTransIntervalNom;                /* The nominal time for the duration
                                                            of the post-transition (in msec).
                                                            Ignored in DM3: always 0 */
} GC_CASPROT_TRANS;
```

CAS Pulse Signal

```
/* Data structure of CAS Pulse signal */
typedef struct {
    char            OffPulseCode[GCVAL_CAS_CODE_SIZE];   /* ABCD pulse off code */
    char            OnPulseCode[GCVAL_CAS_CODE_SIZE];    /* ABCD pulse on code */
    unsigned short  PrePulseInterval;                    /* The minimum time for the duration
```

```
                                                            of the pre-pulse (in msec) */
    unsigned short  PostPulseInterval;                   /* The minimum time for the duration
                                                            of the post-pulse (in msec) */
    unsigned short  PrePulseIntervalNom;                 /* The nominal time for the duration
                                                    of the pre-pulse. Ignored in DM3: always 0 */
    unsigned short  PostPulseIntervalNom;                /* The nominal time for the duration
                                          of the post-pulse (in msec). Ignored in DM3: always 0 */
    unsigned short  PulseIntervalMin;                    /* The minimum time for the duration
                                                            of the pulse interval (in msec) */
    unsigned short  PulseIntervalNom;                    /* The nominal time for the duration
                                                            of the pulse interval (in msec) */
    unsigned short  PulseIntervalMax;                    /* The maximum time for the duration
                                                            of the pulse interval (in msec) */
} GC_CASPROT_PULSE;
```

CAS Train Signal

```
/* Data structure of CAS Train signal */
typedef struct {
    char            OffPulseCode[GCVAL_CAS_CODE_SIZE];    /* ABCD pulse off code */
    char            OnPulseCode[GCVAL_CAS_CODE_SIZE];     /* ABCD pulse on code */
    unsigned short  PreTrainInterval;                     /* The minimum time for the duration
                                                             of the pre-train (in msec) */
    unsigned short  PostTrainInterval;                    /* The minimum time for the duration
                                                             of the post-train (in msec) */
    unsigned short  PreTrainIntervalNom;                  /* The nominal time for the duration
                                                     of the pre-train. Ignored in DM3: always 0 */
    unsigned short  PostTrainIntervalNom;                 /* The nominal time for the duration
                                          of the post-train (in msec). Ignored in DM3: always 0 */
    unsigned short  PulseIntervalMin;                     /* The minimum time for the duration
                                                             of the pulse interval (in msec)*/
    unsigned short  PulseIntervalNom;                     /* The nominal time for the duration
                                                             of the pulse interval (in msec)*/
    unsigned short  PulseIntervalMax;                     /* The maximum time for the duration
                                                             of the pulse interval (in msec)*/
    unsigned short  InterPulseIntervalMin;                /* The minimum time for the duration
                                                             of inter-pulse interval (in msec)*/
    unsigned short  InterPulseIntervalNom;                /* The nominal time for the duration
                                                             of inter-pulse interval (in msec)*/
    unsigned short  InterPulseIntervalMax;                /* The maximum time for the duration
                                                            of inter-pulse interval (in msec) */
} GC_CASPROT_TRAIN;
```

CAS Signal Type Defines

The following new value types for CAS signal parameter are defined in the *gccfgparm.h* file to represent the CAS Transition, CAS Pulse, and CAS Train types, respectively. These defines are used by the **gc_QueryConfigData( )** for the value type of CAS signal.

```
    GC_VALUE_CAS_TRANS    =    0x10,    /* CAS Transition data struture ==> GC_CASPROT_TRANS */
    GC_VALUE_CAS_PULSE    =    0x11,    /* CAS Pulse data struture ==> GC_CASPROT_PULSE */
    GC_VALUE_CAS_TRAIN    =    0x12,    /* CAS Train data struture ==> GC_CASPROT_TRAIN */
```

Other value types (for example, integer, string, long, etc.) have already been defined in the *gccfgparm.h* file.

New Set IDs and Parm IDs

This feature uses the following new Set IDs and Parm IDs:

Set ID	Parm IDs
CCSET_LINE_CONFIG	CCPARM_LINE_TYPE CCPARM_CODING_TYPE
GCSET_PROTOCOL	GCPARM_PROTOCOL_ID GCPARM_PROTOCOL_NAME
PRSET_CAS_SIGNAL (defined in *dm3cc_parm.h*)	The parm ID is dynamically generated.
PRSET_TSC_VARIABLE (defined in *dm3cc_parm.h*)	The parm ID is dynamically generated.

1.42.3 Restrictions and Limitations

The following restrictions and limitations apply:

- This feature supports the redefinition of CAS signals and the setting of CDP variable values for a specific protocol variant, which will affect all channels running that protocol variant. It does not, however, support the getting or setting of protocol parameters on an individual channel basis. Getting and setting CAS signal definitions or CDP variable values is only supported for PDK protocols.
- Prior to changing parameters of a protocol, all channels running the protocol should be in the Idle state (that is, there should be no call activity on the channels). Once the parameter value change is complete, it is recommended to reset the channels running the affected protocol.
- At lease one time slot has to remain open while setting or retrieving CAS signal definitions or CDP variable values.
- Using this feature to set/get multiple CAS signal definitions in a single GC_PARM_BLK via the **gc_SetConfigData( )** and **gc_GetConfigData( )** functions or mixing CAS signal definitions with other parameters is not supported. Only one CAS signal definition (and no other parameters) can be included in any one function call.
- The API for this feature can be used only after the board firmware has been downloaded.
- Configuration files are not updated with changes made using this API for this feature. The API does not save or store the changes made and if the firmware is re-downloaded, all information configured using this API will be lost. It is the API user's responsibility to save or store the changed configuration information and reset via the API in the event of a re-download.
- This feature supports the redefinition of CAS Transition, CAS Pulse and CAS Train signals only. In addition, this feature does not support the changing of the CAS signal type during redefinition. For example, the CAS_WINKRCV signal type cannot be changed from a CAS Pulse to a CAS Transition.
- Error checking and ensuring the validity of parameters passed through this API are the responsibilities of the API user.
- The list of parameters that need to be modified must be managed at run time. Parameter updates are sent to the firmware one at a time as opposed to the parallel procedures used to set parameters at firmware download time. It is recommended to keep the list of parameters that need modification to a minimum.

- This feature supports the setting and retrieval of multiple CDP variable values in a single API call, but it does not support the mixing of CDP variables with other parameters when setting or retrieving values.
- To set the values of the CDP_IN_ANI_Enabled and CDP_OUT_ANI_Enabled parameters in the *pdk_us_mf_io.cdp* file, the user is required to remove feature_ANI from the SYS_FEATURES section of the CDP file. Similarly, to set the values of the CDP_IN_DNIS_Enabled and CDP_OUT_DNIS_Enabled parameters, the user is required to remove feature_DNIS from the SYS_FEATURES section.

1.42.4 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the Dialogic® Global Call API, see the following documents:

- *Dialogic® Global Call API Programming Guide*
- *Dialogic® Global Call API Library Reference*

For features specific to E1 and T1 technology, see:

- *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*

For features specific to ISDN technology, see:

- *Dialogic® Global Call ISDN Technology Guide*

Note: The online bookshelf has not been updated for this feature, so the manuals above do not contain information relating to this feature.

1.43 Analog Line Adaptation Utility (LineAdapt)

This section provides detailed information on how to use the line adaptation utility. The following topics are covered:

- Supported Products
- Feature Description
- Line Adaptation Utility Overview
- Line Adaptation Procedures
- LineAdapt Utility Command Line Interface
- Documentation

1.43.1 Supported Products

The line adaptation utility (LineAdapt) is supported on the following analog front-end boards.

Springware Boards:

- Dialogic® D/41JCT-LS Media Boards (North American version)
- Dialogic® D/120JCT-LS Rev. 2 Media Boards (North American version)

DM3 Boards:

- Dialogic® DMV160LP Media Boards (but not the Dialogic® DMV160LPHIZ high impedance, or HiZ version)
- Dialogic® DI0408LSAR2 Switching Boards (this is the Rev. 2 version)

1.43.2 Feature Description

The line adaptation utility is a configuration utility 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. (Some impedance deviation is present in all analog telephone lines.) The utility helps to correct trunk environments where the analog telephone line and the analog board front end impedance do not conform. In some extreme cases, this situation can cause a transmitter side line echo that disrupts perceived voice quality and triggers DTMF termination events. The utility normalizes the impedance mismatch by discovering the optimum settings for individual channels or ports on the analog board and initializing the board to use the optimum settings.

1.43.3 Line Adaptation Utility Overview

Line adaptation consists of running the host-based LineAdapt utility to discover and store the optimum impedance settings for individual channels on analog boards that are connected to analog telephone lines. The stored settings are then used whenever the boards are downloaded, such as at system startup when the Dialogic® System Services are started on the boards.

Therefore, the line adaptation process consists of the following parts:

- Configuration: Discovery and storage of optimum settings (typically performed once)
- Initialization: Using the stored optimum settings at startup (performed whenever the boards are downloaded or initialized)

Line adaptation procedures and related information are described in Section 1.43.4, “Line Adaptation Procedures”, on page 139.

The command line interface for the utility is documented in Section 1.43.5, “LineAdapt Utility Command Line Interface”, on page 145.

1.43.4 Line Adaptation Procedures

This section contains the following topics:

- Location and Description of the Utility and Component Files
- Line Adaptation Procedure
- Configuration File Tasks

1.43.4.1 Location and Description of the Utility and Component Files

The following list provides a brief description and the location of the line adaptation utility along with its required components and associated files. All locations are relative to the Dialogic® Software home directory as specified by the %INTEL_DIALOGIC_DIR% environment variable.

LineAdapt.exe
: The line adaptation utility executable, which provides a command line interface. Location: **bin** directory.

LineAdapt.log
: The utility log file. Contains information output during execution of the LineAdapt utility, including results, errors, and warnings. Location: **log** directory.

LineAdaptDefCoefficients.xml
: The data file used by the line adaptation utility during the optimum coefficient discovery process. It contains the list of boards supported and the default coefficient sets used for discovery. Location: **cfg** directory.

Note: Do not edit or modify the *LineAdaptDefCoefficients.xml* file. The list of boards and the default data sets are preset and must not be changed.

LineAdaptOptCoef.xml
: The configuration file used by the line adaptation utility. As a result of running the LineAdapt utility in discovery mode, the utility stores in this file the optimum Quad Subscriber Line Audio-processing Circuit (QSLAC) filter coefficients for specific boards and lines. These optimum coefficients are then used to initialize the board at startup. Location: **cfg** directory. For related information on the file, see Section 1.43.4.3, "Configuration File Tasks", on page 143.

LineAdapt Tone Files
: The tone files (e.g., LineAdapt510.WAV, LineAdapt1020.WAV, and LineAdapt2020.WAV) used by the utility during optimum coefficient discovery. Location: **cfg** directory.

Note: Do not edit or modify the tone files.

1.43.4.2 Line Adaptation Procedure

This section provides information on the following topics:

- Requirements for Line Adaptation
- Set Up the System and Configure Target Boards for Discovery
- Execute the LineAdapt Utility to Discover Optimum Settings

- Post-Discovery: Reconfigure and Initialize the Boards, and Test the System for Normal Use

Requirements for Line Adaptation

You must meet the following requirements for the line adaptation discovery phase.

- The analog telephone lines and boards that require adaptation have been identified (referred to as the **target** boards/lines).
- The target boards are configured and able to function.
- The telephone lines are attached to the target boards/channels.
- There must be no inbound telephone calls during execution of the LineAdapt utility (the incoming calls should be deactivated at the central office prior to running the utility).
- Application and other programs must not access the target boards and telephone lines during execution of the utility.
- Prior to line adaptation, you must perform the procedures in Set Up the System and Configure Target Boards for Discovery. Afterwards, you must perform the procedures in Post-Discovery: Reconfigure and Initialize the Boards, and Test the System for Normal Use.
- The instructions in this section assume that you are familiar with the Dialogic® Configuration Manager (DCM). For information on using DCM, you can use the DCM online help and you can also refer to the documentation in the Configuration Guide for your board (separate configuration guides are provided for Dialogic® Springware Boards and for Dialogic® DM3 Boards). The *Dialogic® System Software for PCI Products on Windows® Administration Guide* may also be of use.

Set Up the System and Configure Target Boards for Discovery

Use the following procedure to configure the target boards with Continuous Speech Processing (CSP) firmware for performing adaptation discovery:

1. If the target boards are running, use DCM to stop them.
2. Select the target board in DCM, open the "Misc" property sheet, and locate the parameter that specifies the firmware file for the board. For **DM3** Boards, the firmware file is identified by the **PCDFileName** parameter, and for **Springware** Boards, it is identified by the **FirmwareFileName** parameter.
3. Before making any changes, *make note of your existing firmware file name*. Then select one of the following firmware file names that is applicable to your board (if different from the existing file name).

   Note: Proper optimization requires that one of the following firmware files, which support CSP, be downloaded to the board for the discovery procedure.

Board	PCDFileName (*DM3* Boards)	FirmwareFileName (*Springware* Boards)
D/41JCT-LS		D41JCSP.FWL
D/120JCT-LS		D120CSP.FWL

Board	PCDFileName (*DM3* Boards)	FirmwareFileName (*Springware* Boards)
DMV160LP	DMV160LP.PCD	
DI0408LSAR2	DI0408LSA_REV2_ML2.PCD DI0408LSA_REV2_ML3.PCD DI0408LSA_REV2_ML4.PCD	

4. Save the configuration and repeat the configuration procedure for all target boards.
5. Start the target boards using DCM.

Execute the LineAdapt Utility to Discover Optimum Settings

Use the following procedure to execute the LineAdapt utility and discover the optimum settings for the target boards:

1. Open a command window.
2. Execute the LineAdapt utility (located in the **bin** directory) in **Prompting Mode** to select and adapt the target boards and channels as follows:

   LineAdapt

   Note: For running the utility in **Command Mode** or to use command line parameters, see Section 1.43.5, "LineAdapt Utility Command Line Interface", on page 145.
3. The utility displays a list of all supported boards that are recognized (started) and shows the logical ID (for DM3 Boards) or board ID (for Springware Boards), location (bus/slot), and the number of channels for each board, along with a prompt for selecting the target boards, similar to the following:

```
Please Enter the Boards you would like to Adapt. 'A' for All or 'Q' to
Quit
BoardNumber  BUS  SLOT  BoardName
01           00   10     D/41JCT-LS #0 in Slot 0/10
                         Number Of Channels = 4
02           02   04     D/120JCT-LS-Rev2 #1 in slot 2/4
                         Number Of Channels = 12
03           01   03     DI0408-LS-A-R2 #2 in slot 1/3
                         Number Of Channels = 4
Selected Board Options >
```

   Note: To quit the utility at any time, type **Q** or **q** and press the Enter key. All keyboard entries are case-insensitive.
4. To select **all boards**, type **A** or **a** and press the Enter key.

   To select **one or more individual boards**, type the board numbers (listed in the first column) separated by a space (e.g., to select the D/41JCT-LS and the DI0408LSAR2 Boards, type **1 3** and press the Enter key).

   To select a **range of boards**, type the board numbers (listed in the first column) separated by a dash or hyphen (e.g., to select the D/41JCT-LS and the D/120JCT-LS Boards, type **1-2** and press the Enter key).

Note: You can run the utility again later and select individual boards to add to the configuration.

For each board selected, the utility displays a prompt for selecting the target channels, similar to the following:

```
Enter the Channels you would like to adapt on <boardname> #<id> in slot
<bus>/<slot>. 'A' for All or 'Q' to Quit >
```

5. To select **all channels** on the specified board, type **A** or **a** and press the Enter key.

   To select **one or more individual channels**, type the channel numbers separated by a space (e.g., to select the last channels on the D/41JCT-LS Board, type **3 4** and press the Enter key).

   To select a **range of channels**, type the channel numbers separated by a dash or hyphen (e.g., to select the first three channels on the D/41JCT-LS Board, type **1-3** and press the Enter key).
6. After you select the channels for each board, the utility displays progress messages as it performs tests on the selected targets. It takes the target channels off-hook, detects dial tone, dials a digit to obtain silence, and then performs tests to measure and calculate the best settings. It selects the optimum QSLAC filter coefficients that will adapt the channels to their particular analog telephone lines and stores these settings in the *LineAdaptOptCoef.xml* file in the *cfg* directory. The utility records the results and any warnings or errors in the *LineAdapt.log* file in the *log* directory, and it initializes (downloads) the boards with the optimum settings.
7. After the discovery process is complete, check the *LineAdapt.log* in the *log* directory for errors or warnings to ensure that the adaptation was successful.

Post-Discovery: Reconfigure and Initialize the Boards, and Test the System for Normal Use

After discovery has been successfully completed, perform the following steps to reconfigure, initialize, and test the system:

1. Use DCM to stop the target boards.
2. Restore the firmware file name back to the original file name that was used before you changed it according to the instructions in the section on Set Up the System and Configure Target Boards for Discovery.
3. Start the target boards and then test the system to confirm that the adaptation was successful. Verification testing should include running a user application program to ensure that line echo performance is acceptable.

   Note: The boards will be initialized with the optimum line impedance configuration upon startup as long as the *LineAdaptOptCoef.xml* file is present in the *cfg* directory, and it contains the optimum coefficients discovered from running the utility.

1.43.4.3 Configuration File Tasks

The following list describes some important tasks for using the LineAdapt utility and its configuration file. These tasks relate to certain actions that affect the contents of the configuration file. For more information on these actions and their effects, see Discovery, Initialization, and Storage of Optimum Settings in Section 1.43.5, "LineAdapt Utility Command Line Interface", on page 145.

Back up the configuration file

It is good practice to back up the *LineAdaptOptCoef.xml* when adaptation is complete. Having a backup copy of the file is a good idea especially if you remove a board and replace it later, or if you move a board to a different slot.

Add to the configuration

You can add optimum coefficients to the configuration at a later time. You can run the utility multiple times and select individual boards to add to the configuration without changing or replacing the configuration of targets that have already been optimized. When you run the utility, simply select the new targets to adapt (boards/channels) and these will be added to the configuration.

Remove a board from the configuration **or**
Remove selected channels from the configuration

You can remove from the configuration file the line adaptation settings for all channels on a target board and return that board to its default configuration. You do this by physically removing the board from the chassis and then running the utility to adapt any channel on an existing board. You can also remove a board or selected channels from the configuration by editing the *LineAdaptOptCoef.xml* configuration file (see the following topic on Board and Channel Identification in the LineAdaptOptCoef.xml Configuration File).

Note: To remove the optimum settings for all boards, you can delete the *LineAdaptOptCoef.xml* file or execute LineAdapt -r.

Replace a board in the chassis (in case of board failure)

If you replace a board with the same type of board in the same slot and attach the trunk cables to the same port locations on the board, you do not need to make any changes to the configuration file or re-adapt for the new board. The same optimum settings apply to the new board, because optimum settings apply to the board type in a given slot and to the line conditions on the trunk.

Move a board to another slot (in case of a slot failure) **or**
Move analog trunks/lines to another board of same type

If you want to move a board that has optimum settings existing in the *LineAdaptOptCoef.xml* configuration file to another slot, as when a slot failure occurs, you can edit the *LineAdaptOptCoef.xml* configuration file to change its location. This allows you to reconfigure the board for the new location without the necessity of running the LineAdapt utility to rediscover its optimum settings. You can change the slot number only if the same telephone lines remain attached or are re-attached to the same ports on the board.

Similarly, if you want to move the analog trunks/lines from one board that has optimum settings existing in the *LineAdaptOptCoef.xml* configuration file to another board of the same type but in different slot, you can edit the *LineAdaptOptCoef.xml* configuration file to specify the slot location of the board. You can change the slot number only if the same telephone lines are attached to the same port locations on the same type of board.

See the following topic on Board and Channel Identification in the LineAdaptOptCoef.xml Configuration File.

Note: You must also change other configuration files in which the board is identified. For information on making other configuration changes, see the *Dialogic® System Software for PCI Products on Windows® Administration Guide* and also the Configuration Guide for your board (separate configuration guides are provided for Dialogic® Springware Boards and for Dialogic® DM3 Boards). If a board is in a slot that becomes defective or fails, moving the board to another slot is treated the same as removing the board and then adding it to the system.

Board and Channel Identification in the LineAdaptOptCoef.xml Configuration File

The following information describes the valid format for identifying boards and channels in the *LineAdaptOptCoef.xml* configuration file. This information is provided in case you need to edit an existing board identification line to change its logical ID (for DM3 Boards) or board ID (for Springware Boards), and bus/slot location, or in case you wish to delete existing board or channel settings from the file. The configuration file is located in the *cfg* directory.

Warning: The *LineAdaptOptCoef.xml* configuration file is generated by the LineAdapt utility, and proper board functioning depends upon its integrity. If you change the file, you run the risk of introducing an error into the file and you do so at your own risk.

Notes:
1. You must use an XML editor to edit the *LineAdaptOptCoef.xml* configuration file properly. You must not edit or modify any data other than the data specified below. Do not attempt to edit the optimum coefficient values.
2. Make sure to create a backup copy of the configuration file before attempting any changes.
3. Make sure to keep a record of your changes.
4. If you introduce an error into the file, try reverting to your backup copy of the file or run the discovery procedure according to the instructions in Section 1.43.4.2, “Line Adaptation Procedure”, on page 139.

Board Identification

The *BOARD* keyword and its parameters identify the board name, logical ID (for DM3 Boards) or board ID (for Springware Boards), bus/slot location, and architecture type. The BOARD identification line is followed by a channel identification line and the optimum coefficient settings that apply to the channel. The format of the BOARD identification line is shown through the following examples:

```
<<BOARD Name="DI/0408-LS-A-R2 #0 in slot 1/11" Type="DM3">
```

```
<<BOARD Name="D/41JCTls #1 in slot 2/3" Type="Springware">
```

The #0 and #1 represent the logical ID or board ID. The slot 1/11 and slot 2/3 represent the bus and slot numbers, which identify the unique location of the boards. See the following valid values for the *Name* and *Type* parameters.

Warning: Do not change the board *Name* or *Type*. The optimum settings are specific to the category of board (board name and architecture type). The values for these parameters are only shown to help you identify specific boards.

Valid **DM3** Board names for **Name** parameter:

- DI/0408-LS-A-R2
- DMV160LP

Valid **Springware** Board names for **Name** parameter:

- D/41JCTls
- D/120JCT-LS-Rev 2

Valid values for **Type** parameter:

- DM3
- Springware

Channel Identification

The *Channel number* keyword identifies the channel on the board. The channel identification line is followed by the optimum coefficient settings that apply to the channel. The format of the channel identification line is shown through the following example:

```
<Channel number="1">
```

This specifies channel number 1.

Note: Do not attempt to edit the optimum coefficient values.

Terminator for End of Board Data Section

Each board section is terminated by an end-board line as follows:

```
</BOARD>
```

1.43.5 LineAdapt Utility Command Line Interface

For the location of the utility program executable and its files, see Section 1.43.4.1, “Location and Description of the Utility and Component Files”, on page 139.

This section provides information on the following topics:

- Prompting Mode and Command Mode
- Discovery, Initialization, and Storage of Optimum Settings

- LineAdapt Command Line Parameters

Prompting Mode and Command Mode

The LineAdapt utility can run in either Prompting Mode or Command Mode:

Prompting Mode

Displays a list of boards and channels and prompts the user to select the adaptation targets. You can invoke the utility in Prompting Mode by running it without any command line parameters. You can also use the -d, -t, and -v parameters in Prompting Mode (see Table 1). After you select adaptation targets in Prompting Mode, the utility performs discovery, stores the optimum settings in the *LineAdaptOptCoef.xml* file for future initializations, and initializes the boards with the settings. For details on Prompting Mode operation, see Execute the LineAdapt Utility to Discover Optimum Settings in Section 1.43.4.2, "Line Adaptation Procedure", on page 139.

Command Mode

Executes the command line and does not prompt for user input. Command Mode can be used for batch files or scripts, or for direct interaction with a user. You can invoke the utility in Command Mode by specifying the -a parameter to perform discovery on all supported and recognized boards.

Note: All parameters can be used in Command Mode (see Table 1).

Table 1. LineAdapt Utility Parameters Applicable to Prompting and Command Modes

Option	Command Mode	Prompting Mode
<none>	no	yes
-a	yes	no
-b	yes	no
-c	yes	no
-d	yes	yes
-h	yes	no
-l	yes	no
-n	yes	no
-r	yes	no
-s	yes	no
-t	yes	yes
-v	yes	yes

Discovery, Initialization, and Storage of Optimum Settings

The following describes adaptation information related to discovery, initialization, and storage of optimum settings:

- The utility adapts a board to the conditions on the trunk or telephone lines. The optimum settings apply to the line conditions on the trunk for the given board type in a specific slot. If you replace a board with the same type of board in the same slot and attach the trunk cables to the same port locations on the board, you do not need to make any changes to the configuration file or re-adapt for the new board. The same optimum settings apply to the new board.

- If you perform discovery with the utility, it always results in board initialization with the optimum settings discovered.
- Storage of optimum settings depends upon the command line parameter used. If you execute the utility in Command Mode with the -t parameter for temporary discovery, the optimum settings discovered **are not stored** in the *LineAdaptOptCoef.xml* configuration file for future initializations. If you execute the utility in Prompting Mode, or in Command Mode with the -a parameter or the -b -s parameters, the optimum settings discovered **are stored** in the *LineAdaptOptCoef.xml* configuration file for future initializations.
- You can execute the utility more than once to add to or change the settings stored in the *LineAdaptOptCoef.xml* configuration file. If you perform adaptation on a new target, the settings will be added to the configuration file. If you perform adaptation on any target that already exists in the configuration file, those settings will replace the ones in the configuration file.
- If you physically remove from its slot a board that has optimum settings existing in the *LineAdaptOptCoef.xml* configuration file and then perform **line adaptation** with stored results for any target, the settings for the “missing” board will be **deleted from the file**. This is true whether you remove the board from the chassis or move it to another slot. However, if you do not perform line adaptation with storage and only perform a system startup or initialize the boards with or without the utility, an error message is recorded in the *LineAdapt.log* file, but it will not delete the settings for the missing board from the *LineAdaptOptCoef.xml* configuration file.

  Note: To remove the optimum settings for all boards, you can delete the *LineAdaptOptCoef.xml* file or execute LineAdapt -r.
- If you disable, stop, or do not start a board that has optimum settings existing in the *LineAdaptOptCoef.xml* configuration file, and then you perform line adaptation for any target, an error message is recorded in the *LineAdapt.log* file, but it will not delete the settings for the disabled board from the *LineAdaptOptCoef.xml* configuration file.

LineAdapt Command Line Parameters

Command Line: `LineAdapt [ -parameter [value] -parameter [value] ... ]`

Square brackets indicate optional items. An ellipsis (...) indicates that the preceding items can be repeated. A vertical bar or pipe symbol (|) indicates that the items on either side of the bar are mutually exclusive.

This utility supports the following command line parameters. Most of the parameters are flags. If more than one parameter is used, they must be separated by a space. For parameters that specify values, a space between the parameter and its value is optional; however, for readability they are shown without a space in examples (so as to distinguish the parameter/value pairs from one another more easily).

Note: All parameters can be used in Command Mode.

<no parameters>

When the command line does not specify any parameters, it executes the utility in **Prompting Mode**, which displays a list of boards and channels and prompts the user to select the adaptation targets. You can also use the -d, -t, and -v parameters in Prompting Mode (see Table 1). For operation details, see Execute the LineAdapt Utility to Discover Optimum Settings in Section 1.43.4.2, “Line Adaptation Procedure”, on page 139.

-a

Adapts *all* channels on *all* supported and recognized boards. Performs discovery, stores the optimum settings in the *LineAdaptOptCoef.xml* file for future initializations, and initializes the boards with the settings. The -a parameter is mutually exclusive with the -b, -s, and -c parameters, which specify target boards and channels, and with the -n parameter, which does not perform discovery.

-b <bus number> -s <slot number> [-c <channel target>] ...

The -b parameter specifies the *bus* number in the chassis where the target board resides. This parameter must be combined with the -s parameter in the order shown; together they identify a specific board as the target for adaptation. When you specify target boards, the utility performs discovery, stores the optimum settings in the *LineAdaptOptCoef.xml* file for future initializations, and initializes the boards with the settings. If desired, you can add the -c parameter following the -b -s parameters to specify target channels on the board (see the -c parameter for details on how to specify more than one channel). You can include more than one target board on the command line by specifying more than one set of -b -s parameters. The -b, -s, and -c parameters specify target boards and channels and are mutually exclusive with the -a parameter, which specifies all applicable boards, and they are mutually exclusive with the -n parameter, which does not perform discovery. Syntax example:

```
LineAdapt -b0 -s1 -b0 -s2 -c1 -b1 -s1 -c1 -c2
```

This command adapts the following targets:

- All channels on the target board in bus 0, slot 1.
- Channel 1 on the target board in bus 0, slot 2.
- Channels 1 and 2 on the target board in bus 1, slot 1.

The command performs discovery on the specified targets, stores the optimum coefficients in the *LineAdaptOptCoef.xml* file for initialization with future downloads, and initializes the boards with the settings.

-c <channel target> ...

Specifies a *channel* target on the board identified by the -b -s parameters. If the -c parameter is used, it must follow the slot number (see the -b parameter for details). The channel target can be a single channel number or a range of channel numbers. In either case, you can include more than one channel target on the command line by specifying more than one -c parameter (i.e., the -c parameter can be repeated). The format to specify a range of channel numbers is to specify a starting channel number and ending channel number separated by a dash or hyphen. Channel numbers must be within the range of channels on the board (use the -l parameter to display the channel numbers on the board). If channel targets are not specified (i.e., if the -c parameter is not used), the default is all channels on the board. Syntax example:

```
    LineAdapt -b0 -s1 -c1 -c2 -c8-12
```

This command adapts channels 1 and 2 and channels 8 through 12 on the target board in bus 0, slot 1. The command performs discovery on the specified targets, stores the optimum coefficients in the *LineAdaptOptCoef.xml* file for initialization with future downloads, and initializes the boards with the settings.

-d <dial string>

Specifies a valid *dial string* to connect to a known silent termination when the utility takes the channel off-hook to perform testing. If not specified, the default is DTMF digit 3, which is used to silence the dial tone. Alternatively, another DTMF digit can be specified in the dial string to silence the dial tone (e.g., LineAdapt -d5). The -d parameter can also be used in Prompting Mode.

-h

Displays online *help* showing all the possible command line arguments. This parameter is not used with any other parameters (it is a stand-alone parameter), but if any other parameters are specified, it takes precedence over them.

-l

Lists board information for supported and recognized (started) boards, including the bus number, slot number, and channels. This parameter is not used with any other parameters (it is a stand-alone parameter).

-n [-DM3 | -Springware]

Specifies *no discovery* (and thus also no storage). This parameter uses the current configuration in the *LineAdaptOptCoef.xml* file to initialize the boards with their optimum coefficients. Since this parameter requires optimum settings in the *LineAdaptOptCoef.xml* file, the utility must have been executed previously for discovery and storage of optimum settings. This parameter is used internally by DCM in the download or system startup sequence, although it can be used independently of it as well. To initialize only DM3 Boards, specify LineAdapt -n -DM3; or to initialize only Springware Boards, specify LineAdapt -n -Springware; otherwise the -n parameter applies to all board types. The -n parameter is mutually exclusive with the -b, -s, and -c parameters, which perform discovery on target boards and channels, and with the -a parameter, which performs discovery on all applicable boards.

-r

Removes (deletes) the *LineAdaptOptCoef.xml* file, which restores the default configuration for channel impedance (no gain) which existed prior to any adaptation. That is, the boards will be downloaded with the default coefficients from the CONFIG or PRM file. This parameter is not used with any other parameters (it is a stand-alone parameter).

-s <slot number>

Specifies the *slot* number in the chassis where the target board resides. Must be used with the -b parameter (see the -b parameter description for details).

-t

Specifies *temporary* discovery and initialization. With this parameter, the optimum coefficients found out are not stored in the *LineAdaptOptCoef.xml* file, although summary information is recorded in the *LineAdapt.log* file. The optimum settings discovered are used to initialize the target boards when the discovery is complete. However, since the settings are not stored, they will not be used to initialize the target boards in the next download or system startup. If the -t parameter is **not** specified, the utility will discover the optimum coefficients for the specified channels and store them in the *LineAdaptOptCoef.xml* file for initialization with future downloads. The -t parameter can also be used in Prompting Mode. When using the -t parameter in Command Mode, the -b, -s, and -c parameters can be used to specify target boards and channels for discovery; otherwise, the -t parameter applies to all supported and recognized (started) boards.

-v

Specifies the flag to turn *verbose* display **on**, which displays activity messages on the screen. The -v parameter can also be used in Prompting Mode to display more detail than when verbose is off. By default verbose is off (suppressed).

1.43.6 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about configuring Dialogic® Springware and DM3 Boards, see the *Dialogic® Springware Architecture Products on Windows® Configuration Guide* and the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Note: The online bookshelf has not been updated for this feature, so the Configuration Guides do not currently include information about the line adaptation utility.

1.44 New QSIG Channel Mapping Parameter for E1 Boards

The Service Update provides a new QSIG channel mapping parameter for Dialogic® Boards that support E1 technology. This parameter specifies that bearer channel time slots be numbered sequentially from 1 to 30 for the QSIG protocol. This mapping scheme is the same as the one used on Dialogic® Springware Boards, and facilitates migration to the newer generation DM3 Boards.

This new functionality is supported on the following Dialogic® Boards:

- Dialogic® DM/V600A-2E1-PCI Media Boards
- Dialogic® DM/V1200A-4E1-PCI Media Boards
- Dialogic® DMV600BTEP Media Boards
- Dialogic® DMV1200BTEP Media Boards

1.44.1 Feature Description

Currently, bearer channel time slots on DM3 Boards are numbered from 1 to 15 and 17 to 31 for the QSIG protocol. Channel 16 is reserved for signaling. Thus, the QSIG stack will reject any calls with the channel identification information element set to 16, as it assumes that channel 16 is reserved for signaling data.

With the Service Update, you can specify that bearer channel time slots use a sequentially-ordered logical channel numbering scheme, from 1 to 30, for the QSIG protocol. This scheme conforms to the ECMA QSIG specification. (See Section 1.44.2, "Documentation", on page 152 for reference information on this specification.)

This functionality is available through a new parameter in the CONFIG file and is enabled on a trunk by trunk basis. You must **manually add** this parameter in the appropriate [CCS.x] section of the CONFIG file and turn the feature on. Next, update the corresponding FCD file by downloading the firmware to the board using the Dialogic® Configuration Manager (DCM). Changed values take effect at the time the firmware is downloaded to the board. For more information about modifying FCD file parameters, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

The new QSIG parameter is described below.

CCS_ALTQSIGCHANMAP_FLAG (Alternate QSIG Channel Mapping)

Number: 0x26

Description: This parameter enables bearer channel time slots to use a sequentially-ordered logical channel numbering scheme, from 1 to 30, for the QSIG protocol. If not enabled, bearer channel time slots are numbered from 1 to 15 and 17 to 31. In this mode, channel 16 is invalid as it is reserved for signaling.

Note: This parameter only applies to E1 boards.

Values:

- 0 (disabled) (default value)
- 1 (enabled)

Guidelines: To enable the alternate QSIG channel mapping scheme, add this new QSIG parameter in the [CCS.x] section of a CONFIG file and set to 1.

Example: This example shows the new QSIG parameter added and enabled in the [CCS.1] section of a CONFIG file:

```
...
[CCS.1]
...
SetParm=0x26,1       ! Enable QSIG sequential channel mapping scheme
```

1.44.2 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about configuration files, configuration parameters, and configuration procedures, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Note: The online bookshelf has not been updated for this feature, so this new parameter is not currently documented in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

For more information about the QSIG protocol standard, see the ECMA Private Integrated Services Network (PISN), Circuit Mode Bearer Services, Inter-Exchange Signaling Procedures and Protocol (QSIG-BC) specification on the ECMA International website at the following link: www.ecma-international.org

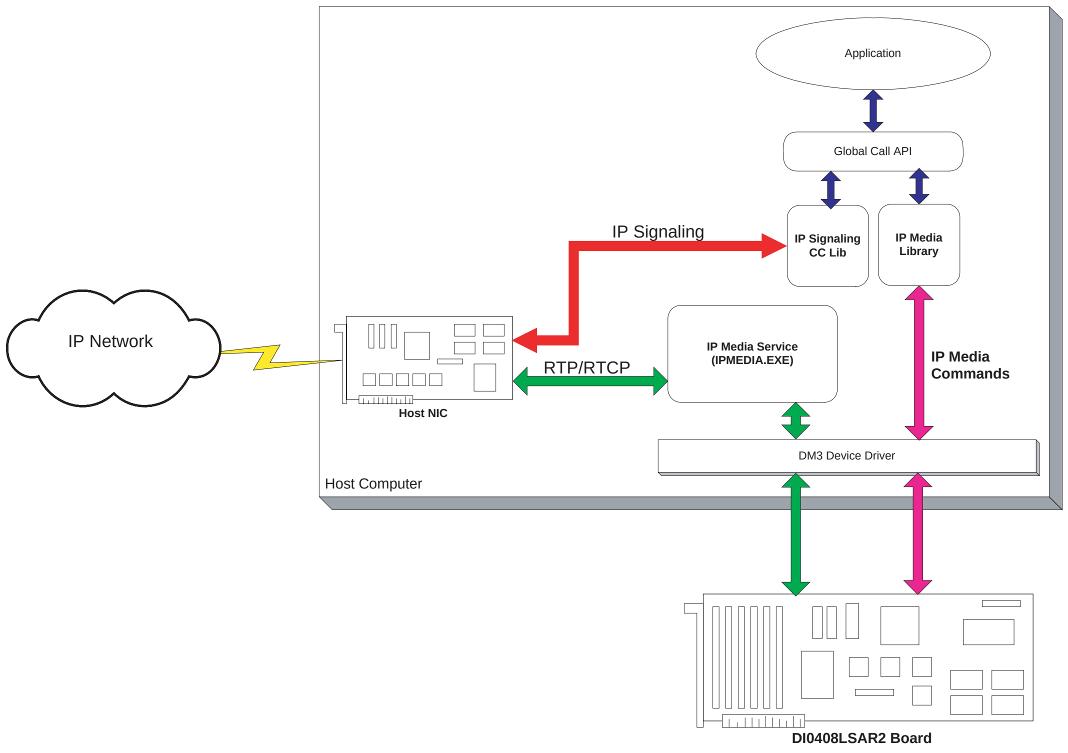
1.45 IP Support on Dialogic® DI0408LSAR2 Boards

This release supports Voice over IP (VoIP) capability on Dialogic® DI0408LSAR2 Switching Boards. This capability allows a VoIP call to be connected to the CT Bus. Using the DI0408LSAR2 Board equipped with a PSTN network front end, you can build a single-board IP-to-PSTN gateway application.

The IP resource implementation of the DI0408LSAR2 Board is equivalent to that developed for the Dialogic® DM/IP Boards as defined in the following documents:

- *Dialogic® Global Call IP Technology Guide*
- *Dialogic® IP Media Library API Programming Guide*
- *Dialogic® IP Media Library API Library Reference*

The IP resource implementation of the DI0408LSAR2 Board has unique design elements illustrated in the following figure. Unlike the DM/IP Board implementation, there is no on-board Ethernet NIC interface on the DI0408LSAR2 Board; therefore, both the IP call control and media processing are done through the host Ethernet NIC. The IP call control is implemented by host-based stack technology (call control library, IPT CCLib). The media processing of the RTP/RTCP packets is performed by the IP Media Service developed for DI0408LSAR2 media loads 3 and 5.



Note: It is also possible to specify one NIC for RTP/RTCP and, via Global Call, assign a different NIC for signaling and data. See Section 1.46, "Dialogic® DI0408LSAR2 Board Support for Host Systems with Multiple NICs", on page 159.

1.45.1 Feature Description

This release introduces support for media load 3 on the DI0408LSAR2 Board. Media load 3 provides the same features as media load 2 with the addition of IP. (For more information about media loads, see the DI0408LSAR2 Media Loads section below. There is also an IP-only media load, media load 5.) The following new features are supported on the DI0408LSAR2 Board when using media load 3:

- Call control implemented on the host by host-based stack technology (call control library, IPT CCLib). Media processing (RTP/RTCP processing) performed on the host, implemented via the IP Media Service developed for DI0408LSAR2 media load 3.
- Host-based IP stacks
- Global Call API support for IP
- IPML support for IP
- RADVISION stack; compliant with ITU-T H.323 V.4 specification, including provision for periodic registration with gatekeeper
- RADVISION SIP stack; compliant with IETF RFC 3261, the Session Initiation Protocol (SIP)

- Full-duplex communication with all coders. Supported coders are:

Coder	Frames per Packet	Frame Size (milliseconds)	VAD
G.711	1	20, 30	N/A
G.723.1, 5.3 kbps	2, 3	30	Supported
G.723.1, 6.3 kbps	2, 3	30	Supported
G.729 Annex A	2-4	10	Disabled
G.729 Annex A with Annex B	2-4	10	Enabled
GSM FR	2, 3		Supported

- H.245 tunneling
- Fast Start and Slow Start compatibility
- QoS/ToS
- RFC 2833
- IP Service Quality - jitter and packet loss
- Threshold alarms
- Simple Network Management Protocol (SNMP): TCP/IP level SNMP
- Support for standard Internet protocols, including TCP/IP, UDP, and RTP/RTCP
- Object ID support when sending non-standard command
- Support for non-standard information element in the Facility message over Q.931 port
- Vendor-specific information sending during call setup
- Voice quality parameters
- Basic DTMF and MF detection

The following features, supported in Dialogic® System Release 6.0 PCI for Windows® for DM/IP Boards, are **not** supported on the DI0408LSAR2 Board using media load 3:

- G.711 with 10 msec frame size. It is recommended that applications use 20 or 30 msec frame size with G.711. RTP/RTCP processing on the host system may cause voice quality issues at 10 msec frame size under heavy system loading.
- Multicasting
- T.38
- Single board start/stop (SBSS)
- Multiprocessor systems

DI0408LSAR2 Media Loads

Media loads are pre-defined sets of features supported by DM3 Boards. A media load consists of a configuration file set (PCD, FCD, and CONFIG files) and associated firmware loads that are downloaded to each board. In most cases, the PCD/FCD/CONFIG file names indicate the associated media load and protocol. For example, the files for media load 3 are *di0408lsa_REV2_ML3.pcd*, *di0408lsa_REV2_ML3.fcd*, and *di0408lsa_REV2_ML3.config*. See the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* for more information about media loads and configuration file sets.

There are five media loads for the DI0408LSAR2 Board. The feature sets available in each media load are described below:

Media Load 1

Media load 1 provides the following features:

- Trunks: call control, caller ID, and dedicated voice (player, recorder, tone generator, signal detector) for the four analog loop start interfaces
- Stations: call control, caller ID, and dedicated voice (player, recorder, tone generator, signal detector) for the eight analog station interfaces
- Capability to switch the signal from the audio input to the CT Bus/SCbus
- Support for up to three conferences with a total of nine parties max between all conferences. Conference resources are shareable across the system via the CT Bus/SCbus. The ability to bridge conference resources for larger conferences is supported; see Section 1.54, “Conference Bridging on Dialogic® DI Boards”, on page 221.
- Two channels of V.17 fax shareable across the system via the CT Bus/SCbus

Media Load 2

Media load 2 provides the following features:

- Eight channels of voice (player, recorder, tone generator, signal detector) shareable across the system via the CT Bus/SCbus
- Trunks: call control, caller ID, and dedicated tone generation and signal detection capability for the four analog loop start interfaces
- Stations: call control, caller ID, and dedicated tone generation and signal detection capability for the eight analog station interfaces
- Capability to switch the signal from the audio input to the CT Bus/SCbus
- Support for up to three conferences with a total of nine parties max between all conferences. Conference resources are shareable across the system via the CT Bus/SCbus. The ability to bridge conference resources for larger conferences is supported; see Section 1.54, “Conference Bridging on Dialogic® DI Boards”, on page 221.
- Two channels of V.17 fax shareable across the system via the CT Bus/SCbus

Media Load 3

Media load 3 provides the same features as media load 2 with the addition of:

- Four channels of IP telephony shareable across the system via the CT Bus/SCbus

Media Load 4

Media load 4 provides the same features as media load 2 with the addition of:

- Four channels of continuous speech processing (CSP)

Media Load 5

Media load 5 provides the following features:

- Twelve channels of IP telephony shareable across the system via the CT Bus/SCbus

DI0408LSAR2 Devices

For the DI0408LSAR2 media loads, device enumeration follows the rules listed below.

Note: The scenario below assumes that the DI0408LSAR2 Board is the only board in the system. Call the **dx_getfeaturelist( )** function to return information about the features supported on the device. (Refer to the *Dialogic® Voice API Library Reference* for function details.)

- **IPT Board Device** - A virtual entity that represents a NIC or NIC address (if one NIC supports more than one IP address). The format of the device name is **iptBx**, where **x** is the logical board number that corresponds to the NIC or NIC address. See the *Dialogic® Global Call IP Technology Guide* for more information.
- **IPT Network Device** - Represents a logical channel over which calls can be made. This device is used for call control (call setup and tear down). The format of the device name is **iptBxTy**, where **x** is the logical board number and **y** is the logical channel number. See the *Dialogic® Global Call IP Technology Guide* for more information.
- **IP Media Device** - Represents a media resource that is used to control RTP streaming, monitoring Quality of Service (QoS), and the sending and receiving of DTMF digits. The format of the device name is **ipmBxCy**, where **x** is the logical board number and **y** is the logical channel number. See the *Dialogic® Global Call IP Technology Guide* and the *Dialogic® IP Media Library API Programming Guide* for more information.
- The four loop start analog interfaces are enumerated as dtiB1T1-dtiB1T4. Trunk call control is supported via Global Call APIs.
- Voice devices associated with the four loop start interfaces are dxxxB1C1-dxxxB1C4. For media load 1, a subset of the dx_ APIs provides support for basic voice functionality. For media loads 2, 3, and 4, a subset of the dx_ APIs provides tone generation and detection support.
- The eight analog station interfaces are enumerated as msiB1C1-msiB1C8. Station call control is supported via the msi_ APIs.
- Voice devices associated with the eight analog station interfaces are dxxxB2C1-dxxxB2C4 and dxxxB3C1-dxxxB3C4. For media load 1, a subset of the dx_ APIs provides support for basic voice functionality. For media loads 2, 3, and 4, a subset of the dx_ APIs provides tone generation and detection support.
- For media loads 2, 3, and 4, eight channels of voice are enumerated as dxxxB4C1-dxxxB4C4 and dxxxB5C1-dxxxB5C4. A subset of the dx_ APIs provides support for basic voice, including transaction record.
- The audio input is enumerated as aiB1. Switching is controlled via the ai_ APIs.
- Conferencing is enumerated as dcbB1D1. Application control of conferencing is provided by either the dcb_ APIs or the ms_ conferencing APIs.
- The two fax channels are enumerated as follows:
  - For media load 1: dxxxB4C1 and dxxxB4C2
  - For media load 2: dxxxB6C1 and dxxxB6C2

  - For media load 3: dxxxB6C1 and dxxxB6C2
  - For media load 4: dxxxB7C1 and dxxxB7C2

  Application control is provided by the fx_ APIs.

- For media load 3 only: the four channels of IP are designated as follows:
  - For IPT network devices: iptB1T1-iptB1T4
  - For IP media devices: ipmB1C1-ipmB1C4
  - For multiple DI0408LSAR2 Board configurations:
    - board 2: IPT network devices: iptB1T5-iptB1T8; IP media devices: ipmB2C1-ipmB2C4
    - board 3: IPT network devices: iptB1T9-iptB1T12; IP media devices: ipmB3C1-ipmB3C4
    - ...
- For media load 5 only: the 12 channels of IP are designated as follows:
  - For IPT network devices: iptB1T1-iptB1T12
  - For IP media devices: ipmB1C1-ipmB1C12
- For media load 4 only: the four channels of continuous speech processing (CSP) are enumerated as dxxB6C1-dxxB6C4. Application control is provided by the ec_ APIs.

1.45.2 Configuring the Software

This section contains information about configuring IP parameters on DI0408LSAR2 Boards; this information supplements the configuration information in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Selecting a Firmware File

The first time you run the Dialogic® Configuration Manager (DCM) with a DM3 Board in your system, you are prompted to select the firmware files (PCD and FCD) to be downloaded to each board. As discussed above in the DI0408LSAR2 Media Loads section, the PCD/FCD files determine the media load supported by the board.

DCM displays a list of PCD files. Select the PCD file for the media load that you want, for example, *di0408lsa_REV2_ML3.pcd* for media load 3. The selected PCD file and corresponding FCD file will be downloaded when the boards are started.

Setting the ToS Parameter in DCM

With this release, a new parameter for enabling or disabling ToS has been added to DCM for DI0408LSAR2 Boards. The parameter is called **DI_TOS** and it appears on the Misc property sheet. The default value for **DI_TOS** is Enable. If **DI_TOS** is set to Disable, the IP Media Service disables ToS processing for RTP packets transmitted from IP Media channels. In this configuration, all RTP packets transmitted from DI0408LSAR2 Board IP Media channels will have the ToS field in their IP packet header set to zero.

Configuring ToS in the CONFIG File

With this release, a new parameter for configuring ToS has been added to the DI0408LSAR2 Board CONFIG files that support IP. Previously, this parameter was applicable to DM/IP Boards only.

Note: For more detailed information about modifying the CONFIG file and generating a new FCD file, refer to the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Type of Service (ToS) is a category of Quality of Service (QoS) that performs Layer 3 tagging of packets to improve the mobility of the packets. When the ToS feature is used, four bits are specified in the packet header to determine the type of service as follows:

1000: minimize delay
  selects a minimum delay link or circuit for the datagram

0100: maximize throughput
  selects a high throughput link or circuit for the datagram

0010: maximize reliability
  selects a high reliability link or circuit for the datagram

0001: minimize monetary cost
  selects a minimum cost link or circuit for the datagram

0000: normal service (not activated)
  the datagram has no priority assigned

The above are defined RFC-1349 specified values. Note that other values are valid; however, they have not been explicitly characterized in the RFC-1349 specification. The feature allows setting of any combinations of the four bits. The processing of packets with such combinations is assumed understood by the administrator and downstream routers.

There are two ways to implement the ToS feature (the same as when using ToS with DM/IP Boards):

- ToS can be activated on a channel by channel basis at run time by using the Global Call API functions **gc_SetConfigData( )**, **gc_SetUserInfo( )**, and **gc_MakeCall( )**. The Set ID = IPSET_CONFIG and the Parameter ID = IPPARM_CONFIG_TOS. For further information, see the *Dialogic® Global Call IP Technology Guide*. Settings made at run time are not persistent.
- ToS can be set at the board level, before board initialization, via the CONFIG file. This causes the ToS bits to be set to the designated value in all RTP packets transmitted.

To specify the ToS in the CONFIG file, set the following parameter:

```
[0x1d]
SetParm=0x1d01,0   ! PrmTOS (LOWDELAY 0x10 THROUGHPUT 0x08 RELIABILITY
0x04 MINCOST 0x02 No Priority 0)
```

This parameter sets the ToS bits in the IP header of transmitted datagrams to improve the mobility of packets. Values are:

0x10: LOWDELAY
: selects a minimum delay link or circuit for the datagram

0x08: THROUGHPUT
: selects a high throughput link or circuit for the datagram

0x04: RELIABILITY
: selects a high reliability link or circuit for the datagram

0x02: MINCOST
: selects a minimum cost link or circuit for the datagram

0x00: No Priority
: the datagram has no priority assigned

1.45.3 Restrictions and Limitations

The following restrictions and limitations exist for IP support on the DI0408LSAR2 Boards:

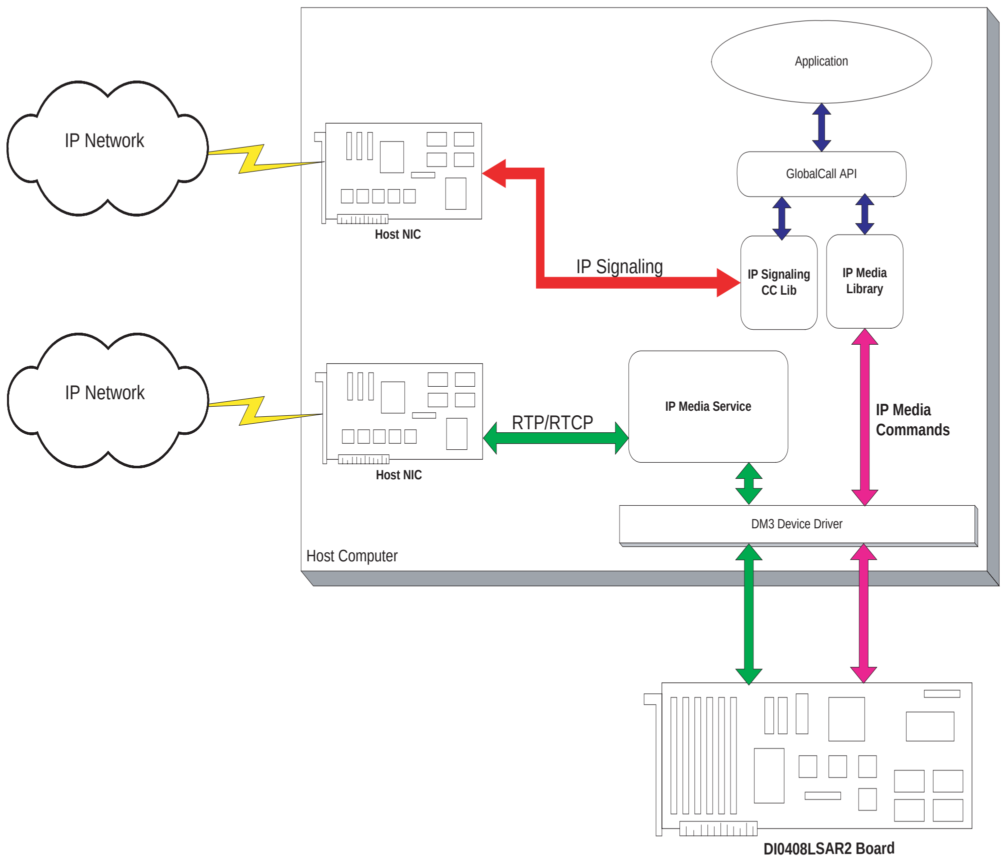
- Single processor systems only – DI0408LSAR2 ML3 and ML5 are not supported on multiprocessor systems.
- PTR 30285: When using VAD with G.729A/B codec, the application must explicitly set the VAD field in the IP_AUDIO_CAPABILITY structure to GCPV_ENABLE. Otherwise, the application could hang when invoking **gc_AnswerCall( )**.

1.46 Dialogic® DI0408LSAR2 Board Support for Host Systems with Multiple NICs

This release provides the capability to specify an explicit IP address for use by all Dialogic® DI0408LSAR2 Boards for RTP/RTCP processing in the system. If an explicit IP address is not specified, the first Network Interface Card (NIC) address returned from the socket function call **gethostbyname( )** is selected automatically and assigned as the IP address.

The explicit IP address can be set with the Dialogic® Configuration Manager (DCM) as well as with the NCM API (**NCM_SetValue( )** and **NCM_GetValue( )** functions). The new parameter is **HostIpMediaNetworkAddress** and appears on the Misc property sheet in DCM. It is a global-level parameter that applies to all DI0408LSAR2 Boards in the system. Enter the parameter value using standard decimal notation, xxx.xxx.xxx.xxx.

Note: This is the IP address for the IP media (not the IP call control signaling) and only for boards using the Host IP Media Service (not Dialogic® DM/IP Boards with on-board NICs). Refer to the figure below.



The IP Media Service uses the specified NIC IP address when establishing RTP media sessions. If no IP address was specified, the selection is done automatically as explained above.

The user-specified address is validated at board initialization to ensure that the address is recognized by the operating system. If the specified address is not valid, board initialization will fail with the reason for failure logged (i.e., invalid NIC IP address).

As with other configuration parameters, the **HostIpMediaNetworkAddress** parameter can be changed at any time; however, the change will not take effect until the system has been stopped and restarted.

1.47 Support for QSIG NCAS Calls on Dialogic® DM3 Boards

With the Service Update, the ability to initiate Non-Call Associated Signaling (NCAS) calls is supported for the QSIG protocol (E1 or T1) on Dialogic® DM3 Boards. The DM3 Boards that support this feature are:

- Dialogic® DMV1200BTEP Media Boards
- Dialogic® DMV600BTEP Media Boards
- Dialogic® DMV960A-4T1 Media Boards
- Dialogic® DMV1200A-4E1 Media Boards

The feature is only supported on media loads that use the QSIG T1 or E1 protocol, for example, ml2_qs2_qsige1.

NCAS can establish a virtual call within the network without actually associating the B channel with the call. The call only exists on the D channel, which is normally used for signaling. Once this virtual connection has been established, the customer premise equipment (CPE) can send Facility messages to the switch or terminal equipment (TE) to convey additional information. For example, Message Waiting Indicator (MWI) supplementary service information can be encoded in a Facility IE and sent in a Q.931 Setup message. (The application is responsible for encoding/decoding the Facility IE.)

1.47.1 Feature Description

NCAS allows users to communicate by user-to-user signaling without setting up a circuit-switched connection. This signaling does not occupy B channel bandwidth. A temporary signaling connection is established (and cleared) in a manner similar to the control of a circuit-switched connection.

Applications must use a specific channel for NCAS calls. For E1 interfaces, this is channel 30, that is, dtiBxT30. For T1 interfaces, this is channel 23, that is, dtiBxT23.

For outbound calls, when the call is set up with the Bearer Capabilities IE indicating that it is an NCAS call, the call is sent out on the D channel, without an associated B channel. Once the NCAS connection is established, the application can transmit user-to-user messages using the call reference number (CRN) associated with the NCAS call.

For inbound calls, the Dialogic® Software provides the ability to detect if the incoming call is an NCAS call or a standard circuit switched call by analyzing the information associated with the GCEV_OFFERED event triggered by the incoming call.

With DM3 Boards, the Dialogic Software and firmware support 8 NCAS calls per span, that is, 32 simultaneous NCAS calls per quad-span board. The 8 NCAS calls per span are **in addition** to the normal calls that you can have. For example, with T1, you can have 23 calls per span (including one on dtiBxT23), **plus** 8 NCAS calls on dtiBxT23 at the same time.

Note the following differences between NCAS implementation on Dialogic® Springware Boards and on DM3 Boards:

	DM3 Boards	Springware Boards
Channel used to make NCAS calls for T1 spans	23 (dtiBxT23)	24 (dtiBxT24)
Channel used to make NCAS calls for E1 spans	30 (dtiBxT30)	30 (dtiBxT30)
Number of simultaneous NCAS calls per D channel	8 (32 total for board with 4 spans)	16

For more information about using NCAS, see the *Dialogic® Global Call ISDN Technology Guide*.

New Parameters

For outbound calls, two new parameter IDs have been added to the GCIS_SET_BEARERCHNL parameter set for setting up NCAS calls on DM3 Boards:

GCIS_PARM_CODINGSTANDARD
: Set to ISDN_CODINGSTD_INTL or ISDN_CODINGSTD_CCITT.

GCIS_PARM_TRANSFERCAP
: Set to BEAR_CAP_UNREST_DIG.

In addition, the parameter IDs that already exist (described in the *Dialogic® Global Call ISDN Technology Guide*) are:

GCIS_PARM_TRANSFERMODE
: Set to ISDN_ITM_CIRCUIT.

GCIS_PARM_TRANSFERRATE
: Set to PACKET_TRANSFER_MODE, which is a new value for GCIS_PARM_TRANSFERRATE that has been defined for this feature.

For inbound calls, a new GCIS_SET_CALLTYPE parameter set with one new parameter has been added:

GCIS_PARM_CALL_TYPE
: Set to CALLTYPE_NCAS (to indicate an NCAS call) or CALLTYPE_CIRCUIT (to identify a standard circuit-switched call).

Outbound QSIG NCAS Call Scenarios

After opening the channel (T23 or T30), the **gc_util_insert_parm_val( )** function must be called to set up the four parameters in the GCIS_SET_BEARERCHNL parameter set. For example:

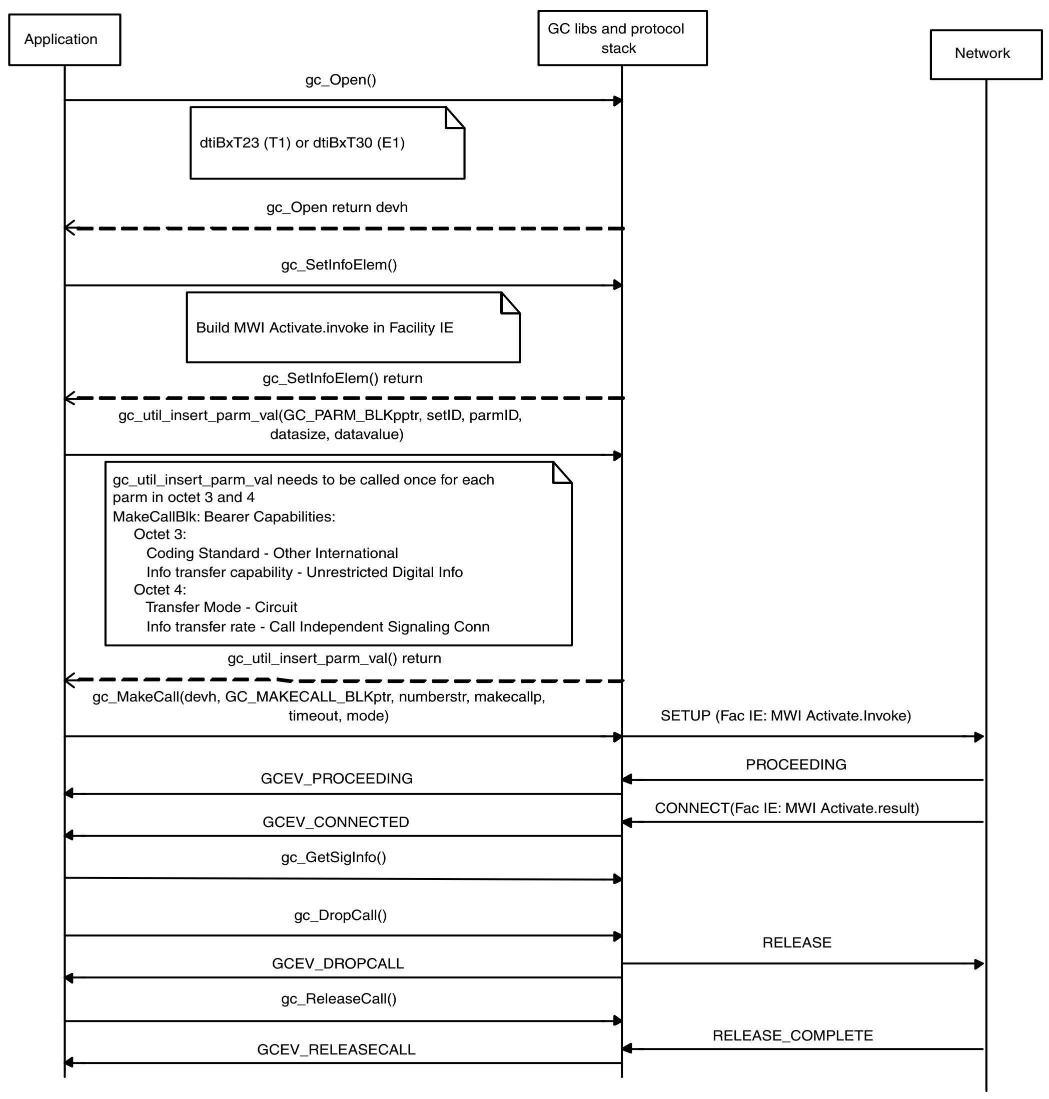
```
gc_util_insert_parm_val(pParmBlk, GCIS_SET_BEARERCHNL, GCIS_PARM_CODINGSTANDARD, sizeof(int),
ISDN_CODINGSTD_INTL);

gc_util_insert_parm_val(pParmBlk, GCIS_SET_BEARERCHNL, GCIS_PARM_TRANSFERCAP, sizeof(int),
BEAR_CAP_UNREST_DIG);

gc_util_insert_parm_val(pParmBlk, GCIS_SET_BEARERCHNL, GCIS_PARM_TRANSFERMODE, sizeof(int),
ISDN_ITM_CIRCUIT);
```

```
gc_util_insert_parm_val(pParmBlk, GCIS_SET_BEARERCHNL, GCIS_PARM_TRANSFERRATE, sizeof(int),
PACKET_TRANSFER_MODE);
```

The application must also build the Facility IE (e.g., with MWI information) using **gc_SetInfoElem( )** before making the call using **gc_MakeCall( )**.

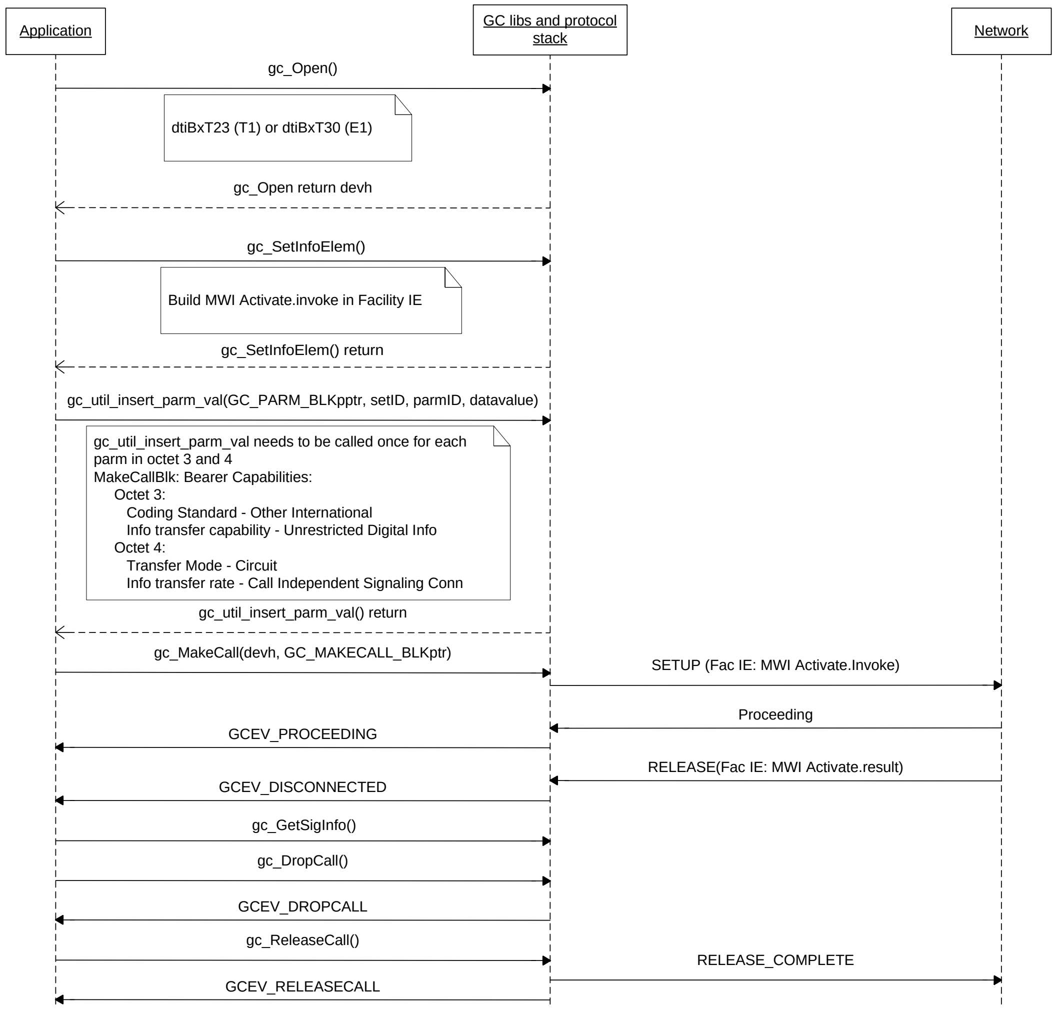
The following diagram illustrates the API sequence for an MWI activation with connect scenario.

Successful MWI Activate.Invoke (with Connect)


Application
GC libs and protocol stack
Network
gc_Open()
dtiBxT23 (T1) or dtiBxT30 (E1)
gc_Open return devh
gc_SetInfoElem()
Build MWI Activate.invoke in Facility IE
gc_SetInfoElem() return
gc_util_insert_parm_val(GC_PARM_BLKpptr, setID, parmID, datasize, datavalue)
gc_util_insert_parm_val needs to be called once for each parm in octet 3 and 4
MakeCallBlk: Bearer Capabilities:
Octet 3:
Coding Standard - Other International
Info transfer capability - Unrestricted Digital Info
Octet 4:
Transfer Mode - Circuit
Info transfer rate - Call Independent Signaling Conn
gc_util_insert_parm_val() return
gc_MakeCall(devh, GC_MAKECALL_BLKptr, numberstr, makecallp, timeout, mode)
SETUP (Fac IE: MWI Activate.Invoke)
PROCEEDING
GCEV_PROCEEDING
CONNECT(Fac IE: MWI Activate.result)
GCEV_CONNECTED
gc_GetSigInfo()
gc_DropCall()
RELEASE
GCEV_DROPCALL
gc_ReleaseCall()
RELEASE_COMPLETE
GCEV_RELEASECALL

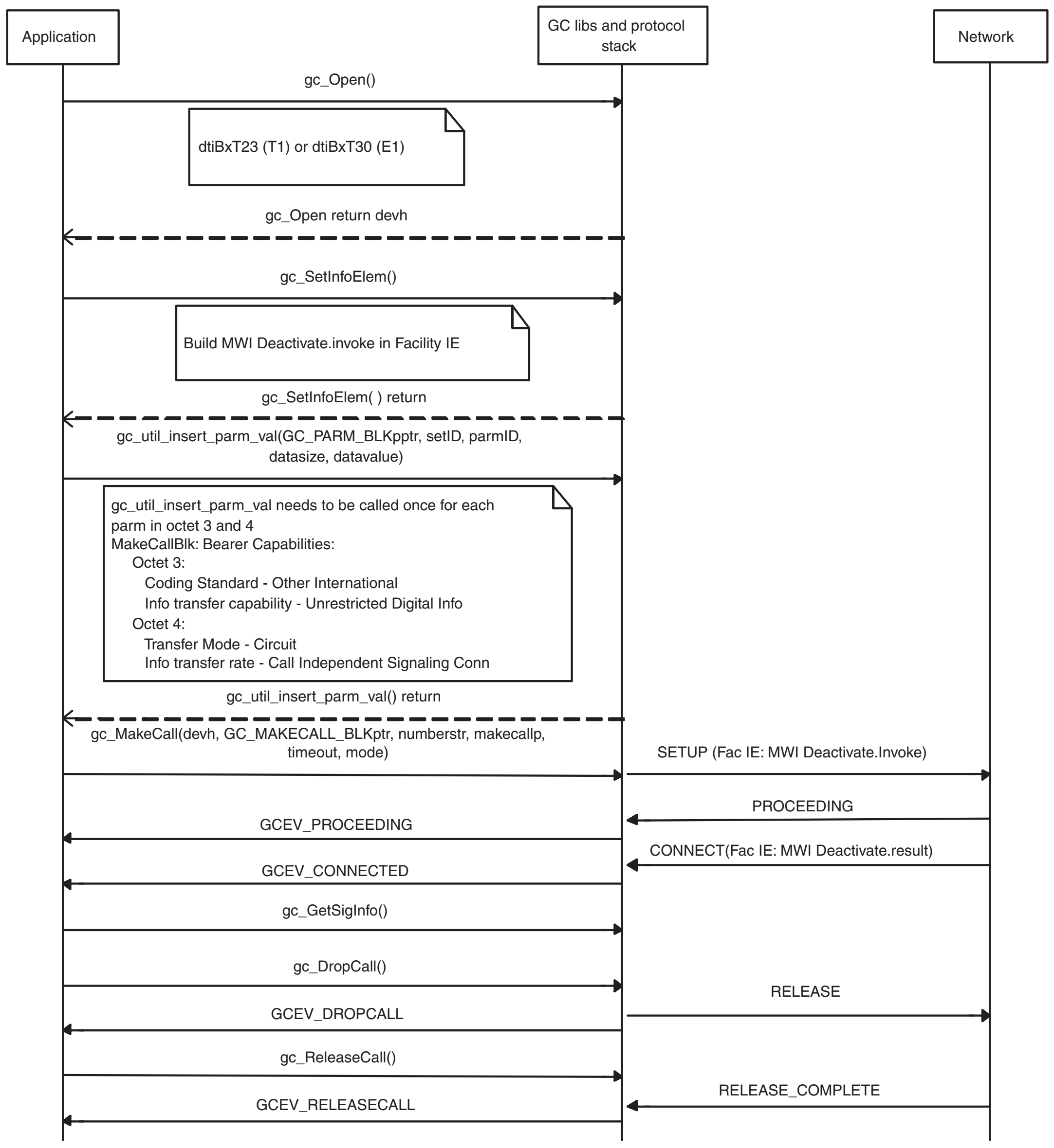
The following diagram illustrates the API sequence for an MWI activation without connect scenario.

Successful MWI Activate.Invoke (without Connect)


Application
GC libs and protocol stack
Network
gc_Open()
dtiBxT23 (T1) or dtiBxT30 (E1)
gc_Open return devh
gc_SetInfoElem()
Build MWI Activate.invoke in Facility IE
gc_SetInfoElem() return
gc_util_insert_parm_val(GC_PARM_BLKpptr, setID, parmID, datavalue)
gc_util_insert_parm_val needs to be called once for each parm in octet 3 and 4
MakeCallBlk: Bearer Capabilities:
Octet 3:
Coding Standard - Other International
Info transfer capability - Unrestricted Digital Info
Octet 4:
Transfer Mode - Circuit
Info transfer rate - Call Independent Signaling Conn
gc_util_insert_parm_val() return
gc_MakeCall(devh, GC_MAKECALL_BLKptr)
SETUP (Fac IE: MWI Activate.Invoke)
Proceeding
GCEV_PROCEEDING
RELEASE(Fac IE: MWI Activate.result)
GCEV_DISCONNECTED
gc_GetSigInfo()
gc_DropCall()
GCEV_DROPCALL
gc_ReleaseCall()
RELEASE_COMPLETE
GCEV_RELEASECALL

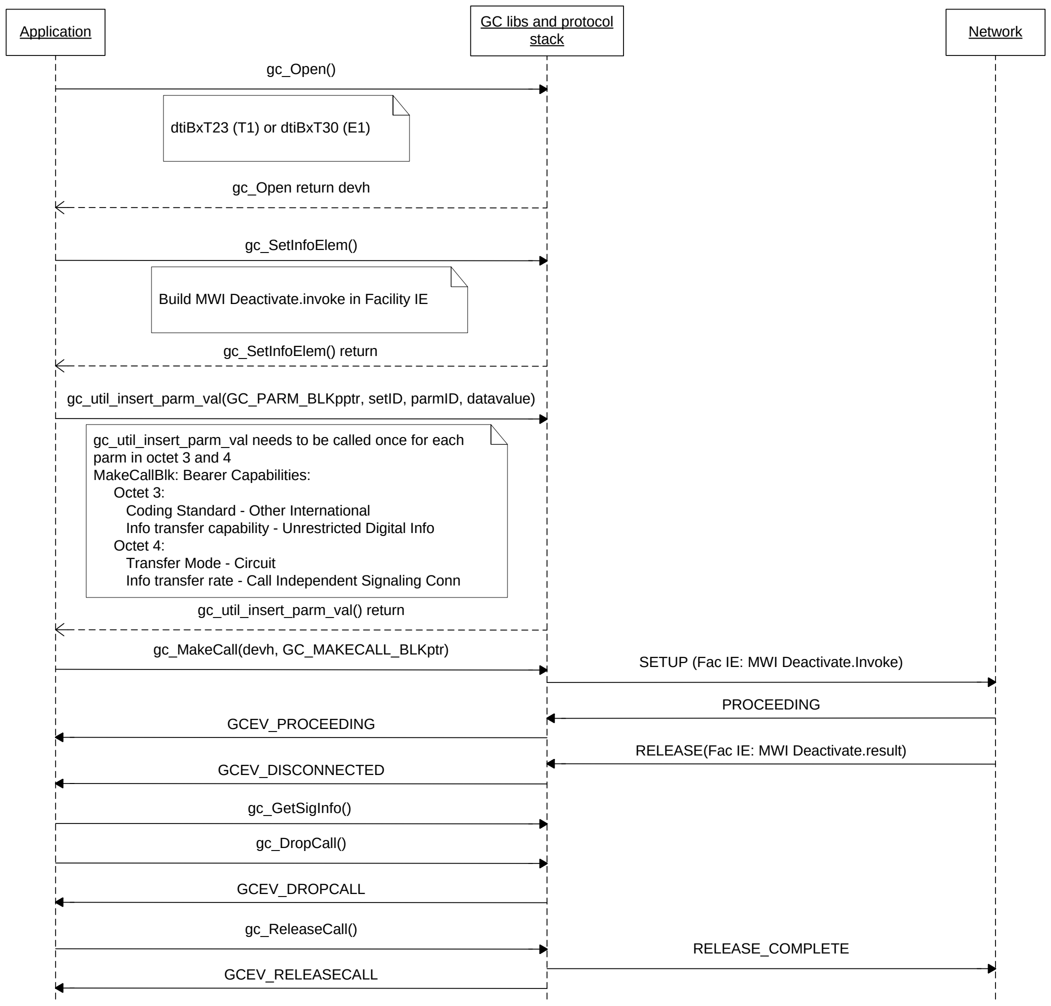
The following diagram illustrates the API sequence for an MWI deactivate with connect scenario.

Successful MWI Deactivate.Invoke (with Connect)


Application
GC libs and protocol stack
Network
gc_Open()
dtiBxT23 (T1) or dtiBxT30 (E1)
gc_Open return devh
gc_SetInfoElem()
Build MWI Deactivate.invoke in Facility IE
gc_SetInfoElem( ) return
gc_util_insert_parm_val(GC_PARM_BLKpptr, setID, parmID, datasize, datavalue)
gc_util_insert_parm_val needs to be called once for each parm in octet 3 and 4
MakeCallBlk: Bearer Capabilities:
Octet 3:
Coding Standard - Other International
Info transfer capability - Unrestricted Digital Info
Octet 4:
Transfer Mode - Circuit
Info transfer rate - Call Independent Signaling Conn
gc_util_insert_parm_val() return
gc_MakeCall(devh, GC_MAKECALL_BLKptr, numberstr, makecallp, timeout, mode)
SETUP (Fac IE: MWI Deactivate.Invoke)
PROCEEDING
GCEV_PROCEEDING
CONNECT(Fac IE: MWI Deactivate.result)
GCEV_CONNECTED
gc_GetSigInfo()
gc_DropCall()
RELEASE
GCEV_DROPCALL
gc_ReleaseCall()
RELEASE_COMPLETE
GCEV_RELEASECALL

The following diagram illustrates the API sequence for an MWI deactivate without connect scenario.

Successful MWI Deactivate.Invoke (without Connect)

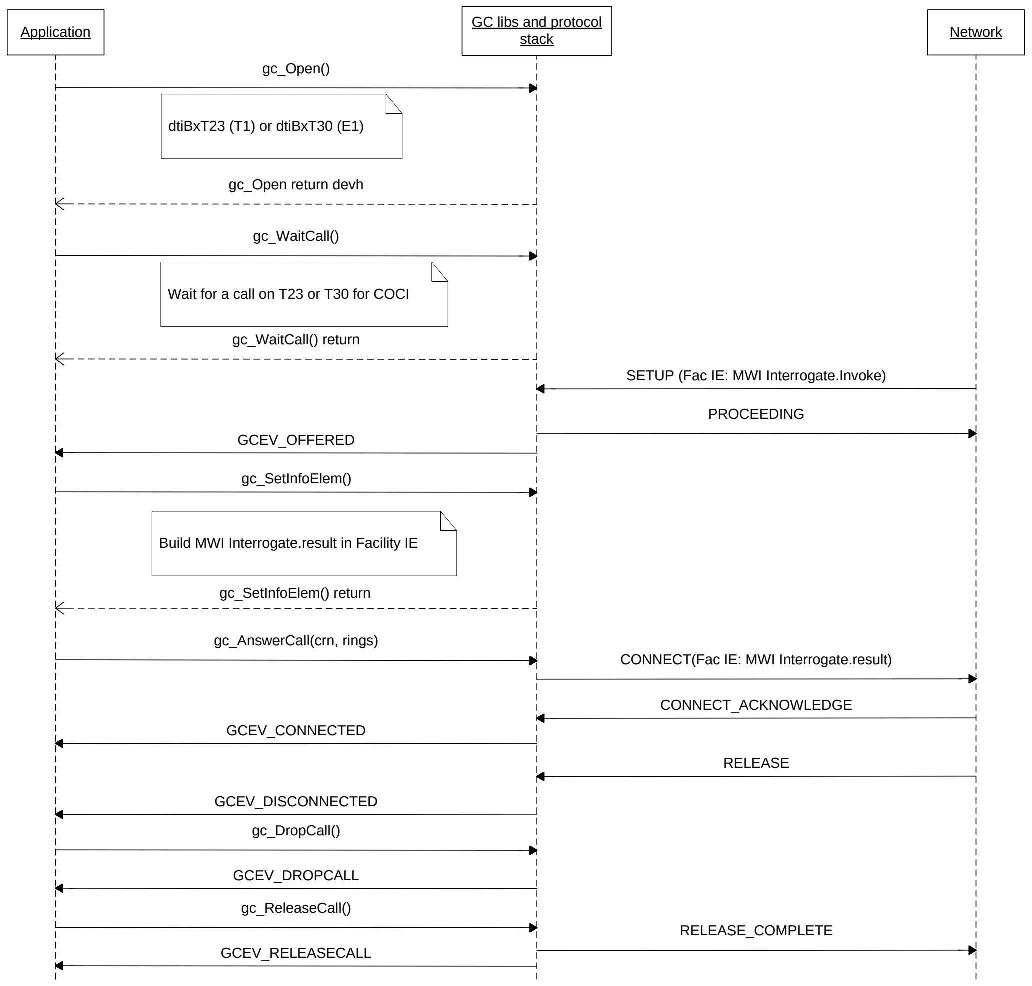


Application
GC libs and protocol stack
Network
gc_Open()
dtiBxT23 (T1) or dtiBxT30 (E1)
gc_Open return devh
gc_SetInfoElem()
Build MWI Deactivate.invoke in Facility IE
gc_SetInfoElem() return
gc_util_insert_parm_val(GC_PARM_BLKpptr, setID, parmID, datavalue)
gc_util_insert_parm_val needs to be called once for each parm in octet 3 and 4
MakeCallBlk: Bearer Capabilities:
Octet 3:
Coding Standard - Other International
Info transfer capability - Unrestricted Digital Info
Octet 4:
Transfer Mode - Circuit
Info transfer rate - Call Independent Signaling Conn
gc_util_insert_parm_val() return
gc_MakeCall(devh, GC_MAKECALL_BLKptr)
SETUP (Fac IE: MWI Deactivate.Invoke)
PROCEEDING
GCEV_PROCEEDING
RELEASE(Fac IE: MWI Deactivate.result)
GCEV_DISCONNECTED
gc_GetSigInfo()
gc_DropCall()
GCEV_DROPCALL
gc_ReleaseCall()
RELEASE_COMPLETE
GCEV_RELEASECALL

Inbound QSIG NCAS Call Scenarios

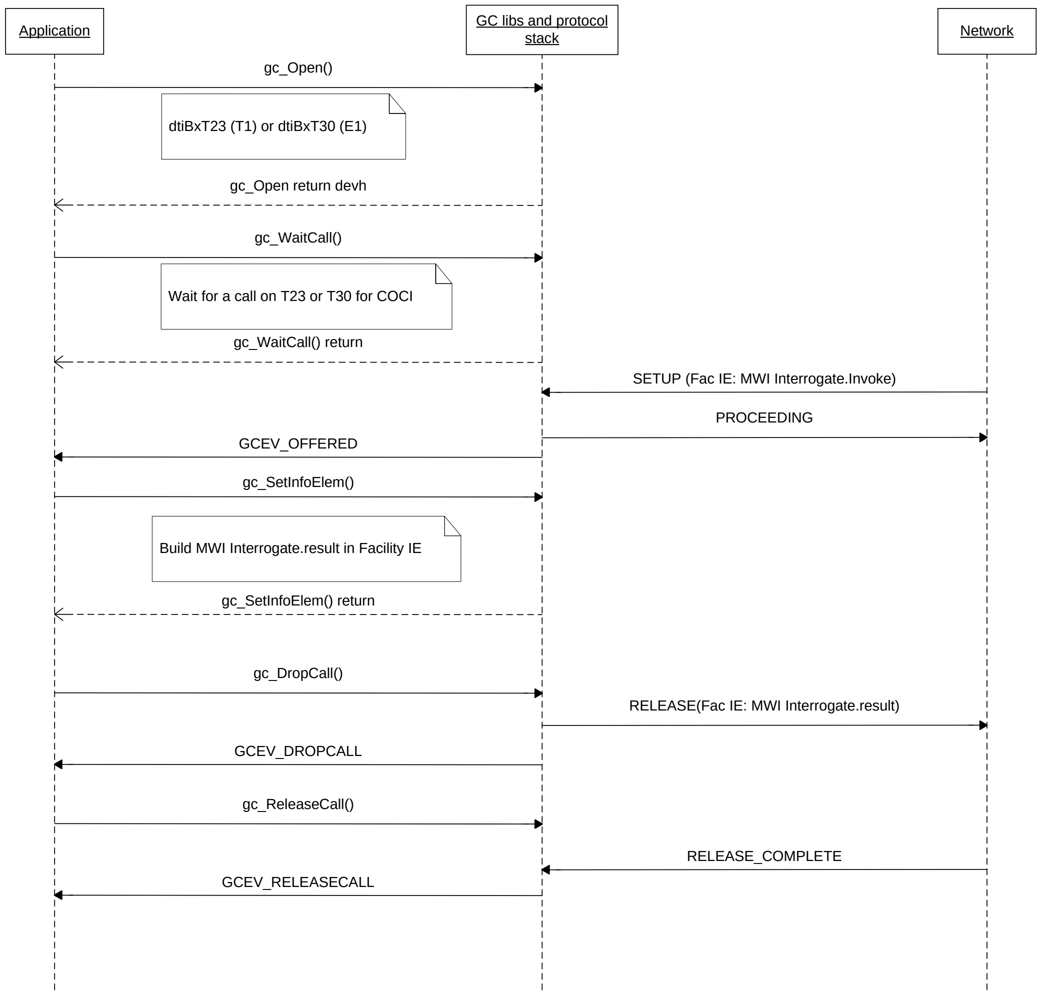
The following diagram illustrates the API sequence for an MWI interrogate with connect scenario.

Successful MWI Interrogate.Result (with Connect)



The following diagram illustrates the API sequence for an MWI interrogate without connect scenario.

Successful MWI Interrogate.Result (without Connect)


Application
GC libs and protocol stack
Network
gc_Open()
dtiBxT23 (T1) or dtiBxT30 (E1)
gc_Open return devh
gc_WaitCall()
Wait for a call on T23 or T30 for COCI
gc_WaitCall() return
SETUP (Fac IE: MWI Interrogate.Invoke)
PROCEEDING
GCEV_OFFERED
gc_SetInfoElem()
Build MWI Interrogate.result in Facility IE
gc_SetInfoElem() return
gc_DropCall()
RELEASE(Fac IE: MWI Interrogate.result)
GCEV_DROPCALL
gc_ReleaseCall()
RELEASE_COMPLETE
GCEV_RELEASECALL

Inbound QSIG NCAS Call Detection Code Example

The following code segment demonstrates how to retrieve the call type from the GCEV_OFFERED event triggered by an incoming call to determine if the call is an NCAS call or a standard circuit-switched call.

```
case GCEV_OFFERED:
    GC_PARM_BLKP gcParmBlkp = NULL;
    GC_PARM_DATAP t_gcParmDatap = NULL;
    EXTENSIONEVTBLK *ext_evtblkp = NULL;

    ext_evtblkp = (EXTENSIONEVTBLK *)meta_event.extevtdatap;
    gcParmBlkp = &ext_evtblkp->parmblk;
    while (t_gcParmDatap = gc_util_next_parm(gcParmBlkp, t_gcParmDatap))
    {
        switch (t_gcParmDatap->set_ID)
        {
            case GCIS_SET_CALLTYPE:
                switch(t_gcParmDatap->parm_ID)
                {
                    case GC_PARM_CALL_TYPE:
                    // Determine the Call Type.
                    switch (t_gcParmDatap->value_buf)
                    {
                        case CALLTYPE_NCAS:
                            cout << "NCAS call detected" << endl;
                            break;
                        case CALLTYPE_CIRCUIT:
                            cout << "Regular call detected" << endl;
                            break;
                    }
                break;

            default:
                cout << "Unknown PARM ID" << endl;
                break;
            }
        break;

        default:
            cout << "Unknown SET ID" << endl;
            break;
        }
    }
    break;
```

1.47.2 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the Dialogic® Global Call API, see the following documents:

- *Dialogic® Global Call API Programming Guide*
- *Dialogic® Global Call API Library Reference*

For features specific to ISDN technology, see:

- *Dialogic® Global Call ISDN Technology Guide*

Note: The online bookshelf has not been updated for this feature, so the *Dialogic® Global Call ISDN Technology Guide* does not currently indicate that NCAS is supported for QSIG on DM3 Boards.

1.48 Loop Current Reversal Detection on Dialogic® DMV160LP Boards

The Service Update adds support for loop current reversal detection on the Dialogic® DMV160LP Media Board.

1.48.1 Feature Description

Certain service providers furnish polarity reversal to subscribers to signal that the called (far end) party has answered a call (polarity reversal on seizure). Upon detection of polarity reversal, the call should transition to the CONNECTED state.

With Dialogic® Springware Boards, the Voice API **dx_setevtmsk( )** function can be used to enable detection of loop current on, loop current off, and loop current reversal call status transition events (DM_LCOFF, DM_LCON, DM_LCREV). However, detection of these events using **dx_setevtmsk( )** was not supported for DMV160LP Boards.

With this new feature, the Global Call API **gc_SetConfigData( )** function can now be used to provide similar functionality for DMV160LP Boards. The unsolicited GCEV_EXTENSION event is used to notify the application when any of the new call status transitions occur:

CC_CST_LCON
    loop current on detected

CC_CST_LCOFF
    loop current off detected

CC_CST_LCREV
    loop current reversal detected

A typical sequence of events is:

Call Flow Sequence	Function	Events Received
Pre-call - Loop current is off.		
Application makes outbound call.	**gc_MakeCall( )**	GCEV_EXTENSION(CC_CST_LCON) GCEV_DIALING GCEV_ALERTING
Called (far end) party answers the call.		GCEV_EXTENSION(CC_CST_LCREV) GCEV_CONNECTED
Near end drops the call.	**gc_DropCall( )**	GCEV_EXTENSION(CC_CST_LCOFF) GCEV_DROPCALL
Application frees the device for another call.	**gc_ReleaseCallEx( )**	GCEV_RELEASECALL

Note: An extra LCON event may be seen after the first expected LCON is received but before LCREV is received. This may occur due to the polarity reversal. When a polarity reversal occurs, there is a momentary loss of loop current. If this duration is 100+ ms, this triggers an LCON event (since an LCON event is defined as a transition from no loop current -> loop current). The longer the duration of no loop current, the more likely it is that this will cause the extra LCON event. Since the LCON is preceded by the LS_Net_Answer CAS signal (seen in log with TSPTrace), this extra event is not an error, but merely the current line condition.

Also, an extra LCREV event may be seen before LCOFF. This is because the Central Office may reverse the polarity on the line, to counteract the polarity reversal done previously, before terminating the call. This sequence of events is acceptable but may not be seen in all cases. It depends on the Central Office.

1.48.2 Enabling Reception of the GCEV_EXTENSION Event

The GCEV_EXTENSION event indicates that unsolicited information is received from the network or remote end point. Information about the event is contained in the EXTENSIONEVTBLK structure, which is referenced via the extevtdatap pointer in the METAEVENT structure associated with the GCEV_EXTENSION event.

The new GCEV_EXTENSION events are disabled by default. Use the standard Global Call procedure for enabling reception of the GCEV_EXTENSION event by using **gc_util_insert_parm_val( )** to build a GC_PARM_BLK, followed by **gc_SetConfigData( )** to enable the event. See the *Dialogic® Global Call API Library Reference* for further information about these functions.

The following code snippets show how to enable and process the GCEV_EXTENSION event. Note that:

- EXTENSIONEVT_CALLSTATUS_TRANSITION is the new bitmask to enable/disable reception of call status transition events.
- CCSET_CALLSTATUS_TRANSITION is the setID.
- CCPARM_CST_TYPE is the parmID, with values:
  - CC_CST_LCON
  - CC_CST_LCOFF

- CC_CST_LCREV

These defines will be part of the GC_PARM_BLK structure that will be associated with the GCEV_EXTENSION event that the application receives. The application parses the GC_PARM_BLK to determine the call status transition reason.

Enable GCEV_EXTENSION Event

```
int EnableCallStatusInformation()
{

     GC_PARM_BLKP pParmBlock = NULL;
     long requestID;

     int iRetCode = gc_util_insert_parm_val(&pParmBlock, CCSET_EXTENSIONEVT_MSK,
          GCACT_ADDMSK, sizeof(long), EXTENSIONEVT_CALLSTATUS_TRANSITION);

     int rc = gc_SetConfigData(GCTGT_CCLIB_CHAN,
          m_DevHdl,
          pParmBlock,
          0,
          GCUPDATE_IMMEDIATE,
          &requestID,
          EV_ASYNC);

     if(rc != GC_SUCCESS) {
          cout << "failed to set evt mask" << endl;
          return GC_ERROR;
     } else {
          Cout << "gc_SetConfigData() called - Call Status Transition event reception enabled"
                 << endl;
     }

     gc_util_delete_parm_blk(pParmBlock);

     return 0;

}
```

Process GCEV_EXTENSION Event

```
GC_PARM_BLKP gcParmBlkp = NULL;
GC_PARM_DATAP t_gcParmDatap = NULL;
EXTENSIONEVTBLK *ext_evtblkp = NULL;

ext_evtblkp = (EXTENSIONEVTBLK *)meta_event.extevtdatap;
gcParmBlkp = &ext_evtblkp->parmblk;

cout << "Received GCEV_EXTENSION event with ExtID = " << ext_evtblkp->ext_id << endl;
while (t_gcParmDatap = gc_util_next_parm(gcParmBlkp, t_gcParmDatap))
{
     switch (t_gcParmDatap->set_ID)
     {
          case CCSET_CALLSTATUS_TRANSITION:
               switch(t_gcParmDatap->parm_ID)
               {
                    case CCPARM_CST_TYPE:
                         // Determine the CST Type.
                         switch (t_gcParmDatap->value_buf)
                         {
                              case CC_CST_LCON:
                                   cout << "LCON detected" << endl;
                              break;
                              case CC_CST_LCOFF:
                                   cout << "LCOFF detected" << endl;
                              break;
                              case CC_CST_LCREV:
                                   cout << "LCREV detected" << endl;
                              break;
                         }
                    break;
                    default:
                         cout << "Unknown PARM ID" << endl;
                    break;
               }
               break;
          default:
               cout << "Unknown SET ID" << endl;
          break;
     }
}
```

1.48.3 Updating the CONFIG File

After installing the Service Update, the following parameter requires configuration in the *dmv160lp.config* file in order to receive polarity reversal events:

- In the **Variant 2** section of file, change **Variant PolarityDetection** from 0 to 1.

Whenever a CONFIG file has been modified, a new FCD file must be generated. This procedure is described in detail in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

1.48.4 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about generic Dialogic® Global Call features, see the following documents:

- *Dialogic® Global Call API Programming Guide*
- *Dialogic® Global Call API Library Reference*

For features specific to Analog technology, see:

- *Dialogic® Global Call Analog Technology Guide*

Note: The online bookshelf has not been updated for this feature, so the *Dialogic® Global Call Analog Technology Guide* does not currently discuss loop current reversal detection on the DMV160LP Board.

1.49 Adjusting DTMF Characteristics through the CONFIG File

With the Service Update, Dialogic® DM/V-A and DM/V-B Media Boards now support the ability to modify DTMF parameter values. This new functionality is provided through the configuration file set. Changed values take effect at the time the firmware is downloaded to the board using the Dialogic® Configuration Manager (DCM) utility.

This new functionality is supported by the following boards:

- Dialogic® DM/V480A-2T1 Media Boards
- Dialogic® DM/V600A-2E1 Media Boards
- Dialogic® DM/V960A-4T1 Media Boards
- Dialogic® DM/V1200A-4E1 Media Boards
- Dialogic® DMV600BTEP Media Boards
- Dialogic® DMV1200BTEP Media Boards
- Dialogic® DMV3600BP Media Boards

1.49.1 Feature Description

Previously, DTMF characteristics were hardcoded and not adjustable by the user. With the Service Update, you can now adjust DTMF parameter values, such as amplitudes and on/off durations, in the Tone Templates [tonegen] section of a particular media load CONFIG file. Default values are provided that are consistent with previous service updates and system releases to preserve backward compatibility.

After adjusting one or more DTMF parameter values in a CONFIG file, you must download the firmware to the board using the Dialogic® Configuration Manager (DCM). For more information on modifying FCD file parameters, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

DTMF parameter values are not modifiable through API functions.

Note: Use caution when modifying DTMF parameter values. The DCM does not prevent you from specifying values outside of industry specifications.

1.49.2 DTMF Characteristics and Default Values

The DTMF characteristics, such as amplitudes and on/off durations, are described as follows for each DTMF:

Table 2. DTMF Characteristics and Default Values

DTMF	Characteristic	Default Value
DTMF 1	Signal Id	58977
	Label	1
	Segment Count	1
	Segment Signal Type	2
	Segment Frequency 1 (Hz)	697
	Segment Amplitude 1 (.25 dbm)	-24
	Segment Frequency 2 (Hz)	1209
	Segment Amplitude 2 (.25 dbm)	-24
	Segment On Duration (125 µs)	800
	Segment Off Duration (125 µs)	400
	Segment Reps	1
	Next Segment	65535
DTMF 2	Signal Id	58978
	Label	2
	Segment Frequency 1	697
	Segment Frequency 2	1336
DTMF 3	Signal Id	58979
	Label	3
	Segment Frequency 1	697
	Segment Frequency 2	1477
DTMF 4	Signal Id	58980
	Label	4
	Segment Frequency 1	770
	Segment Frequency 2	1209
DTMF 5	Signal Id	58981
	Label	5
	Segment Frequency 1	770
	Segment Frequency 2	1336
DTMF 6	Signal Id	58982
	Label	6
	Segment Frequency 1	770
	Segment Frequency 2	1477

Table 2. DTMF Characteristics and Default Values (Continued)

DTMF	Characteristic	Default Value
DTMF 7	Signal Id	58983
	Label	7
	Segment Frequency 1	852
	Segment Frequency 2	1209
DTMF 8	Signal Id	58984
	Label	8
	Segment Frequency 1	852
	Segment Frequency 2	1336
DTMF 9	Signal Id	58985
	Label	9
	Segment Frequency 1	852
	Segment Frequency 2	1477
DTMF 0	Signal Id	58986
	Label	0
	Segment Frequency 1	931
	Segment Frequency 2	1336
DTMF a	Signal Id	58987
	Label	A
	Segment Frequency 1	697
	Segment Frequency 2	1633
DTMF b	Signal Id	58988
	Label	B
	Segment Frequency 1	770
	Segment Frequency 2	1633
DTMF c	Signal Id	58989
	Label	C
	Segment Frequency 1	852
	Segment Frequency 2	1633
DTMF d	Signal Id	58990
	Label	D
	Segment Frequency 1	941
	Segment Frequency 2	1633

Table 2. DTMF Characteristics and Default Values (Continued)

DTMF	Characteristic	Default Value
DTMF #	Signal Id	58991
	Label	#
	Segment Frequency 1	941
	Segment Frequency 2	1477
DTMF *	Signal Id	58992
	Label	*
	Segment Frequency 1	941
	Segment Frequency 2	1209
DTMF comma	Signal Id	58993
	Label	CommaHack
	Segment Count	1
	Segment Signal Type	1
	Segment Frequency 1 (Hz)	0
	Segment Amplitude 1 (dbm)	0
	Segment Frequency 2 (Hz)	0
	Segment Amplitude 2 (dbm)	0
	Segment On Duration (125 μs)	0
	Segment Off Duration (125 μs)	20000
	Segment Reps	1
	Next Segment	65535

1.49.3 Media Loads and CAS Protocols Supported

On DM/V-A Boards, the following media loads support the new functionality to modify DTMF parameter values: media load 2 and media load 5bc for T1 and E1 protocols.

On DM/V-B Boards, all media loads support the new functionality except for media loads 9b, 9c, and 9d (these support conferencing only).

The following CAS protocols support the new functionality: pdk_us_ls_fxs_io (T1 CAS), pdk_us_mf_io (T1 CAS), pdk_sw_e1_mcls_io (E1 CAS), and pdk_sw_e1_luls_io (E1 CAS).

1.49.4 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about configuration files, configuration parameters, and configuration procedures, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Note: The online bookshelf has not been updated for this feature, so the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* does not currently include information about adjusting DTMF characteristics through the CONFIG file.

1.50 Single Board Start/Stop for Selected Dialogic®JCT Boards

The ability to stop and start a single Dialogic® JCT Board (after the system has initially started) is now supported. With this feature, it is not necessary to shut down the entire system while repairing a defective board or resetting a blocked channel.

Related to this is another new feature, firmware assert notification for JCT Boards. With this feature, an application can be notified when a firmware assert takes place, so the application can stop sending calls to the board. Previously, the application had to wait for time-outs, open failures, etc., to determine that a board was not working.

Single board start/stop is supported on the following boards:

- Dialogic® D/41JCT-LS Media Boards
- Dialogic® D/120JCT-LS Media Boards
- Dialogic® D/82JCT-U Media Boards
- Dialogic® D/480JCT-2T1 Media Boards
- Dialogic® D/600JCT-2E1 Media Boards

Other JCT Boards can co-exist with these boards, but they will not be capable of single board start/stop.

1.50.1 Stopping and Starting Boards

Single boards can be stopped and started using the **NCM_StopBoard( )** and **NCM_StartBoard( )** functions. Single boards can also be stopped and started using **Stop Device** and **Start Device** in the Dialogic® Configuration Manager (DCM).

For single board start/stop capable boards to be stopped and started, the entire system must be downloaded first. The system must be started, and only those boards that have been detected, downloaded, and enabled are candidates for single board start/stop.

Boards that are disabled or did not start during the system initialization cannot be started using the single board start/stop feature. The entire system must be stopped and then restarted with all desired boards up and running prior to any single board start/stop activity.

Changing a board's characteristics (e.g., increasing/decreasing the number of devices) is not allowed when performing a single board start/stop operation. No loads or any parameter changes that could impact the density are allowed, otherwise download will fail. If this has to be done, then the entire system has to be stopped and restarted.

Addressable unit identifiers (AUIDs) may change for virtual boards (e.g., dxxxB4) and virtual devices (e.g., dtiB2C3) after a single board stop/start. After each single board stop/start, the board name (e.g., brdBx) and AUID should be retrieved in order to perform any board operations (**brd_SendAlive( )**, **brd_Open( )**, etc.). For information about AUIDs, see the *Dialogic® System Software for PCI Products on Windows® Administration Guide*. For information about **brd_SendAlive( )** and other brd_ operations, see the *Dialogic® Board Management API Library Reference*.

Notes: ***1.*** To use the single board start/stop feature, each board in the system must have a unique Board ID. For information about setting Board IDs, see the Dialogic® Quick Install Card that comes with the board.

2. Single board start/stop does **not** work if **Start Selective (Good Devices Only)** has been specified from the DCM Settings menu.

3. Single board start/stop is supported only in H.100 (CT Bus) mode. The bus mode is specified by the **TDM Bus Type (User Defined)** parameter in DCM.

Recommended and Mandatory Operations

This section describes mandatory and recommended procedures that must/should be followed when performing a single board stop/start operation.

- Before stopping any board, all active devices (i.e., devices that have been opened and have a valid handle opened retuned from the open request) **must be closed** prior to issuing the stop request. It is the responsibility of the calling application to ensure that each device associated with the target board is closed via a device_close API call (e.g., **dx_close ( )**).
- It is recommended that a stop also be invoked on any active device prior to issuing a stop board request. In the case of a firmware assert, this is not required, as there is no guarantee that a response will be sent from the firmware. Nevertheless, it is good practice to issue both a stop, and then a close prior to issuing a stop board request. The **recommended** sequence is as follows:

  a. Perform a stop on all active devices (e.g., **dx_stop( )**).

  b. Perform a close on all active devices (e.g., **dx_close( )**).
- It is **mandatory** that the application perform a device close on all active devices.

Note: Performing a single board stop/start could potentially result in unrecoverable memory (approximately 5K per active device) if active devices are not closed prior to the single board stop/start. This could eventually lead to degraded system performance over extended periods of time.

1.50.2 Stand-Alone Configuration

A new feature is supported for system configuration, with each board functioning independently without TDM bus connectivity. Each board is configured as a Primary Master, deriving its clock reference either externally via its first network interface (digital boards) or by using its internal oscillator (analog boards). No inter-board connectivity (i.e., routing or resource sharing) is possible. This configuration is considered a stand-alone configuration that will eliminate a single point of failure with respect to clocking. This feature is intended for D/41JCT-LS, D/120JCT-LS, and D/82JCT-U system configuration.

This configuration is supported only in H.100 (CT Bus) mode. Single board start/stop may also be performed on the boards listed above. Each board can be stopped or started without affecting clocking for any other board. An option must be selected prior to system initialization. (See description of **Using Cable Mode** parameter below.)

Note: There cannot be a mixed CT Bus and non-CT Bus configuration (e.g., three boards cabled and two boards not attached).

The following new parameters have been added.

NFASPrimary

The **NFASPrimary** parameter, which is only for single board start/stop capable boards, is read-only parameter with a value of Yes or No. The default value is No. The value is Yes if the user has configured even one span of a board to be the NFASPrimary. Programmatically, an application can query the parameter value via **NCM_GetValueEx( )** prior to invoking **NCM_StopBoard( )** on a board chosen to be stopped. With DCM, if a board that is an NFAS Master is being stopped, a warning dialog box is displayed and the user has the option to continue stopping the board or to exit without stopping the board. This option is applicable to ISDN capable boards only (e.g., D/480JCT-2T1, D/600JCT-2E1).

Using Cable Mode

The **Using Cable Mode** parameter has been added to the Bus-0 page under TDM Bus in DCM. Its value is initialized to Default, which applies to a configuration where the boards are connected using the CT Bus cable. Other possible values are Yes and No. (Yes is the same as Default.) This parameter is intended to be set when operating in stand-alone configuration. To operate in stand-alone configuration, this parameter has to be set to No and the CT Bus cable physically removed. If the value is set to No and the CT Bus cable is not removed, download will fail.

In stand-alone configuration, each board is configured as a Primary Master.

The following restrictions apply when using stand-alone configuration:

- Stand-alone configuration can be used only with the following boards: D/41JCT-LS, D/120JCT-LS, and D/82JCT-U.
- Each board is downloaded as a Primary Master capable of producing its own clock. References to Secondary Master, Reference Master, and Slaves are not applicable in this configuration.
- The TDM Bus 0 information should be ignored for all clocking related information.

1.50.3 Firmware Assert Notification

In order to enable the firmware assert notification feature, an application should call **brd_Open( )** with new a **mode** parameter, **BRD_FW_ASSERT_ENABLE**. (Previously, the **mode** parameter was documented as reserved for future use.) The new **mode** parameter is found in *devmgmt.h* header. The application will need to link with the device management library (libdevmgmt).

Each physical board that the application wants firmware assert notification for must be opened with **brd_Open( )**. If a firmware assert occurs, a new event, DMEV_FW_ASSERT, will be posted to the application. This event is found in *devmgmt.h*. The application should poll for this event and when an assert occurs, close all devices on this board before doing a single board stop/start. Closing should include **brd_Close( )** as well as the specific technology close such as **dx_close( )** or **dt_close( )**.

The firmware assert notification is available for all Dialogic® Springware JCT Boards.

Example Code

```
#include <windows.h> /* For Windows applications only */
#include "srllib.h"
#include "dxxxlib.h"
#include "devmgmt.h"

void main( )
{

      int nDev, nDev1, nRet, nEvtType;

      nDev = dx_open("dxxxB1C1", 0);

      if (nDev == -1)
      {
            printf("open failed err = %d %s\n", ATDV_LASTERR(nDev), ATDV_ERRMSGP(nDev));
            exit(0);
      }

      printf("opened %d\n", nDev);

      nDev1 = brd_Open("brdB1", BRD_FW_ASSERT_ENABLE); // enable fw assert notification

      if (nDev1 == -1)
      {

            printf("Brd open failed err = %d %s\n", ATDV_LASTERR(nDev1), ATDV_ERRMSGP(nDev1));
            exit(0);

      }

      nRet = sr_waitevt(10000);

      printf("waitevt returned\n");

      if(nRet == -1)
      {
            // ERROR
            printf("srl timeout error\n");
      }

      nEvtType = sr_getevttype(0);
```

```
        printf("event %x\n", nEvtType);

        printf("data: %s\n", sr_getevtdatap());

        dx_close(nDev);
        brd_Close(nDev1);
    }
```

1.50.4 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about **NCM_StopBoard( )**, **NCM_StartBoard( )**, and other NCM API functions, see the following documents:

- *Dialogic® Native Configuration Manager API Programming Guide*
- *Dialogic® Native Configuration Manager API Library Reference*

For information about the **brd_Open( )**, **brd_Close( )**, and other board management functions, see the following document:

- *Dialogic® Board Management API Library Reference*

1.51 SIP Call Transfer

With the Service Update, SIP call transfer on Dialogic® DM/IP Boards is now supported.

The six Global Call API functions that support IP call transfer are documented in the *Global Call API Library Reference*; protocol-specific information about the individual call transfer APIs is presented in the subsections of Section 1.51.4, "SIP Variances for Call Transfer Functions", on page 209.

1.51.1 Call Transfer Scenarios When Using SIP

The following topics describe the call transfer capabilities provided when using the SIP call transfer supplementary service:

- General Conditions for SIP Call Transfers
- Endpoint Behavior in Unattended SIP Call Transfers
- Successful Unattended SIP Call Transfer Scenarios
- Endpoint Behavior in Attended SIP Transfers
- Successful SIP Attended Call Transfer Scenarios
- Unsuccessful Call Transfer Scenarios

1.51.1.1 General Conditions for SIP Call Transfers

SIP call transfer uses the REFER method (with NOTIFY support) to reroute a call (a SIP dialog) after the call has been established; in other words, after two endpoints have an established media path.

There are two fundamental types of call transfer:

- Unattended transfer, which is referred to as “blind transfer” in most other technologies and protocols. In this type of transfer the transferring party (called the Transferor in SIP) has a call (or SIP dialog) with the transferred party (called the Transferee in SIP) but not with the transferred-to party (called the Transfer Target in SIP).
- Attended transfer, which is referred to as “supervised transfer” in most other technologies and protocols. In this type of transfer, the Transferor has a dialog with both the Transferee and the Transfer Target.

In its simplest terms, a SIP call transfer involves the Transferor issuing a REFER to the Transferee to cause the Transferee to issue an INVITE to the Transfer Target. The Transferee and Transfer Target negotiate the media without regard to the media that had been negotiated between the Transferor and the Transferee, just as if the Transferee had initiated the INVITE on its own.

Once a transfer request is accepted by the Transferee, the Transferor is not allowed to send another transfer request to the Transferee. Only if a transfer request is rejected or fails is the Transferor allowed to attempt another transfer request to Transferee.

The disposition of the media streams between the Transferor and the Transferee is not altered by the REFER method. A successful REFER transaction does not terminate the session between the Transferor and the Transferee; if those parties wish to terminate their session, they must do so with a subsequent BYE request.

In the SIP call transfer protocol the Transferor is notified when the Transferee accepts the REFER transfer request. The Dialogic® Global Call Library allows this notification to be signaled to the application as a GCEV_INVOKE_XFER_ACCEPTED event. This event is optional, and is disabled (or masked) by default. The party A application can enable and disable this event at any time after the line device is opened using the **gc_SetConfigData( )** function. See Section 1.51.3.1, “Enabling GCEV_INVOKE_XFER_ACCEPTED Events”, on page 204, for more information.

When performing a call transfer operation, all involved call handles must be on the same stack instance. This imposes the following application restrictions for call transfer operations:

- When performing an attended call transfer at party A, both the consultation line device and the transferring line device must be on the same virtual board.
- When performing a call transfer (either attended or unattended) at party B, both the transferring line device and the transferred line device must be on the same virtual board.
- When performing an attended call transfer at party C, both the consultation line device and the transferred-to line device must be on the same virtual board.

Interoperability Issues

The latest standards for the SIP REFER method are defined in IETF RFC 3515, published in April 2003. The current Global Call implementation is compliant with RFC 3515, but many existing implementations of REFER are based on the previous draft of the REFER method and are not fully compliant. The most significant non-compliance issues are:

- No initial NOTIFY after sending out 202 accept to REFER request.
- No subscription state information in NOTIFY message.
- No NOTIFY generated by the Transferee (Transferred party) after the call is terminated.
- Any NOTIFY received by the Transferor (Transferring party) after the subscription is terminated or the call is terminated will be rejected. Note that the subscription can be terminated implicitly after receiving NOTIFY of 180 Ringing.

1.51.1.2 Endpoint Behavior in Unattended SIP Call Transfers

The precondition for unattended call transfer (commonly referred to as “blind call transfer” in other technologies and protocols) is that the transferring endpoint (party A, or Transferor in SIP terminology) and the transferred endpoint (party B or Transferee in SIP terms) are participating in an active call, known as the primary call. From the perspective of the Global Call API, both parties are in the GCST_CONNNECTED state. Completion of a successful unattended transfer results in the eventual termination of the primary call, and the creation of the transferred call between party B and the Transfer Target (party C).

Transferor or Transferring Endpoint (Party A)

The Transferor (party A) initiates an unattended transfer by calling the **gc_InvokeXfer( )** function on the CRN of the primary call (CRNp), which results in the sending a REFER message to the Transferee (party B). The Refer-To header in the REFER request is constructed from either the char *numberstr or the GC_MAKECALL_BLK *makecallp parameter in the **gc_InvokeXfer( )** function, following the same rules as **gc_MakeCall( )**. The Referred-By header is automatically constructed with the local URI—the same as the From or To header, depending on the direction of the initial call INVITE. Optionally, the Transferor can override the default Referred-By header by inserting a Referred-By header in the **gc_InvokeXfer( )** parm block. Party A will be notified if REFER is accepted or rejected by transferred endpoint (party B).

If party A receives a 2xx response to the REFER (indicating that is was accepted by party B), a GCEV_INVOKE_XFER_ACCEPTED event may optionally be generated. This optional event is disabled by default; after the line device has been opened, the event can be enabled or disabled at any time by use of the **gc_SetConfigData( )** function.

The primary call may be terminated by either party before transferred call is completed. Note that in an H.450.2 implementation, party A will actually get INVOKE_XFER_REJ event locally if party A terminates the primary call before receiving final status from party B. Unlike an H.450.2 transfer, party A in a SIP transfer will **not** get any transfer termination event if party A terminates the primary call before receiving final status from party B. This is because there is no way to be sure if the transfer is successful or if it failed and it is party A's responsibility to update the application transfer states in this case. This is a common scenario in blind transfer where party A does not care about the transferred call status and drops the primary call immediately after receiving a INVOKE_XFER_ACCEPTED event.

When the REFER subscription is terminated, party A rejects subsequent NOTIFY messages. Any of the following events terminate the REFER subscription:

- a NOTIFY with subscription state terminated is received
- a NOTIFY of 180 Ringing is received
- a 2xx-6xx final response is received
- the primary call is terminated

If the primary call remains connected and the REFER subscription is alive, party A **may** be notified of the final status of transferred call from party B. The notification of transferred call status is optional depending on party B.

From party A's perspective, a call transfer is considered successful as long as GCEV_INVOKE_XFER_ACCEPTED (if enabled) and GCEV_INVOKE_XFER events are received. If the optional GCEV_INVOKE_XFER_ACCEPTED event type is enabled, that event is generated by receiving a 2xx response (to the REFER request) from party B. The GCEV_INVOKE_XFER event is generated by receiving from party B either a NOTIFY of termination of the REFER subscription or a NOTIFY of 180 Ringing or 2xx final status on the transferred call.

The REFER subscription will be terminated and the primary call will also be disconnected locally immediately after generating a GCEV_INVOKE_XFER event. From the Global Call API perspective, the primary call is terminated at the transferring endpoint as indicated by the GCEV_DISCONNECTED event implying the Transferor endpoint is then responsible for dropping and releasing the primary call.

Transferee or Transferred Endpoint (Party B)

The endpoint to be transferred (party B, or Transferee in SIP terms) is notified of the request to transfer from the initiating endpoint via a GCEV_REQ_XFER event on CRNp. If party B accepts the transfer request via **gc_AcceptXfer( )** function call on CRNp, a 202 Accepted response is sent to party A. Sending 202 Accepted to party A starts the REFER subscription, whereupon party B automatically sends a NOTIFY of 100 Trying (with default expiration time of 300 seconds) to party A on CRNp. No further notification of 100 Trying is sent from party B to party A during the call transfer process.

Party B retrieves the destination address information from the unsolicited transfer request via the GC_REROUTING_INFO structure passed with the GCEV_REQ_XFER event. Party B uses the rerouting address information (Refer-To address) to initiate a call to the new destination party via **gc_MakeCall( )** on CRNt. From the perspective of the application, this transferred call is treated in the same manner as a normal singular call and the party receives intermediate call state events as to the progress of the call (e.g., GCEV_DIALING, GCEV_ALERTING, GCEV_PROCEEDING, and GCEV_CONNECTED).

If the CRNp number is included during the **gc_MakeCall( )** on CRNt and the primary call is in the connected state, then a GCEV_XFER_CMPLT event is generated on CRNp once the transferred call is connected. If the CRNp number is not included, there will be no notification to the primary call and/or party A of the transferred call status. The CRNp number must not be included in the **gc_MakeCall()** if primary call was disconnected prior to making transferred call.

When party B receives any provisional response except 100 Trying from Party C and if the REFER subscription is still alive, party B automatically sends NOTIFY to party A with such transferred call status.

When party B receives the indication from party C that the call transfer was successful (200 OK), the party B application is notified of the success via a GCEV_XFER_CMPLT event on CRNp. If the primary call is still connected, party B will notify party A of the transfer status (200 OK) and terminate the REFER subscription. Then party B implicitly, without user/application initiation, disconnects the primary call with the party A. Although the primary call to party A is implicitly dropped, the call itself must still be explicitly dropped via **gc_DropCall( )** and released via **gc_ReleaseCallEx( )** to resynchronize the local state machine.

Either the party A or party B application may terminate the primary call after party B accepts the transfer request. If the primary call is terminated by party A before receiving any call transfer termination event (GCEV_INVOKE_XFER or GCEV_INVOKE_XFER_FAIL), party B will not notify party A of the transfer status. If the primary call is terminated by party B before receiving any transferred call provisional or final response from party C, party B *will* send NOTIFY to party A with 200 OK and terminate the REFER subscription before sending BYE to party A.

If the primary call is disconnected before making the transferred call to party C, party B must not include the primary call CRN (CRNp) when making the transferred call to party C. Otherwise, a Global Call error will be returned.

Note that the primary call can be disconnected prior to making the transferred call only during an unattended transfer because the transferred call can be established independently from the primary call. During an attended transfer, the transferred call cannot be established after the primary call is disconnected because the primary call database contains the Replaces information that is required by the transferred call.

If the Referred-By header exists in the REFER message, it is passed to the application via the GCEV_REQ_XFER event if SIP message information access was enabled (by setting the IP_SIP_MSGINFO_ENABLE in the sip_msginfo_mask field of the IP_VIRTBOARD data structure) when the virtual board was started.

Transfer Target or Transferred-To Endpoint (Party C)

From the perspective of party C, the transferred call is, for the most part, treated as a typical incoming call. The call is first notified to the application by a GCEV_DETECTED or GCEV_OFFERED event on CRNt. The GCRV_XFERCALL cause value is provided in the event to alert the application that this call offering is the result of a transfer, but only if the incoming INVITE contains Referred-By or Replaces information indicating a new transferred call. Referred-By and Replaces information, if present, is also attached to GCEV_OFFERED events if SIP header access was enabled (by setting the IP_SIP_MSGINFO_ENABLE value in the sip_msginfo_mask field of the IP_VIRTBOARD data structure) when the virtual board was started.

At that point, the application may retrieve the typical calling party information on CRNt. Party C is then provided the same methods of action as a typical incoming call, namely to alert party B that the call is proceeding (typically for gateways), ringback notification that the local user is being alerted, or simply that the call is answered. The only behavior change from this endpoint over typical non-transferred calls is whether to handle the calling party information any differently because it is the result of a transfer.

1.51.1.3 Successful Unattended SIP Call Transfer Scenarios

This section describes various scenarios for successful call transfers under the SIP protocol. The scenarios include:

- Successful Transfer with Notification of Connection
- Successful Transfer with Notification of Ringing
- Successful Transfer with Early Termination of REFER Subscription
- Successful Transfer with Primary Call Cleared prior to Transfer Completion

All of the scenarios indicate all three common naming conventions for the three parties involved in a call transfer: parties (A, B, and C), endpoints (transferring, transferred, and transferred-to), and SIP roles (Transferor, Transferee, and Transfer Target). “IP CClib” refers to the call control library and SIP stack portions of Dialogic® Global Call Software. “Non-Global Call” is used to represent a User Agent that might behave legally but differently than Global Call. Pre and post conditions are explicitly listed in each scenario, but the common pre-condition for all scenarios is that the Transferor (party A) and the Transferee (party B) are participating in an active (primary) call and are in the GCST_CONNNECTED state from the perspective of the Global Call API.

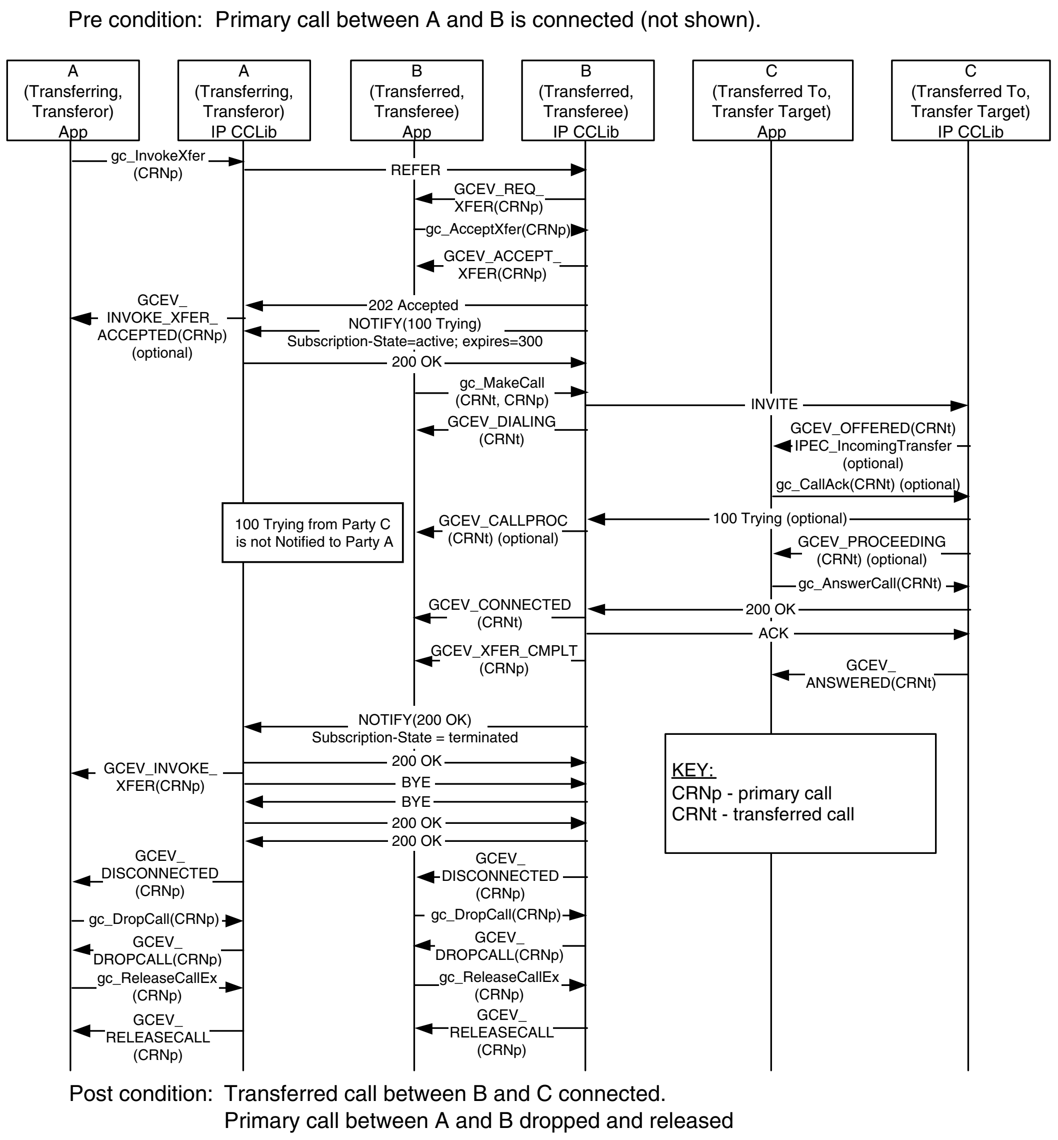
For simplification purposes, none of the figures indicate the opening and closing of logical channels (and the associated media sessions) because the control procedures are consistent with typical non-transfer related SIP calls.

All of the following scenarios illustrate the optional GCEV_INVOKE_XFER_ACCEPTED event, which is disabled by default. The party A application can enable and disable this event at any time after the line device is opened using the **gc_SetConfigData( )** function.

Successful Transfer with Notification of Connection

Figure 1 illustrates the basic successful scenario, with party A receiving notification from party B after the transferred call between party B and party C has been connected. The SIP dialog for the primary call between party A and party B is automatically disconnected, and both parties then tear down the call using **gc_DropCall( )** and **gc_ReleaseCallEx( )**.

Figure 1. Successful SIP Unattended Call Transfer, Party A Notified with Connection

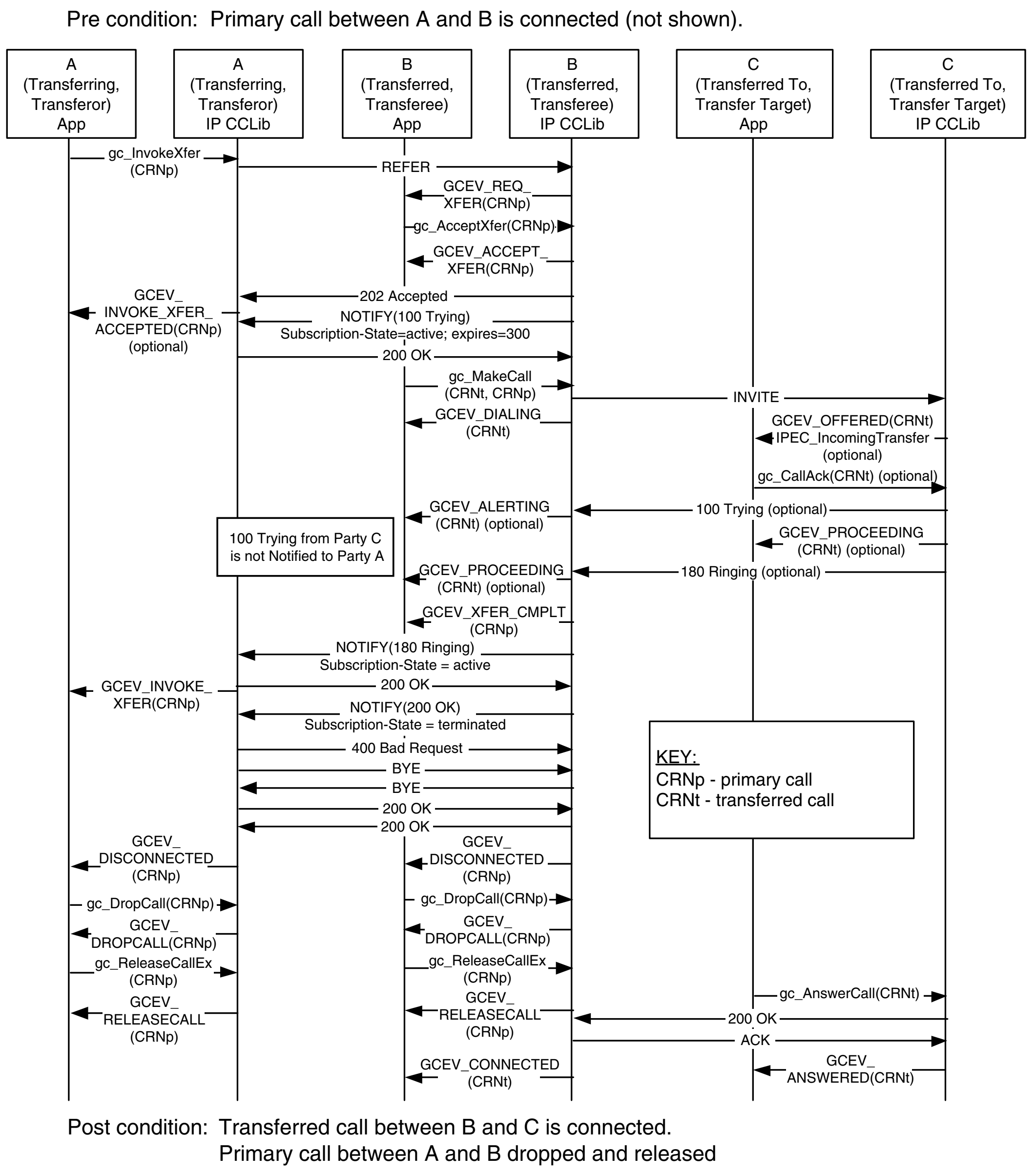


Pre condition: Primary call between A and B is connected (not shown).
A (Transferring, Transferor) App
A (Transferring, Transferor) IP CCLib
B (Transferred, Transferee) App
B (Transferred, Transferee) IP CCLib
C (Transferred To, Transfer Target) App
C (Transferred To, Transfer Target) IP CCLib
gc_InvokeXfer (CRNp)
REFER
GCEV_REQ_XFER(CRNp)
gc_AcceptXfer(CRNp)
GCEV_ACCEPT_XFER(CRNp)
GCEV_INVOKE_XFER_ACCEPTED(CRNp) (optional)
202 Accepted
NOTIFY(100 Trying) Subscription-State=active; expires=300
200 OK
gc_MakeCall (CRNt, CRNp)
INVITE
GCEV_DIALING (CRNt)
GCEV_OFFERED(CRNt)
IPEC_IncomingTransfer (optional)
gc_CallAck(CRNt) (optional)
100 Trying from Party C is not Notified to Party A
GCEV_CALLPROC (CRNt) (optional)
100 Trying (optional)
GCEV_PROCEEDING (CRNt) (optional)
gc_AnswerCall(CRNt)
GCEV_CONNECTED (CRNt)
200 OK
ACK
GCEV_XFER_CMPLT (CRNp)
GCEV_ANSWERED(CRNt)
NOTIFY(200 OK) Subscription-State = terminated
GCEV_INVOKE_XFER(CRNp)
200 OK
BYE
BYE
200 OK
200 OK
KEY:
CRNp - primary call
CRNt - transferred call
GCEV_DISCONNECTED (CRNp)
GCEV_DISCONNECTED (CRNp)
gc_DropCall(CRNp)
gc_DropCall(CRNp)
GCEV_DROPCALL(CRNp)
GCEV_DROPCALL(CRNp)
gc_ReleaseCallEx (CRNp)
gc_ReleaseCallEx (CRNp)
GCEV_RELEASECALL (CRNp)
GCEV_RELEASECALL (CRNp)
Post condition: Transferred call between B and C connected.
Primary call between A and B dropped and released

Successful Transfer with Notification of Ringing

Figure 2 illustrates a scenario where party B notifies party A that the transfer has completed as soon as party C responds to the INVITE with a 100 Trying or 180 Ringing. The Call Control Library at Party A disconnects the primary call with party B after the notification and the application then must tear down the call using **gc_DropCall( )** and **gc_ReleaseCallEx( )**.

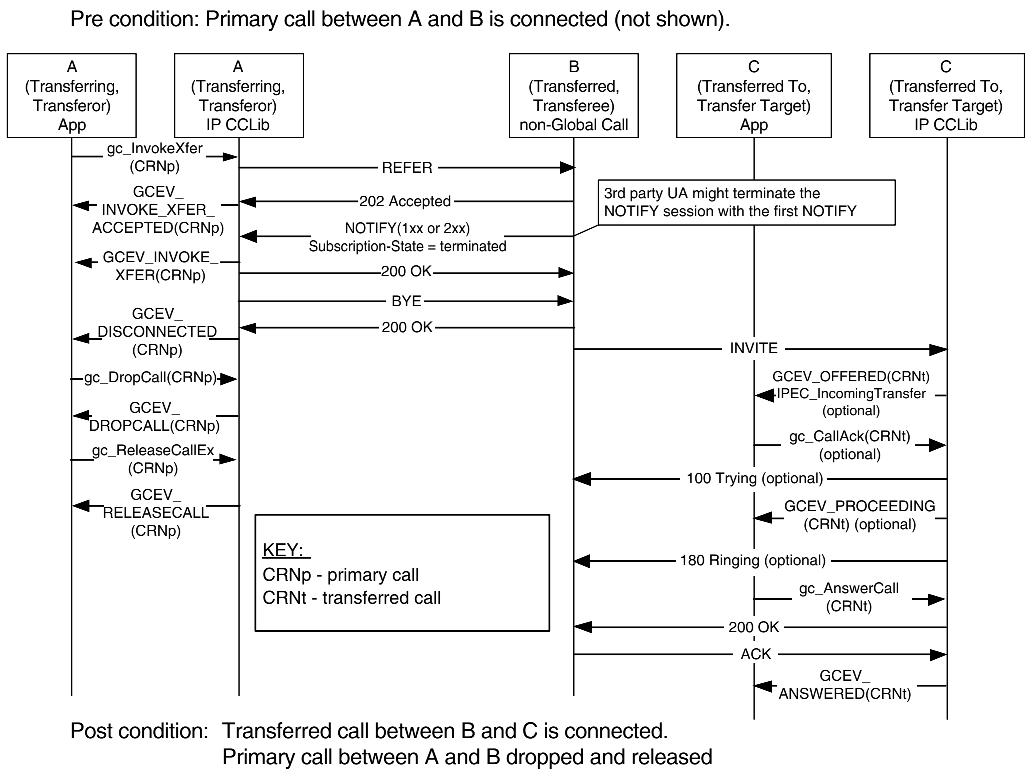
Figure 2. Successful SIP Unattended Call Transfer, Party A Notified with Ringing


Pre condition: Primary call between A and B is connected (not shown).
A (Transferring, Transferor) App
A (Transferring, Transferor) IP CCLib
B (Transferred, Transferee) App
B (Transferred, Transferee) IP CCLib
C (Transferred To, Transfer Target) App
C (Transferred To, Transfer Target) IP CCLib
gc_InvokeXfer (CRNp)
REFER
GCEV_REQ_XFER(CRNp)
gc_AcceptXfer(CRNp)
GCEV_ACCEPT_XFER(CRNp)
202 Accepted
GCEV_INVOKE_XFER_ACCEPTED(CRNp) (optional)
NOTIFY(100 Trying) Subscription-State=active; expires=300
200 OK
gc_MakeCall (CRNt, CRNp)
INVITE
GCEV_DIALING (CRNt)
GCEV_OFFERED(CRNt) IPEC_IncomingTransfer (optional)
gc_CallAck(CRNt) (optional)
GCEV_ALERTING (CRNt) (optional)
100 Trying (optional)
100 Trying from Party C is not Notified to Party A
GCEV_PROCEEDING (CRNt) (optional)
GCEV_PROCEEDING (CRNt) (optional)
180 Ringing (optional)
GCEV_XFER_CMPLT (CRNp)
NOTIFY(180 Ringing) Subscription-State = active
200 OK
GCEV_INVOKE_XFER(CRNp)
NOTIFY(200 OK) Subscription-State = terminated
400 Bad Request
BYE
BYE
200 OK
200 OK
KEY:
CRNp - primary call
CRNt - transferred call
GCEV_DISCONNECTED (CRNp)
GCEV_DISCONNECTED (CRNp)
gc_DropCall(CRNp)
gc_DropCall(CRNp)
GCEV_DROPCALL(CRNp)
GCEV_DROPCALL(CRNp)
gc_ReleaseCallEx (CRNp)
gc_ReleaseCallEx (CRNp)
GCEV_RELEASECALL (CRNp)
GCEV_RELEASECALL (CRNp)
gc_AnswerCall(CRNt)
200 OK
ACK
GCEV_CONNECTED (CRNt)
GCEV_ANSWERED(CRNt)
Post condition: Transferred call between B and C is connected. Primary call between A and B dropped and released

Successful Transfer with Early Termination of REFER Subscription

Figure 3 illustrates a valid scenario for which Global Call does not support the party B role. In this scenario, party B terminates the REFER subscription with the first NOTIFY, before party A can be notified of the transferred call status. The Call Control Library at Party A disconnects the primary call with party B after the terminating NOTIFY and the application then must tear down the call using **gc_DropCall( )** and **gc_ReleaseCallEx( )**.

Figure 3. Successful SIP Unattended Call Transfer, Party B Terminates REFER Subscription prior to Notification of Transferred Call Status


Pre condition: Primary call between A and B is connected (not shown).
A (Transferring, Transferor) App
A (Transferring, Transferor) IP CCLib
B (Transferred, Transferee) non-Global Call
C (Transferred To, Transfer Target) App
C (Transferred To, Transfer Target) IP CCLib
gc_InvokeXfer (CRNp)
REFER
GCEV_ INVOKE_XFER_ ACCEPTED(CRNp)
202 Accepted
3rd party UA might terminate the NOTIFY session with the first NOTIFY
NOTIFY(1xx or 2xx) Subscription-State = terminated
GCEV_INVOKE_ XFER(CRNp)
200 OK
BYE
GCEV_ DISCONNECTED (CRNp)
200 OK
INVITE
gc_DropCall(CRNp)
GCEV_OFFERED(CRNt) IPEC_IncomingTransfer (optional)
GCEV_ DROPCALL(CRNp)
gc_CallAck(CRNt) (optional)
gc_ReleaseCallEx (CRNp)
100 Trying (optional)
GCEV_ RELEASECALL (CRNp)
GCEV_PROCEEDING (CRNt) (optional)
KEY:
CRNp - primary call
CRNt - transferred call
180 Ringing (optional)
gc_AnswerCall (CRNt)
200 OK
ACK
GCEV_ ANSWERED(CRNt)
Post condition: Transferred call between B and C is connected. Primary call between A and B dropped and released

Successful Transfer with Primary Call Cleared prior to Transfer Completion

The SIP protocol supports unattended transfer scenarios where the primary call is cleared or dropped before the transfer completes. In some other technologies and protocols, these scenarios are referred to as “unattended blind transfers” as opposed to “attended blind transfers” where the primary call is maintained until completion. Note that scenarios similar to these are not supported by the H.450.2 protocol.

Figure 4 illustrates a scenario in which party A drops the primary call with party B as soon as it receives notification that party B has accepted the transfer request. In this scenario, party A does not receive any notification that the transfer has completed.

Figure 4. Successful SIP Unattended Call Transfer, Party A Clears Primary Call prior to Transfer Completion

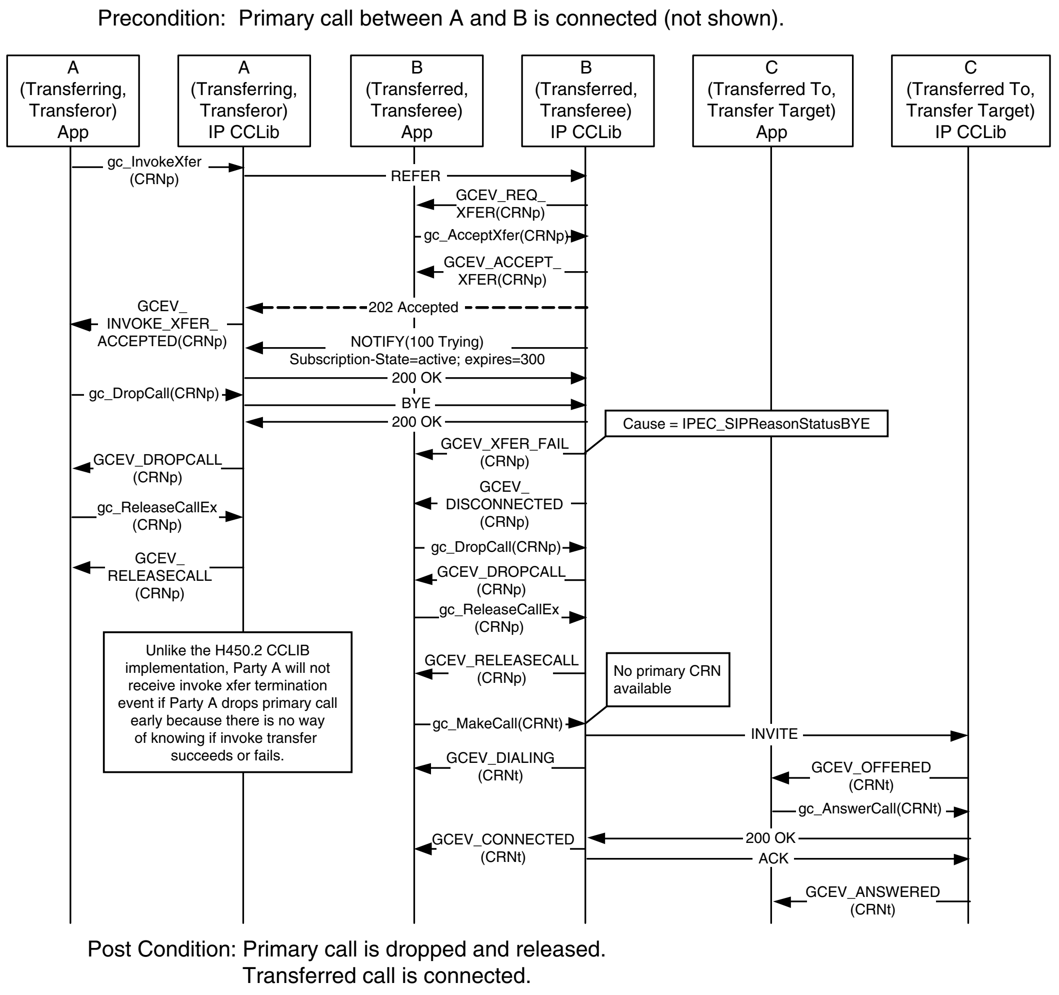
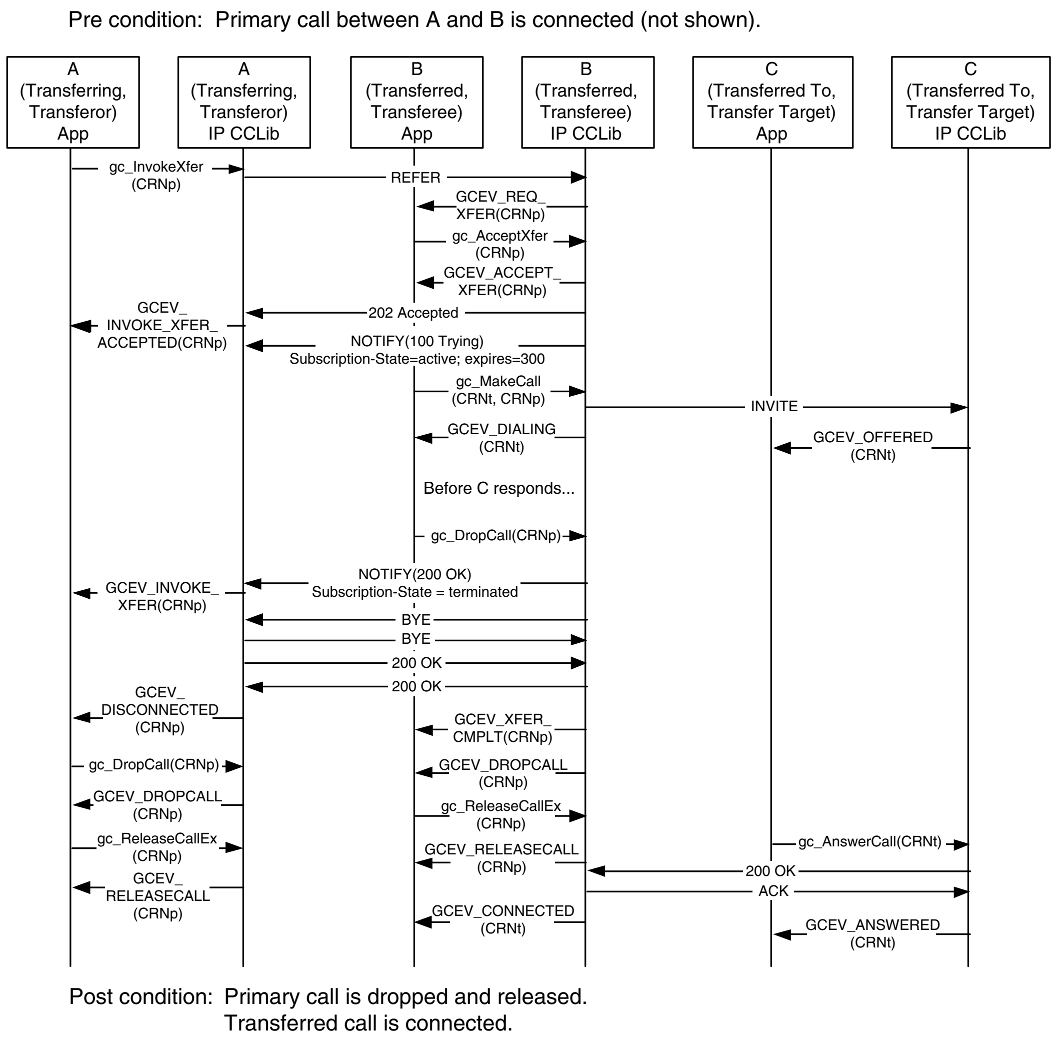

Precondition: Primary call between A and B is connected (not shown).
A (Transferring, Transferor) App
A (Transferring, Transferor) IP CCLib
B (Transferred, Transferee) App
B (Transferred, Transferee) IP CCLib
C (Transferred To, Transfer Target) App
C (Transferred To, Transfer Target) IP CCLib
gc_InvokeXfer (CRNp)
REFER
GCEV_REQ_ XFER(CRNp)
gc_AcceptXfer(CRNp)
GCEV_ACCEPT_ XFER(CRNp)
202 Accepted
GCEV_ INVOKE_XFER_ ACCEPTED(CRNp)
NOTIFY(100 Trying) Subscription-State=active; expires=300
200 OK
gc_DropCall(CRNp)
BYE
200 OK
Cause = IPEC_SIPReasonStatusBYE
GCEV_XFER_FAIL (CRNp)
GCEV_DROPCALL (CRNp)
GCEV_ DISCONNECTED (CRNp)
gc_ReleaseCallEx (CRNp)
gc_DropCall(CRNp)
GCEV_DROPCALL (CRNp)
GCEV_ RELEASECALL (CRNp)
gc_ReleaseCallEx (CRNp)
Unlike the H450.2 CCLIB implementation, Party A will not receive invoke xfer termination event if Party A drops primary call early because there is no way of knowing if invoke transfer succeeds or fails.
GCEV_RELEASECALL (CRNp)
No primary CRN available
gc_MakeCall(CRNt)
INVITE
GCEV_DIALING (CRNt)
GCEV_OFFERED (CRNt)
gc_AnswerCall(CRNt)
200 OK
GCEV_CONNECTED (CRNt)
ACK
GCEV_ANSWERED (CRNt)
Post Condition: Primary call is dropped and released. Transferred call is connected.

Figure 5 illustrates a scenario in which party B drops the primary call with party A after accepting the transfer request and issuing INVITE to party C, but before receiving any response from party C. In this scenario, party B does notify party A, but this notification only signifies that party B has acted on the transfer request and not that the transfer has actually completed.

Figure 5. Successful SIP Unattended Call Transfer, Party B Clears Primary Call prior to Transfer Completion



Pre condition: Primary call between A and B is connected (not shown).
A (Transferring, Transferor) App
A (Transferring, Transferor) IP CCLib
B (Transferred, Transferee) App
B (Transferred, Transferee) IP CCLib
C (Transferred To, Transfer Target) App
C (Transferred To, Transfer Target) IP CCLib
gc_InvokeXfer (CRNp)
REFER
GCEV_REQ_XFER(CRNp)
gc_AcceptXfer (CRNp)
GCEV_ACCEPT_XFER(CRNp)
202 Accepted
GCEV_INVOKE_XFER_ACCEPTED(CRNp)
NOTIFY(100 Trying) Subscription-State=active; expires=300
gc_MakeCall (CRNt, CRNp)
INVITE
GCEV_DIALING (CRNt)
GCEV_OFFERED (CRNt)
Before C responds...
gc_DropCall(CRNp)
NOTIFY(200 OK) Subscription-State = terminated
GCEV_INVOKE_XFER(CRNp)
BYE
BYE
200 OK
200 OK
GCEV_DISCONNECTED (CRNp)
GCEV_XFER_CMPLT(CRNp)
gc_DropCall(CRNp)
GCEV_DROPCALL (CRNp)
GCEV_DROPCALL (CRNp)
gc_ReleaseCallEx (CRNp)
gc_ReleaseCallEx (CRNp)
gc_AnswerCall(CRNt)
GCEV_RELEASECALL (CRNp)
200 OK
ACK
GCEV_RELEASECALL (CRNp)
GCEV_CONNECTED (CRNt)
GCEV_ANSWERED (CRNt)
Post condition: Primary call is dropped and released. Transferred call is connected.

1.51.1.4 Endpoint Behavior in Attended SIP Transfers

The assumed preconditions for attended SIP call transfer (commonly referred to as “supervised call transfer” in other technologies and protocols) are:

- The transferring endpoint (party A, or Transferor in SIP terminology) and the transferred endpoint (party B, or Transferee in SIP terms) are participating in an active call, known as the primary call. From the perspective of the Global Call API, party A and party B are both in the GCST_CONNNECTED state.
- The Transferor and the transferred-to party (party C or the Transfer Target in SIP terminology) are participating in an active call, known as the secondary or consultation call. From the perspective of the Global Call call control library, party A and party C are both in the GCST_CONNNECTED state.

Completion of a successful attended transfer results in the eventual termination of the primary and secondary calls, and the creation of the transferred call between party B and the party C.

Transferor or Transferring Endpoint (Party A)

SIP does not support or require a transfer initiation process to obtain the rerouting number as in H.323/H.450.2 supervised transfer. To be consistent with the generic Global Call supervised transfer scenario, the party A application in a SIP attended transfer can call **gc_InitXfer( )**, but no request / response messages will be exchanged between party A and party C as a result. Following this function call, party A always receives a GCEV_INIT_XFER completion event with a dummy rerouting address. To alert party C of incoming transfer process, party A can only notify party C by application data or human interaction outside of SIP protocol.

Just as in the case of unattended transfers, an attended transfer is actually initiated when the Transferor calls the **gc_InvokeXfer( )** function. The difference between unattended and attended transfer usage is the inclusion of the CRN of the secondary (consultation) call as a parameter in the function call. When the Transferor calls **gc_InvokeXfer( )** with two CRN values, a REFER message with a replace parameter in the Refer-To header is sent to the Transferee (party B).

From this point onward, the behavior at this endpoint is similar to that of a unattended transfer, except that the application must also drop the secondary/consultation call at transfer completion. Unlike H.450.2, Global Call will not disconnect the secondary/consultation call once the transferred call is answered at party C.

Because SIP does not require any pre-invocation setup for attended call transfers, the Transferor (party A) can actually treat either of the two active calls as the primary call, and can send the REFER to either of the remote endpoints. This fact provides a recovery mechanism in case one of the remote endpoints does not support the REFER method, as illustrated in the scenarios in the following section.

Protecting and Exposing the Transfer Target

The ability to direct the REFER to either of the parties to which the Transferor provides the opportunity to protect the Transfer Target.

To protect the Transfer Target, the Transferor simply reverses the primary and secondary call CRNs when calling **gc_InvokeXfer( )** to reverse the roles of the two remote parties. The original Transfer Target will now send INVITE to the original Transferee, so that the Transferee is effectively “called back” by the Transfer Target. This has the advantage of hiding information about the original Transfer Target from the original transferee, although the Transferee’s experience in this scenario will be different that in current systems PBX or Centrex systems.

To expose the Transfer Target and provide an experience similar to current PBX and Centrex systems, the Transferor uses the secondary call to alert the Transfer Target to the impending transfer, but then disconnects the secondary call and completes the transfer as an unattended transfer. In this case, the **gc_InvokeXfer( )** call only includes the CRN of the primary call.

Transferee or Transferred Endpoint (Party B)

This endpoint behaves in the same manner as in unattended transfer with one exception: the INVITE that is sent from Party B to Party C for the transferred call contains a Replaces header that is obtained from the replace parameter in the Refer-To header of the REFER from Party A.

Note that the primary call cannot be disconnected prior to making the transferred call during an attended transfer because the primary call database contains the Replaces information that is required to establish the transferred call.

Transfer Target or Transferred-To Endpoint (Party C)

This endpoint behaves in much the same manner as in an unattended transfer with one additional feature and one additional responsibility.

If the Replaces header exists in the incoming INVITE, Global Call automatically matches the Replaces value with any existing connected call on Party C. If a matching call (the secondary or consultation call) is found, that call's CRNs is passed to the application as a GCPARM_SECONDARYCALL_CRN parameter in the GC_PARM_BLK that is attached to the GCEV_OFFERED event.

The party C application must also drop the secondary/consultation call when the transfer completes. Unlike H.450.2 call transfer, Global Call does not automatically disconnect the secondary call once the transferred call answered at the party C.

1.51.1.5 Successful SIP Attended Call Transfer Scenarios

This section describes the basic scenario for successful SIP call transfer and the scenarios for recovery from two conditions that can block transfer completion. The scenarios include:

- Successful SIP Attended Call Transfer
- Attended Transfer When REFER Is Not Globally Supported
- Attended Transfer When Contact URI Is Not Globally Routable

The scenarios all illustrate the optional GCEV_INVOKE_XFER_ACCEPTED event, which is disabled by default. The Transferor application can enable and disable this event at any time after the line device is opened using the **gc_SetConfigData( )** function.

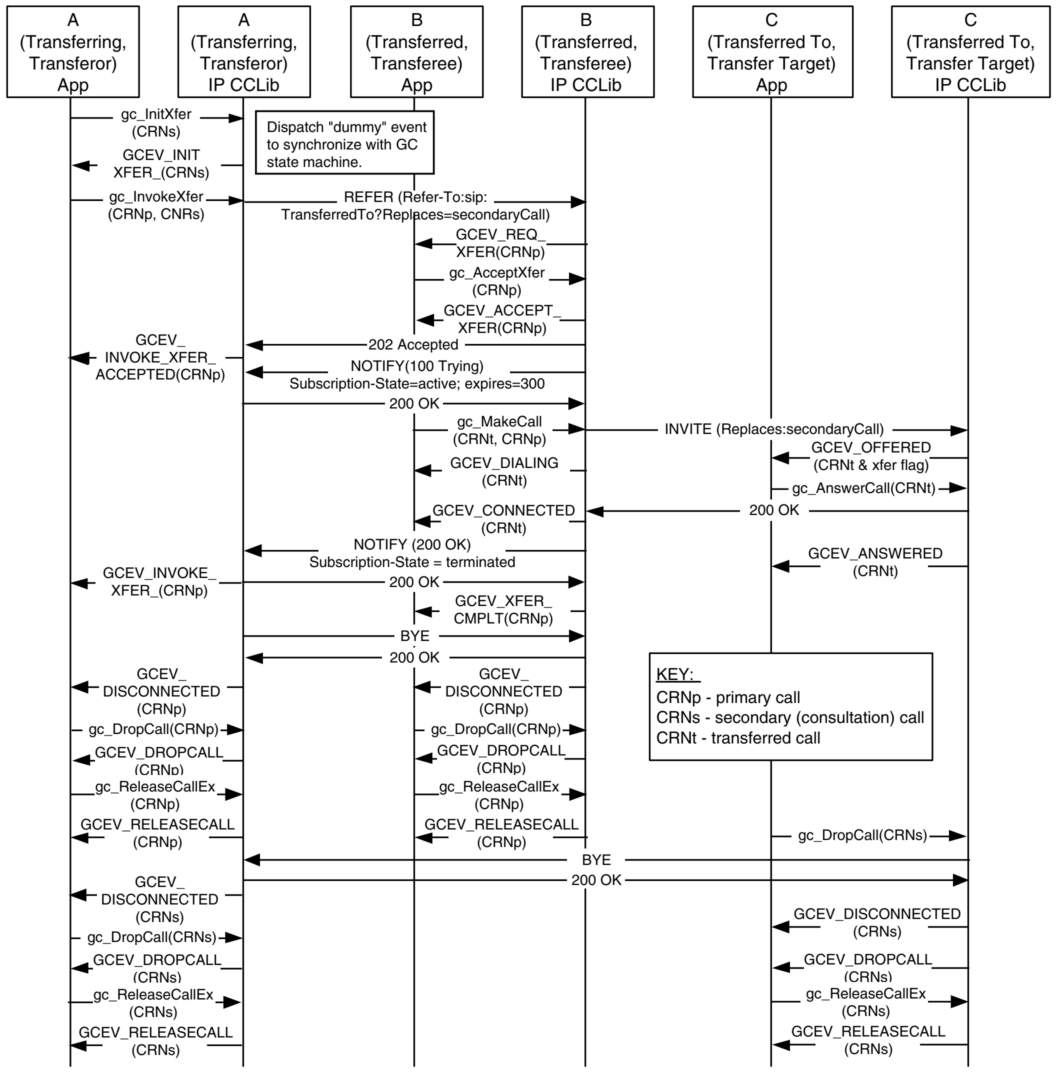
For simplification purposes, none of the figures indicate the opening and closing of logical channels (and the associated media sessions) because the control procedures are consistent with typical non-transfer related SIP calls.

Successful SIP Attended Call Transfer

Figure 6 illustrates the basic scenario for successful SIP attended call transfer. The scenario illustrates the use of a **gc_InitXfer( )** function call, which is not required in SIP. The GCEV_INIT_XFER completion event in this case contains a dummy rerouting address.

Figure 6. Successful SIP Attended Call Transfer

Pre condition: Primary call between A and B is connected (not shown).
Secondary (consultation) call between A and C is connected (not shown).

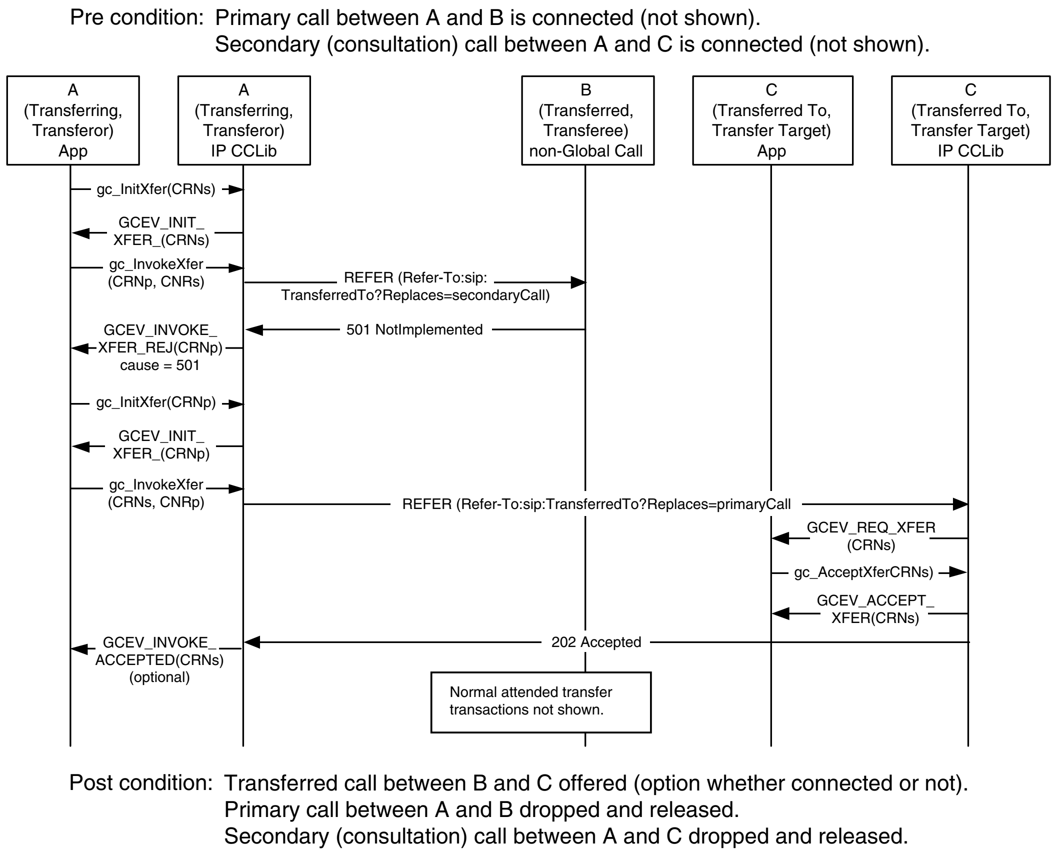

A (Transferring, Transferor) App
A (Transferring, Transferor) IP CCLib
B (Transferred, Transferee) App
B (Transferred, Transferee) IP CCLib
C (Transferred To, Transfer Target) App
C (Transferred To, Transfer Target) IP CCLib
gc_InitXfer (CRNs)
Dispatch "dummy" event to synchronize with GC state machine.
GCEV_INIT XFER_(CRNs)
gc_InvokeXfer (CRNp, CNRs)
REFER (Refer-To:sip: TransferredTo?Replaces=secondaryCall)
GCEV_REQ_ XFER(CRNp)
gc_AcceptXfer (CRNp)
GCEV_ACCEPT_ XFER(CRNp)
202 Accepted
GCEV_ INVOKE_XFER_ ACCEPTED(CRNp)
NOTIFY(100 Trying) Subscription-State=active; expires=300
200 OK
gc_MakeCall (CRNt, CRNp)
INVITE (Replaces:secondaryCall)
GCEV_OFFERED (CRNt & xfer flag)
GCEV_DIALING (CRNt)
gc_AnswerCall(CRNt)
GCEV_CONNECTED (CRNt)
200 OK
NOTIFY (200 OK) Subscription-State = terminated
GCEV_ANSWERED (CRNt)
GCEV_INVOKE_ XFER_(CRNp)
200 OK
GCEV_XFER_ CMPLT(CRNp)
BYE
200 OK
KEY:
CRNp - primary call
CRNs - secondary (consultation) call
CRNt - transferred call
GCEV_ DISCONNECTED (CRNp)
gc_DropCall(CRNp)
GCEV_DROPCALL (CRNp)
gc_ReleaseCallEx (CRNp)
GCEV_RELEASECALL (CRNp)
gc_DropCall(CRNs)
BYE
200 OK
GCEV_ DISCONNECTED (CRNs)
gc_DropCall(CRNs)
GCEV_DROPCALL (CRNs)
gc_ReleaseCallEx (CRNs)
GCEV_RELEASECALL (CRNs)

Post condition: Transferred call between B and C offered (option whether connected or not).
Primary call between A and B dropped and released.
Secondary (consultation) call between A and C dropped and released.

Attended Transfer When REFER Is Not Globally Supported

If protecting or exposing the Transfer Target is not a concern, it is possible to complete an attended transfer when only the Transferor and one other party support REFER. Note that a 405 Method Not Allowed might be returned instead of the 501 Not Implemented response.

Figure 7. SIP Attended Call Transfer, Recovery from REFER Unsupported



Pre condition: Primary call between A and B is connected (not shown).
Secondary (consultation) call between A and C is connected (not shown).
A (Transferring, Transferor) App
A (Transferring, Transferor) IP CCLib
B (Transferred, Transferee) non-Global Call
C (Transferred To, Transfer Target) App
C (Transferred To, Transfer Target) IP CCLib
gc_InitXfer(CRNs)
GCEV_INIT_ XFER_(CRNs)
gc_InvokeXfer (CRNp, CNRs)
REFER (Refer-To:sip: TransferredTo?Replaces=secondaryCall)
501 NotImplemented
GCEV_INVOKE_ XFER_REJ(CRNp) cause = 501
gc_InitXfer(CRNp)
GCEV_INIT_ XFER_(CRNp)
gc_InvokeXfer (CRNs, CNRp)
REFER (Refer-To:sip:TransferredTo?Replaces=primaryCall
GCEV_REQ_XFER (CRNs)
gc_AcceptXferCRNs)
GCEV_ACCEPT_ XFER(CRNs)
202 Accepted
GCEV_INVOKE_ ACCEPTED(CRNs) (optional)
Normal attended transfer transactions not shown.
Post condition: Transferred call between B and C offered (option whether connected or not).
Primary call between A and B dropped and released.
Secondary (consultation) call between A and C dropped and released.

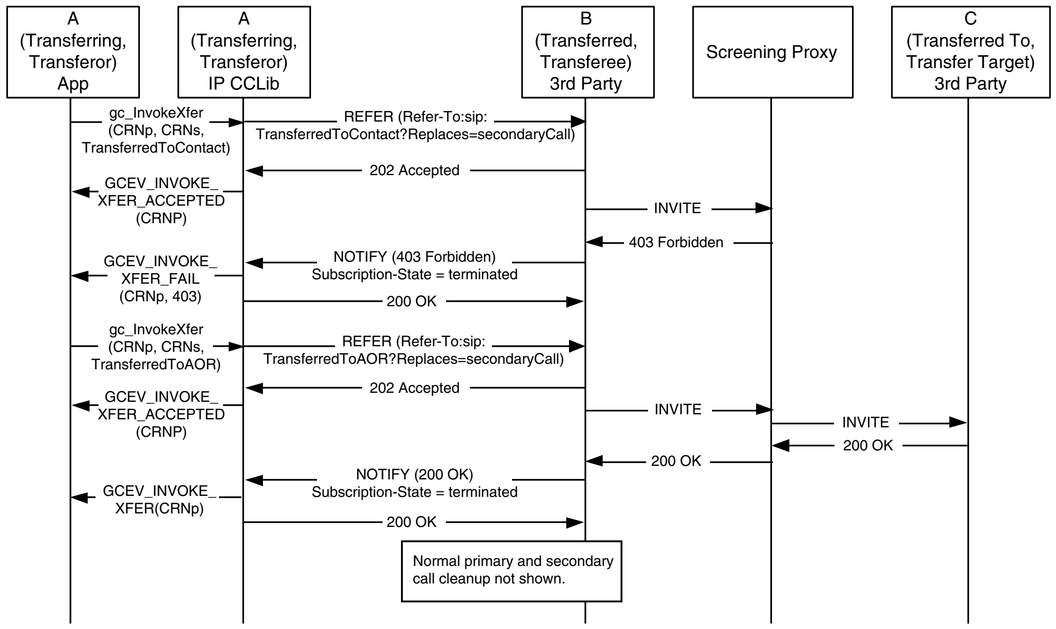
Attended Transfer When Contact URI Is Not Globally Routable

It is a requirement of RFC3261 that a Contact URI be globally routable even outside the dialog. However, due to RFC2543 User Agents and some architectures (NAT/firewall traversal, screening proxies, ALGs, etc.), this will not always be the case. As a result, the methods of attended transfer shown in Figure 6 and Figure 7 may fail since they use the Contact URI in the Refer-To header field. Figure 8 shows such a scenario involving a Screening Proxy in which the transfer initially fails but succeeds on a second try. The failure response (403 Forbidden, 404 Not Found, or a timeout after no response) is communicated back to the Transferor. Since this may be caused by routing problems with the Contact URI, the Transferor retries the REFER, this time with Refer-To containing the Address of Record (AOR) of the Target (the same URI the Transferor used to reach the Transfer Target). However, the use of the AOR URI may result in routing features being activated such as forking or sequential searching which may result in the triggered INVITE reaching the wrong User Agent. To prevent an incorrect UA answering the INVITE, a Require: replaces header field is included in the Refer-To. This ensures that only the UA which matches the Replaces dialog will answer the INVITE, since any incorrect UA which supports Replaces will reply with a 481 and a UA which does not support Replaces will reply with a 420.

Note that there is still no guarantee that the correct endpoint will be reached, and the result of this second REFER may also be a failure. In that case, the Transferor could fall back to unattended transfer or give up on the transfer entirely. Since two REFERs are sent within the dialog, creating two distinct subscriptions, the Transferee uses the ‘id’ parameter in the Event header field to distinguish notifications for the two subscriptions.

Figure 8. SIP Attended Call Transfer, Recovery from URI Not Routable

Pre condition: Primary call between A and B is connected (not shown).
Secondary (consultation) call between A and C is connected (not shown).


A (Transferring, Transferor) App
A (Transferring, Transferor) IP CCLib
B (Transferred, Transferee) 3rd Party
Screening Proxy
C (Transferred To, Transfer Target) 3rd Party
gc_InvokeXfer (CRNp, CRNs, TransferredToContact)
REFER (Refer-To:sip: TransferredToContact?Replaces=secondaryCall)
202 Accepted
GCEV_INVOKE_ XFER_ACCEPTED (CRNP)
INVITE
403 Forbidden
GCEV_INVOKE_ XFER_FAIL (CRNp, 403)
NOTIFY (403 Forbidden) Subscription-State = terminated
200 OK
gc_InvokeXfer (CRNp, CRNs, TransferredToAOR)
REFER (Refer-To:sip: TransferredToAOR?Replaces=secondaryCall)
202 Accepted
GCEV_INVOKE_ XFER_ACCEPTED (CRNP)
INVITE
INVITE
200 OK
200 OK
NOTIFY (200 OK) Subscription-State = terminated
GCEV_INVOKE_ XFER(CRNp)
200 OK
Normal primary and secondary call cleanup not shown.

Post condition: Transferred call between B and C is connected.
Primary and secondary calls are dropped and released.

1.51.1.6 Unsuccessful Call Transfer Scenarios

All of the scenarios in this section apply to both unattended (blind) transfer and attended (supervised) SIP call transfers. The **gc_InitXfer( )** function call and GCEV_INIT_XFER termination event are “dummy” operations that are only used to synchronize the Global Call state machine and can safely be ignored in this context.

Transfer failures can be caused by any of transfer endpoints as shown in scenarios. In all cases, the transferring endpoint (Transferor or party A) is notified by either

INVOKE_XFER_REJ or INVOKE_XFER_FAIL event with cause. No NOTIFY will be sent from party B to party A if REFER is not accepted by 202 Accepted from party B. The primary call and secondary call, if any, remain in connected state after any transfer failure.

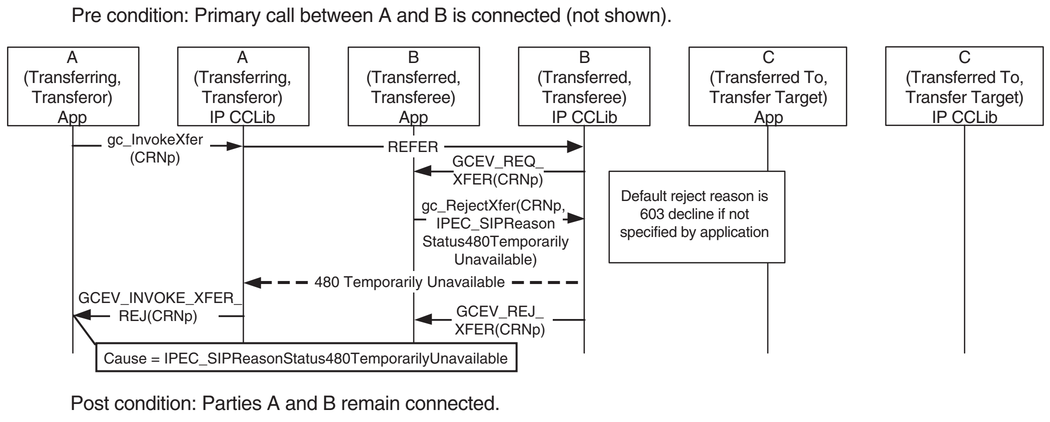
The most common transfer failure scenarios are described in the following topics:

- Party B Rejects Call Transfer
- No Response from Party B
- No Initial NOTIFY after REFER Accepted
- REFER Subscription Expires
- No Response from Party C
- Party B Drops Transferred Call Early
- Party C Is Busy When Transfer Attempted

Party B Rejects Call Transfer

Figure 9 illustrates a scenario in which the application at the transferred endpoint (Transferee or party B) calls **gc_RejectXfer( )** to signal the Transferor (party A) that it cannot participate in a transfer. The application may specify any valid SIP rejection reason, such as the 480 Temporarily Unavailable shown in the figure; if no reason is specified, the default reason sent is 603 Decline. As a result of the rejection, the GCEV_INVOKE_XFER_REJ termination event is received at the Transferor application (party A). The original primary call is left connected and in the GCST_CONNECTED state from the perspective of both party A and party B.

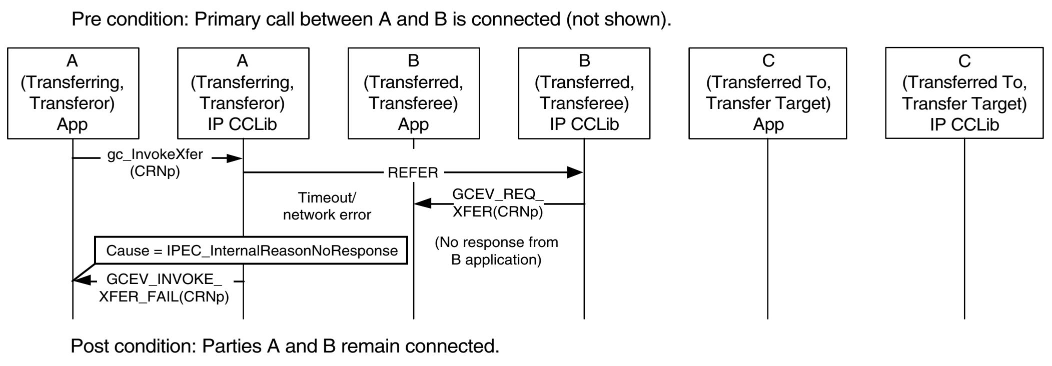
Figure 9. SIP Call Transfer Failure - Party B Rejects Call Transfer


Pre condition: Primary call between A and B is connected (not shown).
A (Transferring, Transferor) App
A (Transferring, Transferor) IP CCLib
B (Transferred, Transferee) App
B (Transferred, Transferee) IP CCLib
C (Transferred To, Transfer Target) App
C (Transferred To, Transfer Target) IP CCLib
gc_InvokeXfer (CRNp)
REFER
GCEV_REQ_XFER(CRNp)
gc_RejectXfer(CRNp, IPEC_SIPReason Status480Temporarily Unavailable)
Default reject reason is 603 decline if not specified by application
480 Temporarily Unavailable
GCEV_INVOKE_XFER_REJ(CRNp)
GCEV_REJ_XFER(CRNp)
Cause = IPEC_SIPReasonStatus480TemporarilyUnavailable
Post condition: Parties A and B remain connected.

No Response from Party B

Figure 10 illustrates a scenario in which the Transferee (party B) does not respond to the REFER, causing the T3 timer at the party A (configured as 20 seconds) to expire. After the timeout, the Transferor application receives the GCEV_INVOKE_XFER_FAIL termination event. The original primary call is left connected and in the GCST_CONNECTED state from the perspective of both party A and party B.

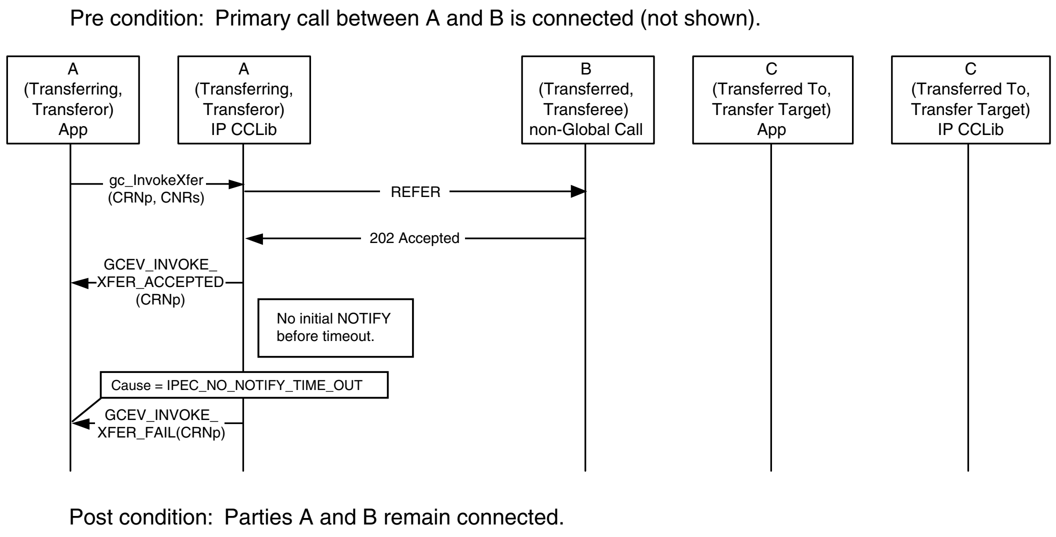
Figure 10. SIP Call Transfer Failure - No Response from Party B


Pre condition: Primary call between A and B is connected (not shown).
A (Transferring, Transferor) App
A (Transferring, Transferor) IP CCLib
B (Transferred, Transferee) App
B (Transferred, Transferee) IP CCLib
C (Transferred To, Transfer Target) App
C (Transferred To, Transfer Target) IP CCLib
gc_InvokeXfer (CRNp)
REFER
Timeout/ network error
GCEV_REQ_ XFER(CRNp)
(No response from B application)
Cause = IPEC_InternalReasonNoResponse
GCEV_INVOKE_ XFER_FAIL(CRNp)
Post condition: Parties A and B remain connected.

No Initial NOTIFY after REFER Accepted

Figure 11 illustrates a scenario in which the Transferee (party B) does not send a NOTIFY after it accepts the REFER, causing the timer at party A to expire. The original primary call is left connected and in the GCST_CONNECTED state from the perspective of both party A and party B.

Figure 11. SIP Call Transfer Failure - No Initial NOTIFY after REFER Is Accepted

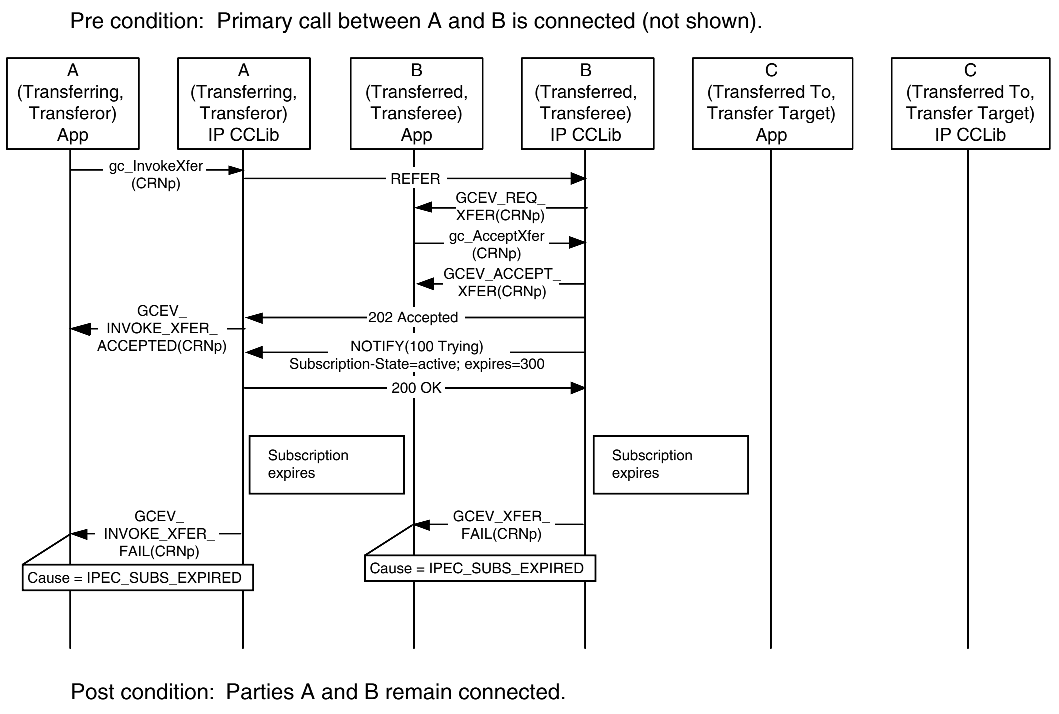

Pre condition: Primary call between A and B is connected (not shown).
A (Transferring, Transferor) App
A (Transferring, Transferor) IP CCLib
B (Transferred, Transferee) non-Global Call
C (Transferred To, Transfer Target) App
C (Transferred To, Transfer Target) IP CCLib
gc_InvokeXfer (CRNp, CNRs)
REFER
202 Accepted
GCEV_INVOKE_ XFER_ACCEPTED (CRNp)
No initial NOTIFY before timeout.
Cause = IPEC_NO_NOTIFY_TIME_OUT
GCEV_INVOKE_ XFER_FAIL(CRNp)
Post condition: Parties A and B remain connected.

REFER Subscription Expires

Figure 12 illustrates a scenario in which the REFER subscription expires, causing both party A and party B to time out. After the timeout, the Transferee application receives a GCEV_XFER_FAIL termination event and the Transferor application receives a

GCEV_INVOKE_XFER_FAIL termination event. The original primary call is left connected and in the GCST_CONNECTED state from the perspective of both party A and party B.

Figure 12. SIP Call Transfer Failure - REFER Subscription Expires

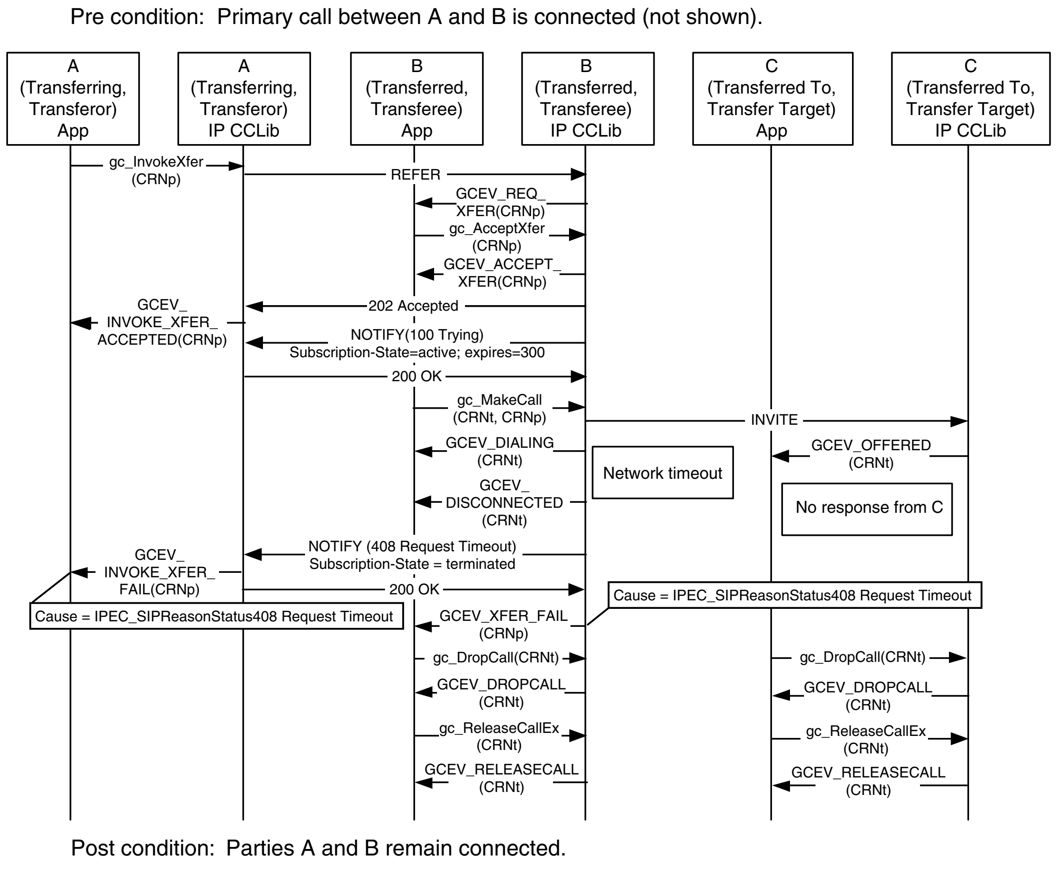


Pre condition: Primary call between A and B is connected (not shown).
A (Transferring, Transferor) App
A (Transferring, Transferor) IP CCLib
B (Transferred, Transferee) App
B (Transferred, Transferee) IP CCLib
C (Transferred To, Transfer Target) App
C (Transferred To, Transfer Target) IP CCLib
gc_InvokeXfer (CRNp)
REFER
GCEV_REQ_ XFER(CRNp)
gc_AcceptXfer (CRNp)
GCEV_ACCEPT_ XFER(CRNp)
202 Accepted
GCEV_ INVOKE_XFER_ ACCEPTED(CRNp)
NOTIFY(100 Trying) Subscription-State=active; expires=300
200 OK
Subscription expires
Subscription expires
GCEV_ INVOKE_XFER_ FAIL(CRNp)
Cause = IPEC_SUBS_EXPIRED
GCEV_XFER_ FAIL(CRNp)
Cause = IPEC_SUBS_EXPIRED
Post condition: Parties A and B remain connected.

No Response from Party C

Figure 13 illustrates a scenario in which the Transfer Target (party C) does not respond to the incoming call from the Transferee (party B) which causes the T4 timer at party B (configured as 20 seconds) to expire. As a result, the Transferee application (party B) receives the GCEV_DISCONNECT event for the transferred call timeout. The original primary call is left connected and in the GCST_CONNECTED state from the perspective of both A and B.

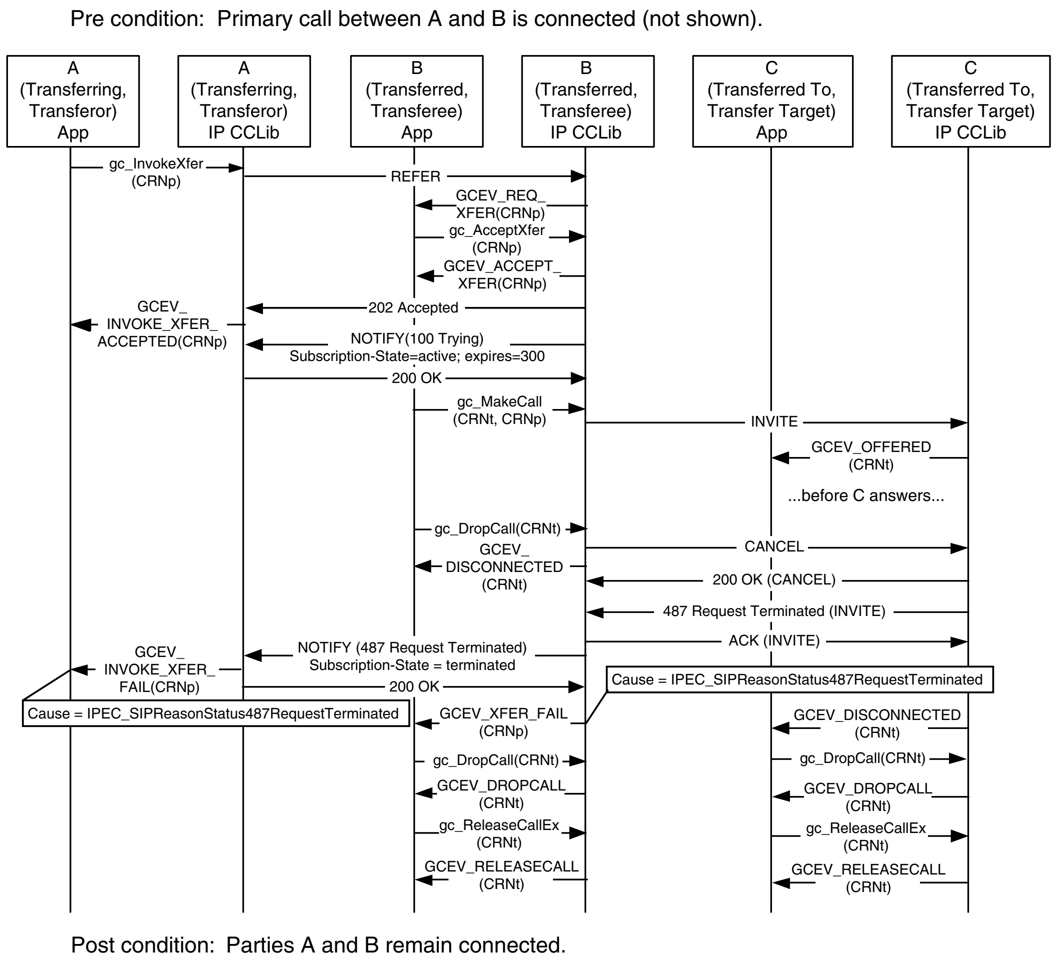
Figure 13. SIP Call Transfer Failure - No Response from Party C


Pre condition: Primary call between A and B is connected (not shown).
A (Transferring, Transferor) App
A (Transferring, Transferor) IP CCLib
B (Transferred, Transferee) App
B (Transferred, Transferee) IP CCLib
C (Transferred To, Transfer Target) App
C (Transferred To, Transfer Target) IP CCLib
gc_InvokeXfer (CRNp)
REFER
GCEV_REQ_XFER(CRNp)
gc_AcceptXfer (CRNp)
GCEV_ACCEPT_XFER(CRNp)
202 Accepted
GCEV_INVOKE_XFER_ACCEPTED(CRNp)
NOTIFY(100 Trying) Subscription-State=active; expires=300
200 OK
gc_MakeCall (CRNt, CRNp)
INVITE
GCEV_DIALING (CRNt)
GCEV_OFFERED (CRNt)
Network timeout
No response from C
GCEV_DISCONNECTED (CRNt)
NOTIFY (408 Request Timeout) Subscription-State = terminated
GCEV_INVOKE_XFER_FAIL(CRNp)
200 OK
Cause = IPEC_SIPReasonStatus408 Request Timeout
Cause = IPEC_SIPReasonStatus408 Request Timeout
GCEV_XFER_FAIL (CRNp)
gc_DropCall(CRNt)
gc_DropCall(CRNt)
GCEV_DROPCALL (CRNt)
GCEV_DROPCALL (CRNt)
gc_ReleaseCallEx (CRNt)
gc_ReleaseCallEx (CRNt)
GCEV_RELEASECALL (CRNt)
GCEV_RELEASECALL (CRNt)
Post condition: Parties A and B remain connected.

Party B Drops Transferred Call Early

Figure 14 illustrates a scenario in which the Transferee (party B) drops the transferred call before receiving a response to the INVITE it sent to party C. As a result, the GCEV_INVOKE_XFER_FAIL termination event is received at the Transferor (party A) and the GCEV_XFER_FAIL termination event is received a the Transferee (party B). The original primary call is left connected and in the GCST_CONNECTED state from the perspective of both A and B.

Figure 14. SIP Call Transfer Failure - Party B Drops Transferred Call Early

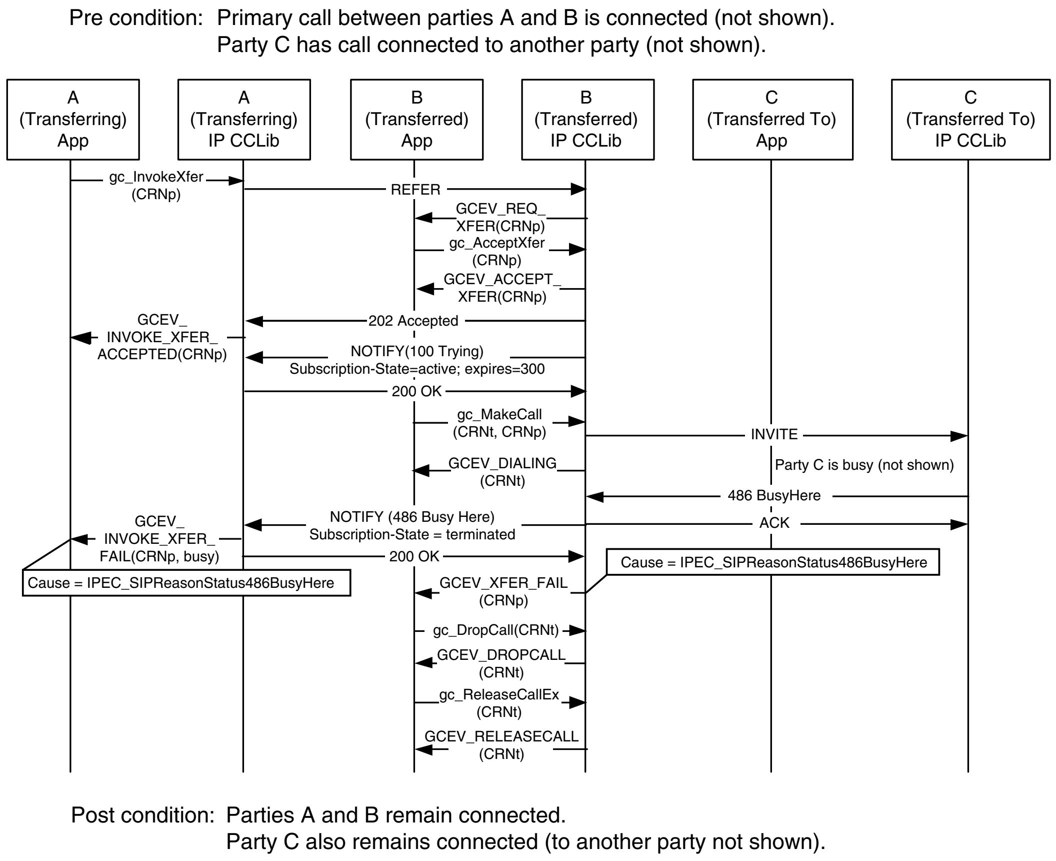


Pre condition: Primary call between A and B is connected (not shown).
A (Transferring, Transferor) App
A (Transferring, Transferor) IP CCLib
B (Transferred, Transferee) App
B (Transferred, Transferee) IP CCLib
C (Transferred To, Transfer Target) App
C (Transferred To, Transfer Target) IP CCLib
gc_InvokeXfer (CRNp)
REFER
GCEV_REQ_XFER(CRNp)
gc_AcceptXfer (CRNp)
GCEV_ACCEPT_XFER(CRNp)
202 Accepted
GCEV_INVOKE_XFER_ACCEPTED(CRNp)
NOTIFY(100 Trying) Subscription-State=active; expires=300
200 OK
gc_MakeCall (CRNt, CRNp)
INVITE
GCEV_OFFERED (CRNt)
...before C answers...
gc_DropCall(CRNt)
CANCEL
GCEV_DISCONNECTED (CRNt)
200 OK (CANCEL)
487 Request Terminated (INVITE)
ACK (INVITE)
GCEV_INVOKE_XFER_FAIL(CRNp)
NOTIFY (487 Request Terminated) Subscription-State = terminated
200 OK
Cause = IPEC_SIPReasonStatus487RequestTerminated
Cause = IPEC_SIPReasonStatus487RequestTerminated
GCEV_XFER_FAIL (CRNp)
GCEV_DISCONNECTED (CRNt)
gc_DropCall(CRNt)
gc_DropCall(CRNt)
GCEV_DROPCALL (CRNt)
GCEV_DROPCALL (CRNt)
gc_ReleaseCallEx (CRNt)
gc_ReleaseCallEx (CRNt)
GCEV_RELEASECALL (CRNt)
GCEV_RELEASECALL (CRNt)
Post condition: Parties A and B remain connected.

Party C Is Busy When Transfer Attempted

Figure 15 illustrates a scenario in which the Transfer Target (party C) is busy at the time the transfer is requested. (This primarily applies to unattended transfers, since the Transferor would be aware that the Transfer Target is busy in an attended transfer.) In this case, the Transferor (party A) receives a GCEV_INVOKE_XFER_FAIL termination event and the Transferee (party B) receives a GCEV_XFER_FAIL termination event. The original primary call is left connected and in the GCST_CONNECTED state from the perspective of both party A and party B.

Figure 15. SIP Call Transfer Failure - Party C Is Busy When Transfer Attempted


Pre condition: Primary call between parties A and B is connected (not shown).
Party C has call connected to another party (not shown).
A (Transferring) App
A (Transferring) IP CCLib
B (Transferred) App
B (Transferred) IP CCLib
C (Transferred To) App
C (Transferred To) IP CCLib
gc_InvokeXfer (CRNp)
REFER
GCEV_REQ_ XFER(CRNp)
gc_AcceptXfer (CRNp)
GCEV_ACCEPT_ XFER(CRNp)
202 Accepted
GCEV_ INVOKE_XFER_ ACCEPTED(CRNp)
NOTIFY(100 Trying) Subscription-State=active; expires=300
200 OK
gc_MakeCall (CRNt, CRNp)
INVITE
GCEV_DIALING (CRNt)
Party C is busy (not shown)
486 BusyHere
NOTIFY (486 Busy Here) Subscription-State = terminated
ACK
GCEV_ INVOKE_XFER_ FAIL(CRNp, busy)
200 OK
Cause = IPEC_SIPReasonStatus486BusyHere
Cause = IPEC_SIPReasonStatus486BusyHere
GCEV_XFER_FAIL (CRNp)
gc_DropCall(CRNt)
GCEV_DROPCALL (CRNt)
gc_ReleaseCallEx (CRNt)
GCEV_RELEASECALL (CRNt)
Post condition: Parties A and B remain connected.
Party C also remains connected (to another party not shown).

1.51.2 Enabling Call Transfer

The call transfer supplementary service is a feature that must be enabled at the time the **gc_Start( )** function is called. Both H.450.2 and SIP call transfer services are enabled at the same time.

The **INIT_IPCCLIB_START_DATA( )** and **INIT_IP_VIRTBOARD( )** functions, which must be called before the **gc_Start( )** function, populate the IPCCLIB_START_DATA and IP_VIRTBOARD structures, respectively, with default values. The default value of the sup_serv_mask field in the IP_VIRTBOARD structure disables the call transfer service for both H.323 and SIP protocols. The default sup_serv_mask field value must therefore be overridden with the value IP_SUP_SERV_CALL_XFER for each Dialogic® IPT Board device on which call transfer is to be enabled. The following code snippet provides an example for two virtual boards:

```
.
.
INIT_IPCCLIB_START_DATA(&ipcclibstart, 2, ip_virtboard);
INIT_IP_VIRTBOARD(&ip_virtboard[0]);
INIT_IP_VIRTBOARD(&ip_virtboard[1]);
ip_virtboard[0].sup_serv_mask = IP_SUP_SERV_CALL_XFER; /* override supp services default */
ip_virtboard[1].sup_serv_mask = IP_SUP_SERV_CALL_XFER; /* override supp services default */
.
.
```

Note: If the application tries to use one of the six IP call transfer functions when call transfer was not explicitly enabled via the IP_VIRTBOARD structure during **gc_Start( )**, the function call fails with an IPERR_SUP_SERV_DISABLED indication.

1.51.3 Using SIP Call Transfer

This section describes specific call transfer procedures when using SIP protocol. The topics covered here include:

- Enabling GCEV_INVOKE_XFER_ACCEPTED Events
- Invoking an Unattended Call Transfer
- Invoking an Attended Call Transfer
- Processing Asynchronous Call Transfer Events
- Handling a Transfer Request
- Making a Transferred Call

1.51.3.1 Enabling GCEV_INVOKE_XFER_ACCEPTED Events

The following code snippet illustrates how to enable the GCEV_INVOKE_XFER_ACCEPTED event type, which is optionally used to notify the application at party A that party B has accepted a transfer request. This event type is disabled by default. This event can be enabled for an individual line device at any time after the line device is opened. The event is enabled in the party A (Transferor) application, and need only be enabled if the application wishes to receive the events. Note that there is no equivalent event in H.450.2.

```
//enable GCEV_INVOKE_XFER_ACCEPTED event

GC_PARM_BLK *t_pParmBlk = NULL;
long request_id;

gc_util_insert_parm_val(&t_pParmBlk, GCSET_CALLEVENT_MSK, GCACT_ADDMSK,
                        sizeof(long), GCMSK_INVOKEXFER_ACCEPTED);

gc_SetConfigData(GCTGT_GCLIB_CHAN,ldev,t_pParmBlk, 0, GCUPDATE_IMMEDIATE, &request_id, EV_SYNC);

gc_util_delete_parm_blk(t_pParmBlk);
```

Disabling the event is done in exactly the same way except that the parameter ID that is set in the GC_PARM_BLK would be GCACT_SUBMSK instead of GCACT_ADDMSK.

1.51.3.2 Invoking an Unattended Call Transfer

The following code snippet illustrates how to invoke an unattended (blind) transfer on a channel that is in the connected state. In this example, the Refer-To header field of the REFER message that is sent is set to “sip:500@192.168.1.10”, while the Referred-By header field is automatically populated by Global Call.

```
int Gc_InvokeXfer(int channel)
{
   INT32  rc;
   GCLIB_MAKECALL_BLK t_gclibmakecallblk;
   GC_MAKECALL_BLK    t_gcmakecallblk = {0};
   char invokeaddr[] = "192.168.1.10";  // party C (TRTSE)
   char phonelist[] = "500";

   /* Invoke transfer */
   memset(&t_gclibmakecallblk, 0, sizeof(GCLIB_MAKECALL_BLK));
   strcpy(t_gclibmakecallblk.destination.address, invokeaddr);
   t_gclibmakecallblk.destination.address_type = GCADDRTYPE_IP;
   t_gclibmakecallblk.destination.address_plan = GCADDRPLAN_UNKNOWN;
   t_gcmakecallblk.gclib = &t_gclibmakecallblk;

   gc_util_insert_parm_ref(&t_pParmBlk, IPSET_CALLINFO, IPPARM_PHONELIST,
                           sizeof(phonelist), phonelist);

   t_gclibmakecallblk.ext_datap = t_pParmBlk;

   rc = gc_InvokeXfer(session[channel].crn, 0, 0, &t_gcmakecallblk, 0, EV_ASYNC);

   gc_util_delete_parm_blk(t_pParmBlk);

   if(GC_SUCCESS != rc)
   {
      printf("GC_APP : [%d] Invoke Xfer failed!!!\n",channel);
      return GC_ERROR;
   }

  return GC_SUCCESS;
}
```

1.51.3.3 Invoking an Attended Call Transfer

Note that it is necessary for the consultation call to be in the connected state at **both** parties before the transfer operation is invoked. If the transferred-to party (party C) is a Global Call application and is not in the connected state when the transfer is invoked, it may fail to receive the Global Call event for the transfer request, which will cause a GCEV_TASKFAIL.

The following code snippet illustrates how a party that is connected to two remote parties, a primary call and a secondary call, invokes a call transfer by sending a REFER to one of the remote parties. The Refer-To, Replaces, and Referred-By header fields in the REFER are automatically filled in by Global Call. Note that the application does not have to specify the Refer-To information in an attended transfer because the secondary call already contains that information.

```
int Gc_InvokeXfer(int primaryChannel, int secondaryChannel)
{
   INT32  rc;
   /* Invoke transfer */
   rc = gc_InvokeXfer(session[primaryChannel].crn, session[secondaryChannel].crn,
                      0, 0, 0, EV_ASYNC);

   if(GC_SUCCESS != rc)
   {
      printf("GC_APP : [%d] Invoke Xfer failed!!!\n",primaryChannel);
      return GC_ERROR;
   }

   return GC_SUCCESS;
}
```

1.51.3.4 Processing Asynchronous Call Transfer Events

The following code snippets illustrate how to handle the asynchronous events that notify applications of the call transfer status as a SIP call transfer proceeds.

```
INT32 processEvtHandler()
{
   METAEVENT    metaEvent;
   GC_PARM_BLK  *parmblkp = NULL;
   :

   int  rc = gc_GetMetaEvent(&metaEvent);
   if (GC_SUCCESS != rc)
   {
      printf("GC_APP : gc_GetMetaEvent() failed\n");
      return rc;
   }

   long evtType = sr_getevttype();
   long evtDev = sr_getevtdev();
   int  g_extIndex = g_lArray[g_evtdev];

   switch (evtType)
   {

      ///////////////////////////////////////////
      // Party A events
      ///////////////////////////////////////////

      case GCEV_INVOKE_XFER_ACCEPTED:
         // remote party has accepted REFER by 2xx response
         printf("Invoke Transfer Accepted By Remote\n");
         break;

      case GCEV_INVOKE_XFER:
         // remote party has notified transfer success in NOTIFY
         printf("Invoke Transfer Successful\n");
         break;

      case GCEV_INVOKE_XFER_FAIL:
         // Invoke Transfer failed by remote NOTIFY or locally
         PrintEventError(&metaEvent);
         break;

      case GCEV_INVOKE_XFER_REJ:
         // Invoke Transfer Rejected by Remote party
         PrintEventError(&metaEvent);
         break;

      /////////////////////////////////////////
      // Party B events
      /////////////////////////////////////////
      case GCEV_REQ_XFER:
         // Incoming transfer request
         GC_REROUTING_INFO *pRerouteInfo = (GC_REROUTING_INFO *)metaEvent.extevtdatap;
         printf("Reroute number = %s\n", pRerouteInfo->rerouting_num);

         if(NULL != pRerouteInfo->parm_blkp)
         {
            // Handle parm blocks
         }

         strcpy(session[g_extIndex].rerouting_num,pRerouteInfo->rerouting_num);
         session[g_extIndex].rerouting_addrblk = *pRerouteInfo->rerouting_addrblkp;

         GC_HandleXferReq(g_extIndex)
         break;

      case GCEV_ACCEPT_XFER:
         // Accepted incoming transfer request
         break;

      case GCEV_ACCEPT_XFER_FAIL:
         // Failed to accept incoming transfer request
         PrintEventError(&metaEvent);
         break;

      case GCEV_REJ_XFER:
         // Rejected incoming transfer request
         break;

      case GCEV_REJ_XFER_FAIL:
         // Failed to reject incoming transfer request
         PrintEventError(&metaEvent);
         break;

      case GCEV_XFER_CMPLT:
         // completed transferred call
         break;

      case GCEV_XFER_FAIL:
         // Failed to complete the transferred call
         PrintEventError(&metaEvent);
         break;

      ////////////////////////////////////////
      // Party C events
      ////////////////////////////////////////

      case GCEV_OFFERED:
         // Received incoming call
         // Normall incoming call handling
         ...
         break;

      ...
   }
   ...
}



void PrintEventError(METAEVENT* pEvent, long evtDev)
{
   int gcError;    /* GlobalCall Error */
   int ccLibId;    /* CC Library ID */
   long ccError;   /* Call Control Library error code */
   char *GCerrMsg; /* GC pointer to error message string */
   char *errMsg;   /* CCLIB pointer to error message string */
   if(gc_ResultValue(pEvent, &gcError, &ccLibId, &ccError)   == GC_SUCCESS)
   {
      gc_ResultMsg(LIBID_GC, (long) gcError, &GCerrMsg);
      gc_ResultMsg(ccLibId, ccError, &errMsg);

      printf("Ld 0x%lx, GC (%d) %s, CC (%ld) %s, (%s)\n",
             evtDev, gcError, GCerrMsg, ccError, errMsg, ATDV_NAMEP(evtDev));
   }
}
```

1.51.3.5 Handling a Transfer Request

The following code snippet illustrates how party B handles an incoming transfer request (REFER). Party B can either reject the request or accept it. Note that if no rejection reason is specified, the default reason, 603 Decline, is used.

```
int Gc_HandleXferReq(int channel)
{
   if(session[channel].ConfigFileParm.autoRejectCallXfer)
   {
      printf("GC_APP : [%d] Reject call xfer request\n",channel);
      if(GC_SUCCESS != gc_RejectXfer(session[channel].crn, IPEC_SIPReasonStatus502BadGateway,
                                  0, EV_ASYNC))
      {
         printf("GC_APP : [%d] Reject call xfer failed on device 0x%lx\n", channel,
                session[channel].ldev);
         PrintEventError(g_evtdev);
         return GC_ERROR;
      }
   }
   else
   {
      printf("GC_APP : [%d] Accept call xfer request\n",channel);
      if(GC_SUCCESS != gc_AcceptXfer(session[channel].crn, 0, EV_ASYNC))
      {
         printf("GC_APP : [%d] Accept call xfer failed on device 0x%lx\n", channel,
                 session[channel].ldev);
         PrintEventError(g_evtdev);
         return GC_ERROR;
      }
   }

   return GC_SUCCESS;
}
```

1.51.3.6 Making a Transferred Call

The following code snippet illustrates how party B makes the transferred call to party C after accepting transfer request from party A

```
int Gc_MakeXferCall(int channelPrimary, int channelXfer)
{

   GC_PARM_BLK         * t_pParmBlk = NULL;
   GCLIB_MAKECALL_BLK  t_gclibmakecallblk ;
   GC_MAKECALL_BLK     t_gcmakecallblk = {0};
   t_gcmakecallblk.gclib = &t_gclibmakecallblk;
   int                 channelXfer;

   memset(&t_gclibmakecallblk, 0, sizeof(GCLIB_MAKECALL_BLK));
   gc_util_insert_parm_val(&t_pParmBlk, GCSET_SUPP_XFER, GCPARM_PRIMARYCALL_CRN,
                           sizeof(unsigned long), session[channelPrimary].crn);

   t_gclibmakecallblk.ext_datap = t_pParmBlk;
   t_gclibmakecallblk.destination = session[channelPrimary].rerouting_addrblk;

   int frc = gc_MakeCall(session[channelXfer].ldev, &session[channelXfer].crn,
                         NULL, &t_gcmakecallblk, 0, EV_ASYNC);

   if((GC_SUCCESS != frc) ||(0 == session[channelXfer].crn))
   {
      printf("GC_APP : [%d] Gc_MakeCall failed: : crn 0x%lx\n", channelXfer,
              session[channelXfer].crn);
      PrintGCError(session[channelXfer].ldev);
   }

   gc_util_delete_parm_blk(t_pParmBlk);

   return GC_SUCCESS;
}
```

1.51.4 SIP Variances for Call Transfer Functions

1.51.4.1 gc_AcceptInitXfer( ) Variances for IP

This function is only available if the call transfer supplementary service was enabled via the sup_serv_mask field in the IP_VIRTBOARD structure when the board device was started.

Variance for SIP

This function does not apply to SIP call transfer. In SIP, party A does not notify party C in advance of requesting an attended (supervised) transfer operation with **gc_InvokeXfer( )**, so there is no opportunity for party C to accept or reject the transfer at the initiation stage.

1.51.4.2 gc_AcceptXfer( ) Variances for IP

This function is only available if the call transfer supplementary service was enabled via the sup_serv_mask field in the IP_VIRTBOARD structure when the board device was started.

The **parmblkp** parameter is ignored for IP technology and should be set to NULL.

The **gc_AcceptXfer( )** function can be used at party B only after receiving a GCEV_REQ_XFER event. The application can obtain information on the rerouting number or address in a GC_REROUTING_INFO data structure dereferenced from the extevtdatap in the METAEVENT structure.

Both the rerouting_num (type char *) and the rerouting_addr (type GCLIB_ADDRESS_BLK) fields of the GC_REROUTING_INFO structure contain the same rerouting address string that was explicitly signaled from party A in SIP call transfers or H.450.2 blind call transfers, or from party C via **gc_AcceptInitXfer( )** in H.450.2 supervised call transfers. The rerouting number to be used in the subsequent

gc_MakeCall( ) at party B can be copied from either element, but must not be a concatenation of both elements because they each contain the same character string.

The remaining elements of the GCLIB_ADDRESS_BLK structure dereferenced from rerouting_addr contain the following:

address_type
GCADDRTYPE_IP

address_plan
GCADDRPLAN_UNKNOWN

sub_address
0 (unused)

sub_address_type
0 (unused)

sub_address_plan
0 (unused)

Variance for SIP

When party B (Transferee or Transferred party) accepts a transfer request via **gc_AcceptXfer( )**, a 202 Accepted message and a NOTIFY(100 Trying) message with Subscription-State= Active is sent to party A (the Transferor or Transferring party). The call control library at party A may optionally generate a GCEV_INVOKE_XFER_ACCEPTED event to notify the application of the acceptance if that event has been enabled for that line device with **gc_SetConfigData( )**.

1.51.4.3 gc_InitXfer( ) Variances for IP

This function is only available if the call transfer supplementary service was enabled via the sup_serv_mask field in the IP_VIRTBOARD structure when the board device was started.

The **parmblkp** and **ret_rerouting_infopp** parameters are ignored and should be set to NULL. The **gc_InitXfer( )** function returns -1 if invalid parameter are specified.

Variance for SIP

The **gc_InitXfer( )** function does not cause any SIP message to be sent to either of the remote parties, and is used only for purposes of synchronizing the Global Call state machine. The GCEV_INIT_XFER termination event that the Transferor receives on the specified CRN after calling **gc_InitXfer( )** is a “dummy” event whose only purpose is to allow synchronization of the Global Call state machine.

1.51.4.4 gc_InvokeXfer( ) Variances for IP

This function is only available if the call transfer supplementary service was enabled via the sup_serv_mask field in the IP_VIRTBOARD structure when the board device was started.

Variance for SIP

The application at party A may optionally be notified by a GCEV_INVOKE_XFER_ACCEPTED event that the transfer request has been accepted by the remote party to which it was sent. (This event has no equivalent in H.450.2.) This event is optional, and is disabled by default. The event may be enabled and disabled on a per-line-device basis via the **gc_SetConfigData( )** function as shown in the following code example.

```
//enable GCEV_INVOKE_XFER_ACCEPTED event for SIP call transfer
GC_PARM_BLK *t_pParmBlk = NULL;
long        request_id;

gc_util_insert_parm_val(&t_parmBlkl, GCSET_CALLEVENT_MSK, GCACT_ADDMSK,
                        sizeof(long), GCMSK_INVOKE_XFER_ACCEPTED);

gc_SetConfigData(GCTGT_GCLIB_CHAN,ldev,t_pParmBlk,0,GCUPDATE_IMMEDIATE,&request_id,EV_SYNC);

gc_util_delete_parm_blk(t_pParmBlk)
```

The specific meaning of the GCEV_INVOKE_XFER termination event for successful transfers is dependant on the application and the transfer scenario(s) it uses. The possible outcomes when Global Call is used by all parties include the following:

- If party A drops the primary call in unattended transfers before the transfer completes, party A does not receive any GCEV_INVOKE_XFER event at all.
- If party B drops the primary call in unattended transfers before the transfer completes, party A receives a GCEV_INVOKE_XFER event that only signifies that party B has sent INVITE to party C.
- For attended transfers or unattended transfers where the primary call is maintained during the transfer, party A receives a GCEV_INVOKE_XFER event which indicates that the transferred call was actually connected between party B and party C.

Table 3 identifies the protocol-specific variances in parameters for **gc_InvokeXfer( )**.

Table 3. gc_InvokeXfer( ) Supported Parameters for SIP

Parameter	Meaning
crn	The CRN of the call between party A and the remote party receiving the transfer request. This is the primary call in an unattended (blind) call transfer, but may be either call for an attended (supervised) transfer.
extracrn	For an attended (supervised) call transfer, the CRN of the call between party A and the remote party *not* receiving the transfer request (i.e. the call not specified in the **crn** parameter). For unattended (blind) call transfers, must be zero.

Table 3. gc_InvokeXfer( ) Supported Parameters for SIP (Continued)

Parameter	Meaning
numberstr	For attended (supervised) call transfers, this parameter is ignored. Set to NULL. For an unattended (blind) call transfer, the address of party C (the rerouting address, which will be signaled to party B) as a string. This address is of the form `user@host; param=value` where: • `user` is a user name or phone number • `host` is a domain name or IP address • `param=value` is an optional additional parameter **Note:** When using the GC_MAKECALL_BLK ***makecallp** parameter to specify the rerouting address, this parameter must be set to NULL.
makecallp	For attended (supervised) call transfers, this parameter is ignored. Set to NULL. For an unattended (blind) call transfer, the address of party C (the rerouting address, which will be signaled to party B) as a GC_MAKECALL_BLK data structure. **Note:** When using the char ***numberstr** parameter to specify the rerouting address, this parameter must be set to NULL.
timeout	Ignored. Set to NULL.

1.51.4.5 gc_RejectInitXfer( ) Variances for IP

This function is only available if the call transfer supplementary service was enabled via the sup_serv_mask field in the IP_VIRTBOARD structure when the board device was started.

Variance for SIP

This function does not apply to SIP call transfer. The SIP stack does not contact the Transfer Target or Transferred-To party (party C) until party A calls **gc_InvokeXfer( )**, so there is no issue of accepting or rejecting the transfer at the initiation stage.

1.51.4.6 gc_RejectXfer( ) Variances for IP

This function is only available if the call transfer supplementary service was enabled via the sup_serv_mask field in the IP_VIRTBOARD structure when the board device was started.

The parameter **parmblkp** is ignored for IP technology.

The **gc_RejectXfer( )** function can only be used at party B, and only after the receipt of a GCEV_REQ_XFER event.

Variance for SIP

The value of the **reason** parameter must be between IPEC_SIPReasonStatusMin and IPEC_SIPReasonStatusMax, as defined in the *gcip_defs.h* header file.

1.52 Early Media

With the Service Update, early media when using H.323 on Dialogic® DM/IP Boards is now supported.

1.52.1 Enabling Early Media

To enable early media on a board level, add the following **PrmEarlyMedia** parameter content in the CONFIG file that corresponds to the PCD file in use on your board:

```
[0x40]
SetParm=0x400a,1        ! PrmEarlyMedia (0=Disabled, 1=Enabled)
```

After the parameter is added, generate an updated FCD file and start system services on the board. See the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* for more information on CONFIG files, PCD files, and FCD files.

Note: Early media support can only be enabled/disabled at the board level via the CONFIG file. Early media support cannot be set at runtime and cannot be set at the channel level.

1.52.2 Early Media Call Setup Scenarios in Global Call

When using IP technology, the establishment of RTP media streaming is normally one of the final steps in establishing and connecting a call. This is in contrast to the public switched telephone network (PSTN), where call progress signaling is commonly provided to the calling party via audible, in-band call progress tones, such as ringback, busy signal, and SIT tones. When implementing a VoIP gateway, it is often imperative to initiate media (RTP) streaming from the local endpoint to the calling party before the call is connected. This capability is commonly referred to as *early media*.

The Dialogic® Global Call IP call control library automatically enables media streaming at the earliest possible point in the pre-connect process. This is generally the earliest point at which the remote endpoint provides the remote RTP/RTCP transport addresses and media capabilities. The precise point at which media can be enabled is dependant on a large number of factors, and the following figures illustrate some common best-case scenarios. Each figure illustrates the Global Call library’s behavior from the application’s perspective, either in the calling party role or in the called party role.

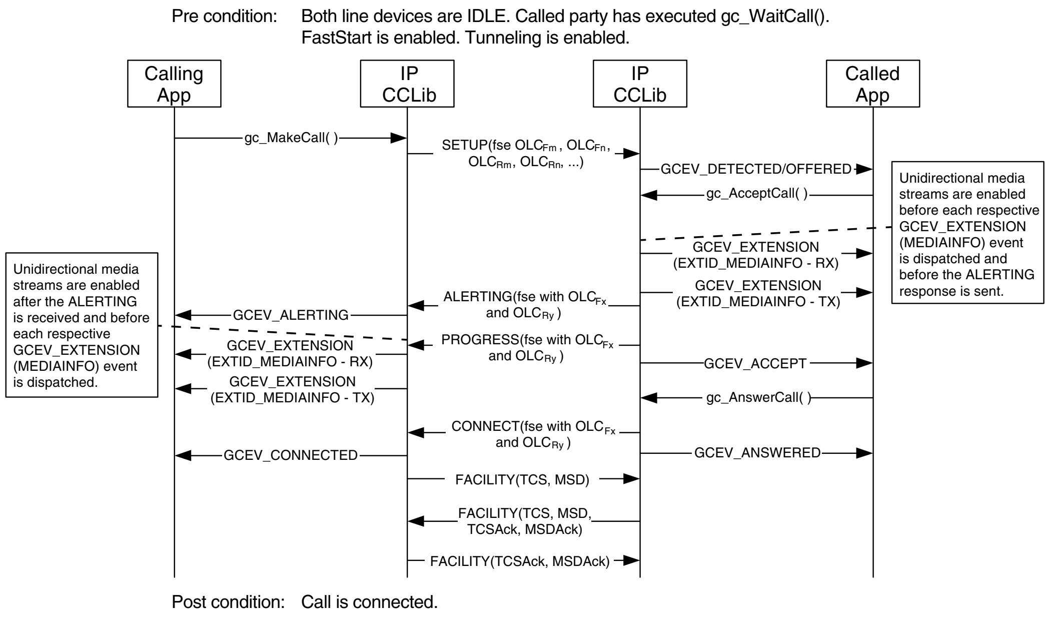
Note that in some cases it is possible to enable streaming in one direction significantly earlier than in the other direction. To take full advantage of this fact, the Global Call IP call control library initially enables a temporary unidirectional connection, then modifies the connection to be full duplex as soon as that is possible. Note that this capability is only supported on Dialogic® IPT Boards, however.

1.52.2.1 H.323 FastStart Mode

The library’s default for H.323 operation enables the Global Call FastStart mode, in which the channel capability information is embedded in a fastStart element (indicated in the figure as “FSE”) that can be sent within the messages of the H.225 Setup exchange rather than using the H.245 messages. (This minimizes the number of round-trip message exchanges and avoids the latency of H.245 channel establishment.) As a calling endpoint, the Global Call library enables media after Alerting is received if the called endpoint supports the fastStart mode. As a called endpoint, the Global Call library enables media in a fastStart connection after the application calls **gc_AcceptCall( )**.

If the calling endpoint sets the MediaWaitForConnect element in the Setup message, the Global Call library does not enable media transmission for a called endpoint until the Connect message is sent. In the case of hardware other than a Dialogic® IPT Board, this means that media is not enabled at all until Connect.

Figure 16. H.323 Early Media, FastStart Mode



Pre condition: Both line devices are IDLE. Called party has executed gc_WaitCall(). FastStart is enabled. Tunneling is enabled.
Calling App
IP CCLib
IP CCLib
Called App
gc_MakeCall( )
SETUP(fse OLCFm , OLCFn, OLCRm, OLCRn, ...)
GCEV_DETECTED/OFFERED
gc_AcceptCall( )
Unidirectional media streams are enabled before each respective GCEV_EXTENSION (MEDIAINFO) event is dispatched and before the ALERTING response is sent.
GCEV_EXTENSION (EXTID_MEDIAINFO - RX)
GCEV_EXTENSION (EXTID_MEDIAINFO - TX)
Unidirectional media streams are enabled after the ALERTING is received and before each respective GCEV_EXTENSION (MEDIAINFO) event is dispatched.
ALERTING(fse with OLCFx and OLCRy)
GCEV_ALERTING
PROGRESS(fse with OLCFx and OLCRy)
GCEV_EXTENSION (EXTID_MEDIAINFO - RX)
GCEV_EXTENSION (EXTID_MEDIAINFO - TX)
GCEV_ACCEPT
gc_AnswerCall( )
CONNECT(fse with OLCFx and OLCRy)
GCEV_CONNECTED
GCEV_ANSWERED
FACILITY(TCS, MSD)
FACILITY(TCS, MSD, TCSAck, MSDAck)
FACILITY(TCSAck, MSDAck)
Post condition: Call is connected.

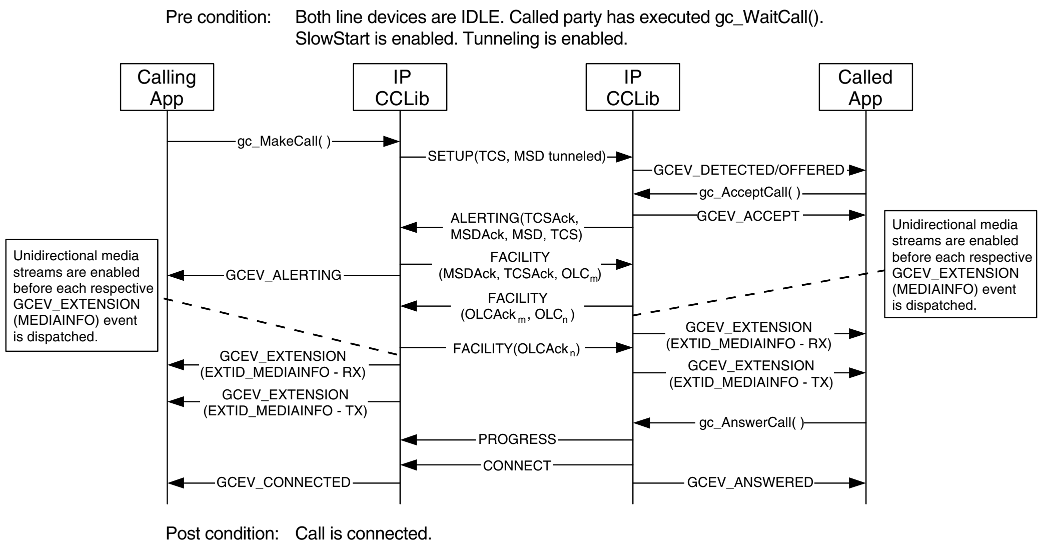
1.52.2.2 H.323 SlowStart Mode

When the application specifies the optional Global Call SlowStart mode, or when one endpoint does not support H.323 fastStart mode, media transmission cannot begin at either endpoint until the remote endpoint has sent its Ack to the appropriate OpenLogicalChannel command.

If the OLCAck that either endpoint receives contains a FlowControlToZero flag parameter that is true, media transmission is not enabled until a subsequent FlowControl message is received. In the case of hardware other than a Dialogic® IPT Board, this means media is not enabled at all until the Flow Control message is received.

If the calling endpoint sets the MediaWaitForConnect element in the Setup message, the called endpoint does not enable media transmission until the Connect message is sent. In the case of hardware other than an IPT Board, this means that media is not enabled at all until Connect.

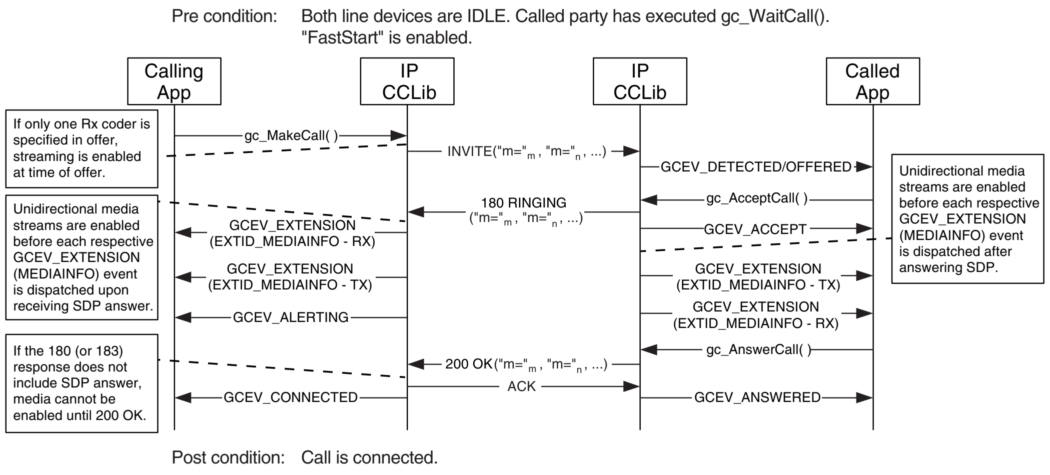
Figure 17. H.323 Early Media, SlowStart Mode


Pre condition: Both line devices are IDLE. Called party has executed gc_WaitCall(). SlowStart is enabled. Tunneling is enabled.
Calling App
IP CCLib
IP CCLib
Called App
gc_MakeCall( )
SETUP(TCS, MSD tunneled)
GCEV_DETECTED/OFFERED
gc_AcceptCall( )
GCEV_ACCEPT
ALERTING(TCSAck, MSDAck, MSD, TCS)
FACILITY (MSDAck, TCSAck, OLCm)
GCEV_ALERTING
FACILITY (OLCAckm, OLCn)
FACILITY(OLCAckn)
GCEV_EXTENSION (EXTID_MEDIAINFO - RX)
GCEV_EXTENSION (EXTID_MEDIAINFO - TX)
GCEV_EXTENSION (EXTID_MEDIAINFO - RX)
GCEV_EXTENSION (EXTID_MEDIAINFO - TX)
gc_AnswerCall( )
PROGRESS
CONNECT
GCEV_CONNECTED
GCEV_ANSWERED
Unidirectional media streams are enabled before each respective GCEV_EXTENSION (MEDIAINFO) event is dispatched.
Unidirectional media streams are enabled before each respective GCEV_EXTENSION (MEDIAINFO) event is dispatched.
Post condition: Call is connected.

1.52.2.3 SIP FastStart Mode (Calling UA Offers SDP)

The SIP protocol does not define distinct “fast start” and “slow start” modes as does H.323, but the Global Call library uses the same FastStart/SlowStart parameter interface to allow applications to specify whether the calling UA offers SDP in its INVITE message or whether it allows the called UA to offer SDP. In the default “FastStart” mode, the calling endpoint offers SDP and the called UA answers.

Figure 18. SIP Early Media, Calling UA Offers SDP

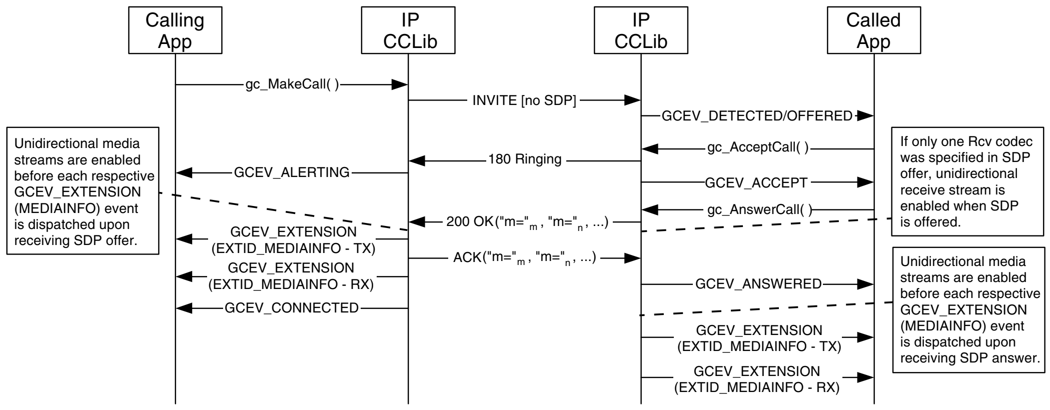

Pre condition: Both line devices are IDLE. Called party has executed gc_WaitCall(). "FastStart" is enabled.
Calling App
IP CCLib
IP CCLib
Called App
If only one Rx coder is specified in offer, streaming is enabled at time of offer.
gc_MakeCall( )
INVITE("m="m, "m="n, ...)
GCEV_DETECTED/OFFERED
gc_AcceptCall( )
180 RINGING ("m="m, "m="n, ...)
GCEV_ACCEPT
Unidirectional media streams are enabled before each respective GCEV_EXTENSION (MEDIAINFO) event is dispatched upon receiving SDP answer.
GCEV_EXTENSION (EXTID_MEDIAINFO - RX)
GCEV_EXTENSION (EXTID_MEDIAINFO - TX)
GCEV_EXTENSION (EXTID_MEDIAINFO - TX)
GCEV_EXTENSION (EXTID_MEDIAINFO - RX)
GCEV_ALERTING
gc_AnswerCall( )
Unidirectional media streams are enabled before each respective GCEV_EXTENSION (MEDIAINFO) event is dispatched after answering SDP.
If the 180 (or 183) response does not include SDP answer, media cannot be enabled until 200 OK.
200 OK("m="m, "m="n, ...)
ACK
GCEV_CONNECTED
GCEV_ANSWERED
Post condition: Call is connected.

1.52.2.4 SIP SlowStart Mode (Calling UA Answers SDP)

When a SIP application sets the optional SlowStart parameter, it specifies that the INVITE message it sends will not contain SDP, so that it is up to the called UA to offer SDP which the calling UA will subsequently answer. In SIP terminology, this is known as *delayed offer*.

Figure 19. SIP Early Media, Calling UA Answers SDP

Pre condition: Both line devices are IDLE. Called party has executed gc_WaitCall(). "SlowStart" is enabled.



Calling App
IP CCLib
IP CCLib
Called App
gc_MakeCall( )
INVITE [no SDP]
GCEV_DETECTED/OFFERED
gc_AcceptCall( )
180 Ringing
GCEV_ALERTING
GCEV_ACCEPT
gc_AnswerCall( )
200 OK("m="m , "m="n , ...)
GCEV_EXTENSION (EXTID_MEDIAINFO - TX)
ACK("m="m , "m="n , ...)
GCEV_EXTENSION (EXTID_MEDIAINFO - RX)
GCEV_ANSWERED
GCEV_CONNECTED
GCEV_EXTENSION (EXTID_MEDIAINFO - TX)
GCEV_EXTENSION (EXTID_MEDIAINFO - RX)
Unidirectional media streams are enabled before each respective GCEV_EXTENSION (MEDIAINFO) event is dispatched upon receiving SDP offer.
If only one Rcv codec was specified in SDP offer, unidirectional receive stream is enabled when SDP is offered.
Unidirectional media streams are enabled before each respective GCEV_EXTENSION (MEDIAINFO) event is dispatched upon receiving SDP answer.

Post condition: Call is connected.

1.52.3 Early Media with Non-Global Call Applications

The **ipm_ModifyMedia( )** function can be used to implement early media from non-Global Call applications (i.e. directly from IPML). Function reference information is given below.

ipm_ModifyMedia( )

Name:	int ipm_ModifyMedia(nDeviceHandle, *pMediaInfo, eDirection, usMode)	
Inputs:	int nDeviceHandle	• IP Media device handle
	IPM_MEDIA_INFO *pMediaInfo	• pointer to media information structure
	eIPM_DATA_DIRECTION eDirection	• data flow direction
	unsigned short usMode	• async or sync mode setting
Returns:	0 on success -1 on failure	
Includes:	srllib.h ipmlib.h	
Category:	Media Session	
Mode:	asynchronous or synchronous	

■ Description

The **ipm_ModifyMedia( )** function modifies various properties of an active media session. This function allows the application to modify the following media session properties:

- direction of the media stream
- IP address and port
- coder properties

For this function to complete successfully, the stream associated with the IP device must be in either active or suspended mode.

The media session properties are changed on the local endpoint as soon the function is called, and this may result in a perceptible artifact (for example, a click or a brief silence) until the remote endpoint makes the corresponding change. For example, if the coder is being changed by the function call, the local endpoint begins transmitting packets using the new coder and stops accepting packets that it receives which use the old coder as soon as the function executes.

Parameter	Description
nDeviceHandle	handle of the IP Media device
pMediaInfo	pointer to structure that contains local channel RTP/RTCP ports and IP address information (or T.38 port and IP address information) See the IPM_MEDIA_INFO data structure page for details.

Parameter	Description
eDirection	media operation enumeration The eIPM_DATA_DIRECTION data type is an enumeration which defines the following values: • DATA_IP_RECEIVEONLY – receive data from the IP network but do not send data • DATA_IP_SENDONLY – send data to the IP network but do not receive data • DATA_IP_TDM_BIDIRECTIONAL – full duplex data path between IP network and TDM • DATA_IP_INACTIVE – allow RTCP while blocking RTP packets • DATA_IP_NULL – do not modify the direction of the current session; the previous direction remains in effect. This value is used when changing the coder and/or IP address without changing the direction.
usMode	operation mode Set to EV_ASYNC for asynchronous execution or to EV_SYNC for synchronous execution

■ Termination Events

In asynchronous mode, the function returns 0 if the operation was initiated successfully. Completion of the operation is indicated by receipt of a termination event:

IPMEV_MODIFY_MEDIA
: Indicates successful completion; that is, modified media information was set and the session has been started.

IPMEV_MODIFY_MEDIA_FAIL
: Indicates that the modify media operation failed. The characteristics of the media session remain as they were before the function was called.

■ Cautions

None.

■ Errors

If the function returns -1 to indicate failure, call **ATDV_LASTERR( )** and **ATDV_ERRMSGP( )** to return one of the following errors:

EIPM_BADPARM
: Invalid parameter

EIPM_BUSY
: Channel is busy

EIPM_INTERNAL
: Internal error

EIPM_INV_MODE
: Invalid mode

EIPM_INV_STATE
: Invalid state. Initial command did not complete before another function call was made.

EIPM_SYSTEM
: System error

■ Example

The following sample code changes the coder from G.711 mu-law to G.711 A-law and also changes the direction.

```
#include <stdio.h>
#include <string>
#include <srllib.h>
#include <ipmlib.h>

typedef long int(*HDLR)(unsigned long);
void CheckEvent();

void main()
{
   /*
   .
   .
   Main Processing
   .
   .
   */

   /*
   Set the media properties for a remote party using IP device handle, nDeviceHandle.
   ASSUMPTION: A valid nDeviceHandle was obtained from prior call to ipm_Open().
   */

   IPM_MEDIA_INFO MediaInfo;
   MediaInfo.unCount = 4;

   MediaInfo.MediaData[0].eMediaType = MEDIATYPE_REMOTE_RTP_INFO;
   MediaInfo.MediaData[0].mediaInfo.PortInfo.unPortId = 2328;
   strcpy(MediaInfo.MediaData[0].mediaInfo.PortInfo.cIPAddress, "111.21.0.9\n");

   MediaInfo.MediaData[1].eMediaType = MEDIATYPE_REMOTE_RTCP_INFO;
   MediaInfo.MediaData[1].mediaInfo.PortInfo.unPortId = 2329;
   strcpy(MediaInfo.MediaData[1].mediaInfo.PortInfo.cIPAddress, "111.41.0.9\n");

   MediaInfo.MediaData[2].eMediaType = MEDIATYPE_REMOTE_CODER_INFO;
   MediaInfo.MediaData[2].mediaInfo.CoderInfo.eCoderType = CODER_TYPE_G711ULAW64K;
   MediaInfo.MediaData[2].mediaInfo.CoderInfo.eFrameSize = (eIPM_CODER_FRAMESIZE) 30;
   MediaInfo.MediaData[2].mediaInfo.CoderInfo.unFramesPerPkt = 1;
   MediaInfo.MediaData[2].mediaInfo.CoderInfo.eVadEnable = CODER_VAD_DISABLE;
   MediaInfo.MediaData[2].mediaInfo.CoderInfo.unCoderPayloadType = 0;
   MediaInfo.MediaData[2].mediaInfo.CoderInfo.unRedPayloadType = 0;

   MediaInfo.MediaData[3].eMediaType = MEDIATYPE_LOCAL_CODER_INFO;
   MediaInfo.MediaData[3].mediaInfo.CoderInfo.eCoderType = CODER_TYPE_G711ULAW64K;
   MediaInfo.MediaData[3].mediaInfo.CoderInfo.eFrameSize = (eIPM_CODER_FRAMESIZE) 30;
   MediaInfo.MediaData[3].mediaInfo.CoderInfo.unFramesPerPkt = 1;
   MediaInfo.MediaData[3].mediaInfo.CoderInfo.eVadEnable = CODER_VAD_DISABLE;
   MediaInfo.MediaData[3].mediaInfo.CoderInfo.unCoderPayloadType = 0;
   MediaInfo.MediaData[3].mediaInfo.CoderInfo.unRedPayloadType = 0;
```

```
    if (ipm_StartMedia(nDeviceHandle, &MediaInfo, DATA_IP_TDM_BIDIRECTIONAL, EV_SYNC) == -1)
    {
       printf("ipm_StartMediaInfo failed for device name = %s with error = %d\n",
       ATDV_NAMEP(nDeviceHandle), ATDV_LASTERR(nDeviceHandle));
       /*
       .
       .
       Perform Error Processing
       .
       .*/
    }
    /*
    .
    . Continue processing
    .
    */

    MediaInfo.unCount = 2;
    MediaInfo.MediaData[0].eMediaType = MEDIATYPE_REMOTE_CODER_INFO;
    MediaInfo.MediaData[0].mediaInfo.CoderInfo.eCoderType = CODER_TYPE_G711ALAW64K;
    MediaInfo.MediaData[0].mediaInfo.CoderInfo.eFrameSize = (eIPM_CODER_FRAMESIZE) 30;
    MediaInfo.MediaData[0].mediaInfo.CoderInfo.unFramesPerPkt = 1;
    MediaInfo.MediaData[0].mediaInfo.CoderInfo.eVadEnable = CODER_VAD_DISABLE;
    MediaInfo.MediaData[0].mediaInfo.CoderInfo.unCoderPayloadType = 0;
    MediaInfo.MediaData[0].mediaInfo.CoderInfo.unRedPayloadType = 0;

    MediaInfo.MediaData[1].eMediaType = MEDIATYPE_LOCAL_CODER_INFO;
    MediaInfo.MediaData[1].mediaInfo.CoderInfo.eCoderType = CODER_TYPE_G711ALAW64K;
    MediaInfo.MediaData[1].mediaInfo.CoderInfo.eFrameSize = (eIPM_CODER_FRAMESIZE) 30;
    MediaInfo.MediaData[1].mediaInfo.CoderInfo.unFramesPerPkt = 1;
    MediaInfo.MediaData[1].mediaInfo.CoderInfo.eVadEnable = CODER_VAD_DISABLE;
    MediaInfo.MediaData[1].mediaInfo.CoderInfo.unCoderPayloadType = 0;
    MediaInfo.MediaData[1].mediaInfo.CoderInfo.unRedPayloadType = 0;

    if (ipm_ModifyMedia(nDeviceHandle, &MediaInfo, DATA_IP_SENDONLY, EV_SYNC) == -1)
    {
       printf("ipm_Modify failed for device name = %s with error = %d\n",
       ATDV_NAMEP(nDeviceHandle), ATDV_LASTERR(nDeviceHandle));
       /*
       .
       .
       Perform Error Processing
       */
    }


    /*
    .
    .
    continue processing
    .
    */

}
```

See Also

- **ipm_StartMedia( )**

1.53 Global Call SS7 Enhancements

The Service Update includes some enhancements to Dialogic® Global Call SS7:

- Enhances the robustness against an abrupt Global Call application exit
- Brings the **gc_MakeCall( )** function **timeout** behavior in line with the Global Call specification, i.e., specifies the time interval (in seconds) during which the call must be established

1.54 Conference Bridging on Dialogic® DI Boards

With the Service Update, the ability to bridge conference resources across Dialogic® DI Boards is now supported. This feature is applicable to the following boards:

- Dialogic® DI0408LSAR2 Switching Boards
- Dialogic® DISI16R2, DISI24R2, and DISI32R2 Switching Boards

1.54.1 Feature Description

Conference bridging allows the parties from separate conferences to speak with and/or listen to one another. Conference bridging can be used to effectively expand a conference beyond the maximum size allowed by your particular configuration.

The following table shows the conference densities for DI Boards with bridging and without bridging. Note that the creation of a conference bridge consumes a conference resource on each end of the bridge. For example, bridging of two 5-party conferences consumes a total of 12 conference resources. For this reason, the maximum number of parties per board is reduced by at least one if bridging is used.

Table 4. Conference Densities on Dialogic® DI Boards

Board	Max. Conferences Per Board	Max. Parties Per Board	Max. Parties Per Conference		Notes (EC = Echo Cancellation)
			Without Bridging	With Bridging†	
DISI16R2	5	16	16	30	with or without EC
DISI24R2	5	16	16	30	with or without EC
DISI32R2	5	16	16	30	without EC
	5	12	12	22	with EC
DI0408LSAR2	3	9	9	16	with or without EC

†Values in this column are for two boards being used with a single bridge (i.e., two conferences bridged together). The maximum parties per board remains unchanged, and the new density is achieved by creating a separate conference on each board and then bridging the two conferences.
Larger conferences can be created by bridging more than two conferences using a star configuration. See the *Dialogic® Audio Conferencing API Programming Guide* for further information.

On DI Boards, bridging is supported via the Dialogic® Audio Conferencing (DCB) API. The **dcb_CreateBridge( )** function establishes a conference bridge, and the

dcb_DeleteBridge( ) function deletes a conference bridge. The conference bridging feature uses the TS_BRIDGECDT data structure to provide information about the conference bridge.

Notes: ***1.*** Since the DI Boards support fixed routing, all resources (player, recorder, etc.) are permanently coupled to the station interface device and cannot be routed to a conference. This means that prompts cannot be played into a conference using resources from these boards. Routable voice resources are needed from another board (e.g., Dialogic® DM/V2400A Boards) in order to play prompts into a conference or record a conference.

2. See the Conference Bridging chapter in the *Dialogic® Audio Conferencing API Programming Guide* for other limitations and for further information about conference bridging.

1.54.2 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the Dialogic® Audio Conferencing (DCB) API, see the following documents:

- *Dialogic® Audio Conferencing API Programming Guide*
- *Dialogic® Audio Conferencing API Library Reference*

1.55 New Parameter for Order of DNIS and ANI

A new parameter has been added to the country dependent parameter (CDP) files for all countries/protocols that use the pdk_r2_io protocol module.

CDP_In_ANIBeforeDNIS (Inbound)

The **CDP_In_ANIBeforeDNIS** parameter specifies the order of DNIS, ANI, and Category digits. The order in which a switch sends DNIS, ANI, and Category information may be different from the default behavior for a country/protocol. So this parameter allows for two scenarios:

- DNIS+CAT1+DNIS+ANI+CAT2 (default)
- DNIS+CAT1+ANI+DNIS+CAT2

Possible values for this parameter are:

- 0 [default]: DNIS digits are received before ANI, in the pattern DNIS+CAT1+DNIS+ANI+CAT2.
- 1: ANI digits are received before the rest of DNIS, in the pattern DNIS+CAT1+ANI+DNIS+CAT2.

For further information about CDP files, see the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*.

1.56 New Channel Block Timer for NTT Protocol

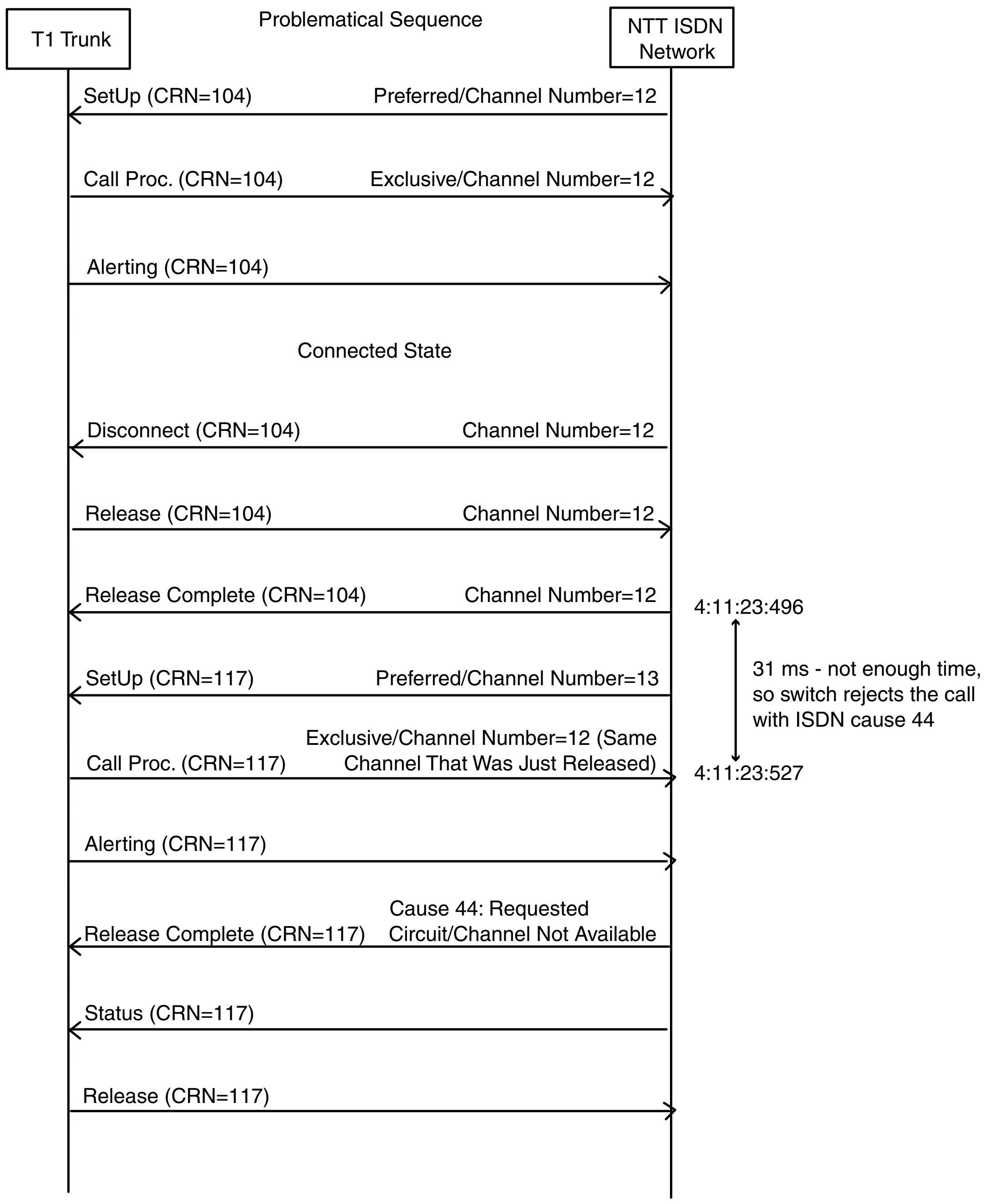
A new channel block timer parameter has been added to the *ntt.prm* file for the ISDN NTT protocol. The purpose of this timer is to block the first response (i.e., PROCEEDING, ALERTING, or CONNECT) to an incoming call (i.e., SETUP message) on a channel, if that same channel has just recently had a previous call released. The channel is blocked for the amount of time specified by this new channel block timer parameter.

This feature is supported on Dialogic® D/240JCT-T1, D/480JCT-1T1, and D/480JCT-2T1 Media Boards:

1.56.1 Feature Description

On some NTT switches, if a call initiated from the switch is responded to with a PROCEEDING message too quickly (i.e., response to the SETUP message) on a recently released channel, the switch rejects the call with ISDN cause 44 (requested circuit/channel not available) and sends an error message (voice message) to the subscriber. This also applies to other first response messages like ALERTING and CONNECT.

For example, the following figure gives an example of a problematical sequence where ISDN cause 44 is returned when channel 12 is reassigned in 31 milliseconds.


Problematical Sequence
T1 Trunk
NTT ISDN Network
SetUp (CRN=104)
Preferred/Channel Number=12
Call Proc. (CRN=104)
Exclusive/Channel Number=12
Alerting (CRN=104)
Connected State
Disconnect (CRN=104)
Channel Number=12
Release (CRN=104)
Channel Number=12
Release Complete (CRN=104)
Channel Number=12
4:11:23:496
SetUp (CRN=117)
Preferred/Channel Number=13
31 ms - not enough time, so switch rejects the call with ISDN cause 44
Call Proc. (CRN=117)
Exclusive/Channel Number=12 (Same Channel That Was Just Released)
4:11:23:527
Alerting (CRN=117)
Release Complete (CRN=117)
Cause 44: Requested Circuit/Channel Not Available
Status (CRN=117)
Release (CRN=117)

For compatibility with these NTT switches, the new channel block timer can be used in order to avoid sending the first response to the B channel that was just released by the previous call. The channel block time is the amount of time to hold off the first response message from being sent out too quickly to the network on a recently released channel. The amount of channel block time that the switches typically need is in the range of 0.7-1.0 second. This helps to prevent the call rejection described above.

Since not all switches operate this way, setting the channel block timer is optional.

When enabled, the channel block timer is started upon the sending or receiving of the final clearing message on a particular channel (clearing message could be RELEASE or RELEASE COMPLETE), and that channel will not be able to accept another call until this timer has expired.

- If a call is received on a channel while the channel block timer is still running, and if the Channel ID IE is set to “Preferred” or “Any Channel”, then the call is simply routed to the next available B channel.
- However, if the Channel ID IE is set to “Exclusive”, or if no B channel on the board is available for which the block timer condition is met, then the call is rejected with ISDN cause 44.

1.56.2 New Parameter

The new parameter in the *ntt.prm* file to set the channel block timer is:

```
;--- The NTT Channel Block Delay value.  LSB is 10 ms.
;--- This is a 2 byte value, but the maximum that will be considered is 255, or 0xFFH.
;--- If a value more than 255 is specified then 255 will be considered.
;
;--- The default value is 00H.
003C 00
```

The channel block delay time (003C parameter) can be set from 0 to 255, where the values are increments of 10 milliseconds. For example, a setting of 255 = 255 x 10 milliseconds or 2.55 seconds of delay. Parameter values must be entered in hexadecimal, so 255 would be entered as FF.

Typical values for this timer have been found to lie in the range of 0.7-1.0 second.

Any non-zero value enables the channel blocking feature for the specified time. The default value for the parameter is zero, which disables the channel blocking feature.

The *ntt.prm* file is installed in the *data* subdirectory of the Dialogic® Software home directory (normally *C:\Program Files\Dialogic\data*). For further information about JCT board configuration, see the *Dialogic® Springware Architecture Products on Windows® Configuration Guide*.

1.57 Mixing ISDN and CAS on Dialogic® DM/V-B Boards

With the Service Update, you can now mix ISDN and CAS protocols on the same Dialogic® DMV600BTEP or DMV1200BTEP Media Board, with automatic A-law/Mu-law conversion.

1.57.1 Feature Description

The Trunk Configuration property sheet of the Dialogic® Configuration Manager (DCM) contains parameters for configuring the interfaces on a DMV600BTEP or DMV1200BTEP Board. The procedure is described in detail in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Previously, there were five different groups of protocols; now there are only two groups. As before, only protocols within the same group can be used on the same board. The groups are:

Group 1	Group 2
4ESS (T1)	DPNSS
5ESS (T1)	DASS2
NTT (T1)	
NI2 (T1)	
DMS (T1)	
QSIGT1 (T1)	
QSIGE1 (E1)	
NET5 (E1)	
T1CC (T1)	
CAS (T1)	
E1CC (E1)	
R2MF (E1)	

Each of the trunks on a DMV600BTEP or DMV1200BTEP Board must be assigned one of the protocols listed above. You can assign a different value to each trunk, but all the values must have the same group number. This allows you to mix ISDN and CAS protocols on the same board. Only DPNSS and DASS2 protocols cannot be mixed with the other protocols.

Note: The DM/V-B Boards also allow the mixing of T1 and E1 protocols on the same board, with automatic A-law/Mu-law conversion. The following considerations apply.

You can set the network interfaces to T1 or E1 in the same system, regardless of the CT Bus PCM encoding method (A-law or Mu-law). For example, if the PCM encoding method on the CT Bus is set to A-law, a DMV600BTEP or DMV1200BTEP Board that has some or all of its network (front end) interfaces configured for T1 will automatically convert the A-law data sent to and received from the CT Bus to Mu-law for transmitting and receiving on the T1 configured front ends. The board will always transmit to and receive from each front end using the PCM encoding method determined by the network interface setting.

1.57.2 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For detailed information about configuring DMV600BTEP and DMV1200BTEP Boards, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Note: The online bookshelf has not been updated for this feature, so the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* does not currently include information about mixing ISDN and CAS protocols on the same board.

1.58 Implementation of ROLM Call Waiting LED

When using Dialogic® D/42JCT-U and D/82JCT-U PBX Integration Boards and PBX Integration software, the **d42_indicators( )** function can now return the LED status of the Call Waiting LED for the ROLMphone 400.

The following table and figure show the ROLMphone 400 direct key dialing strings for feature keys and the bytes containing the indicator status. This supersedes the information that is currently in the *Dialogic® PBX Integration Board User's Guide* and *Dialogic® PBX Integration Software Reference* for the Siemens ROLM PBX.

Table 5. ROLMphone 400 Direct Key Dialing Strings for Feature Keys

Byte	Key Description	Dial String
00	Feature Key 09 - LINE	<ESC>KA
01	Feature Key 08	<ESC>KB
02	Feature Key 07	<ESC>KC
03	Feature Key 06 - CLEAR (flash)	<ESC>KD
04	Feature Key 05	<ESC>KE
05	Feature Key 04	<ESC>KF
06	Feature Key 03	<ESC>KG
07	Feature Key 02	<ESC>KH
08	Feature Key 01 - MAILBOX	<ESC>KI
09	Feature Key 15	<ESC>KJ
10	Feature Key 14	<ESC>KK
11	Feature Key 13	<ESC>KL
12	Feature Key 12	<ESC>KM
13	Feature Key 11	<ESC>KN
14	Feature Key 20 - PROG (program)	<ESC>KO
15	Feature Key 19	<ESC>KP
16	Feature Key 18	<ESC>KQ
17	Feature Key 17	<ESC>KR
18	Feature Key 16	<ESC>KS
19	Feature Key 25	<ESC>KT
20	Feature Key 24	<ESC>KU

Table 5. ROLMphone 400 Direct Key Dialing Strings for Feature Keys (Continued)

Byte	Key Description	Dial String
21	Feature Key 23	<ESC>KV
22	Feature Key 22	<ESC>KW
23	Feature Key 21	<ESC>KX
24	Feature Key 35	<ESC>KY
25	Feature Key 34	<ESC>KZ
26	Feature Key 33	<ESC>Ka
27	Feature Key 32	<ESC>Kb
28	Feature Key 31	<ESC>Kc
29	Feature Key 29	<ESC>Kd
30	Feature Key 28	<ESC>Ke
31	Feature Key 27	<ESC>Kf
32	Feature Key 26	<ESC>Kg
33	Feature Key 37 - MWCTR (Message Waiting Control)	<ESC>Kh
34	Feature Key 36 - SPEAKER	<ESC>Ki
	Feature Key 40 - Volume Down	<ESC>Kj
	Feature Key 39 - Volume Up	<ESC>Kk
	Feature Key 10	<ESC>Kl
	Feature Key 30	<ESC>Km
	Feature Key 38 - XFER	<ESC>Kn
35	Call Waiting LED	

	Feature Key 09	Feature Key 08	Feature Key 07	Feature Key 06	Feature Key 05	Feature Key 04	Feature Key 03	Feature Key 02	Feature Key 01	Feature Key 15	Feature Key 14	Feature Key 13	Feature Key 12	Feature Key 11	Feature Key 20	Feature Key 19	Feature Key 18	Feature Key 17	Feature Key 16	Feature Key 25	Feature Key 24	Feature Key 23	Feature Key 22	Feature Key 21
Data	61	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00	00
Byte	00	01	02	03	04	05	06	07	08	09	10	11	12	13	14	15	16	17	18	19	20	21	22	23

Data	00	00	00	00	00	00	00	00	00	00	00	00	xx	xx	xx	xx	xx	xx	xx	xx	xx	xx	xx	xx
Byte	24	25	26	27	28	29	30	31	32	33	34	35	36	37	38	39	40	41	42	43	44	45	46	47
	Feature Key 35	Feature Key 34	Feature Key 33	Feature Key 32	Feature Key 31	Feature Key 29	Feature Key 28	Feature Key 27	Feature Key 26	Feature Key 37	Feature Key 36	Call Waiting Light												

1.59 Enhanced Special Information Tone Frequency Detection on Dialogic® DM3 Boards

This release provides the following enhancements to Special Information Tone (SIT) frequency detection on Dialogic® DM3 Boards:

- Broader default SIT sequence definitions to allow greater coverage for SIT sequences detected in the field.
- Three new SIT sequence definitions in the SIT tone set for DM3 Boards. The new SIT sequences are: InterLATA no circuit (TID_SIT_NC_INTERLATA), InterLATA reorder tone (TID_SIT_RO_INTERLATA), and ineffective other (TID_SIT_IO).
- A new catch all SIT sequence definition to cover SIT sequences that fall outside the range of the defined SIT sequences.
- DM3 Board support for the **ATDX_CRTNID( )** function to allow retrieval of the SIT ID.

Details on these enhancements are provided next.

1.59.1 New SIT Sequence Definitions

The table below provides the default tone definitions for SIT sequences used on DM3 Boards. This information is not currently documented in the *Dialogic® Voice API Library Reference* in the Dialogic® System Release 6.0 PCI for Windows® bookshelf.

The table describes existing SIT sequences that have broader definitions as well as new SIT sequences.

This table is explained in further detail:

- The values in the Freq. column represent minimum and maximum values in Hz.
- Time refers to minimum and maximum on time in 10 msec units; the maximum off time between each tone is 5 (or 50 msec).
- The repeat count is 1 for all SIT segments.
- N/A means not applicable.
- For TID_SIT_ANY, the frequency and time of the first and second segments are open; that is, they are ignored. Only the frequency of the third segment is relevant.
- The tone IDs have aliases:
  - TID_SIT_NO_CIRCUIT (TID_SIT_NC)
  - TID_SIT_OPERATOR_INTERCEPT (TID_SIT_IC)
  - TID_SIT_VACANT_CIRCUIT (TID_SIT_VC)
  - TID_SIT_REORDER_TONE (TID_SIT_RO)
  - TID_SIT_NO_CIRCUIT_INTERLATA (TID_SIT_NC_INTERLATA)
  - TID_SIT_REORDER_TONE_INTERLATA (TID_SIT_RO_INTERLATA)
  - TID_SIT_INEFFECTIVE_OTHER (TID_SIT_IO)

Table 6. Special Information Tone Definitions (DM3 Boards)

SIT		1st Segment		2nd Segment		3rd Segment	
Tone ID	**Description**	**Freq.**	**Time**	**Freq.**	**Time**	**Freq.**	**Time**
TID_SIT_NC	No Circuit Found	950/1020	32/45	1400/1450	32/45	1740/1850	N/A
TID_SIT_IC	Operator Intercept	874/955	15/30	1310/1430	15/30	1740/1850	N/A
TID_SIT_VC	Vacant Circuit	950/1020	32/45	1310/1430	15/30	1740/1850	N/A
TID_SIT_RO	Reorder (system busy)	874/955	15/30	1400/1450	32/45	1740/1850	N/A
TID_SIT_NC_ INTERLATA	InterLATA No Circuit Found	874/955	32/45	1310/1430	32/45	1740/1850	N/A
TID_SIT_RO_ INTERLATA	InterLATA Reorder (system busy)	950/1020	15/30	1310/1430	32/45	1740/1850	N/A
TID_SIT_IO	Ineffective Other	874/955	32/45	1400/1450	15/30	1740/1850	N/A
TID_SIT_ANY	Catch all tone definition	Open	Open	Open	Open	1725/1825	N/A

1.59.2 ATDX_CRTNID( ) Support on Dialogic® DM3 Boards

The **ATDX_CRTNID( )** function is now supported on Dialogic® DM3 Boards. This information is not currently documented in the *Dialogic® Voice API Library Reference* in the Dialogic® System Release 6.0 PCI for Windows® bookshelf.

On DM3 Boards, the following new tone IDs can now be returned by **ATDX_CRTNID( )**:

Tone ID	Description
TID_SIT_IC TID_SIT_OPERATOR_INTERCEPT	Operator intercept SIT sequence
TID_SIT_IO TID_SIT_INEFFECTIVE_OTHER	Ineffective other SIT sequence
TID_SIT_NC TID_SIT_NO_CIRCUIT	No circuit found SIT sequence
TID_SIT_NC_INTERLATA TID_SIT_NO_CIRCUIT_INTERLATA	InterLATA no circuit found SIT sequence
TID_SIT_RO TID_SIT_REORDER_TONE	Reorder (system busy) SIT sequence
TID_SIT_RO_INTERLATA TID_SIT_REORDER_TONE_INTERLATA	InterLATA reorder (system busy) SIT sequence
TID_SIT_VC TID_SIT_VACANT_CIRCUIT	Vacant circuit SIT sequence
TID_SIT_ANY	Catch all (returned for a SIT sequence that falls outside the range of known default SIT sequences)

Updated example code is provided for this function as follows.

```
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>

main()
{
   DX_CAP  cap_s;
   int    ddd, car;
   char  *chnam, *dialstrg;
   long tone_id;

   chnam    = "dxxxB1C1";
   dialstrg = "L1234";

   /*
    *  Open channel
    */
   if ((ddd = dx_open( chnam, NULL )) == -1 ) {
      /* handle error */
   }

   /*
    *  Dial
    */
   printf("Dialing %s\n", dialstrg );
   car = dx_dial(ddd,dialstrg,(DX_CAP *)&cap_s,DX_CALLP|EV_SYNC);
   if (car == -1) {
      /* handle error */
   }

   switch( car ) {
   case CR_NODIALTONE:
      switch( ATDX_DTNFAIL(ddd) ) {
      case 'L':
         printf(" Unable to get Local dial tone\n");
         break;
      case 'I':
         printf(" Unable to get International dial tone\n");
         break;
      case 'X':
         printf(" Unable to get special eXtra dial tone\n");
         break;
      }
      break;

   case CR_BUSY:
      printf(" %s engaged - %s detected\n", dialstrg,
            (ATDX_CRTNID(ddd) == TID_BUSY1 ? "Busy 1" : "Busy 2") );
      break;

   case CR_CNCT:
      printf(" Successful connection to %s\n", dialstrg );
      break;

   case CR_CEPT:
      printf(" Special tone received at %s\n", dialstrg );
      tone_id  = ATDX_CRTNID(ddd);  //ddd is handle that is returned by
dx_open()

      switch (tone_id) {
     case TID_SIT_NC:
        break;
     case TID_SIT_IC:
        break;
     case TID_SIT_VC:
        break;
     case TID_SIT_RO:
        break;
     case TID_SIT_NC_INTERLATA:
        break;
     case TID_SIT_RO_INTERLATA:
        break;
     case TID_SIT_IO:
        break;
     case TID_SIT_ANY:
        break;
        }
        break;

   default:
      break;
   }

   /*
    *  Set channel on hook
    */
   if ((dx_sethook( ddd, DX_ONHOOK, EV_SYNC )) == -1) {
      /* handle error */
   }

   dx_close( ddd );
}
```

1.59.3 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the Dialogic® Voice API, see the following documents:

- *Dialogic® Voice API Programming Guide*
- *Dialogic® Voice API Library Reference*

Note: The online bookshelf has not been updated for this feature. The following observations are worth noting:

- In the *Dialogic® Voice API Programming Guide*, the section about SIT Frequency Detection is superseded by the information in this Release Update.
- In the *Dialogic® Voice API Library Reference*, the description of the **ATDX_CRTNID( )** function does not currently indicate that it is supported on DM3 Boards, with the new tone IDs shown in this Release Update.

1.60 Enhanced GCAMS on Dialogic® DM3 Boards

This release provides the following enhancements to the Global Call Alarm Management System (GCAMS) for Dialogic® DM/V and DM/V-A Media Boards:

- Ability for the application to be notified of several new T1/E1 alarms, supported on ISDN, CAS, and R2MF protocols through existing GCEV_ALARM, GCEV_BLOCKED, and GCEV_UNBLOCKED
- Ability to change default threshold values for the new alarms through the configuration file
- Ability for the GCAMS functions in the Dialogic® Global Call API library to recognize the new alarms
- Default values of non-blocking and "no notify" for each new alarm with the ability to change each value via **gc_SetAlarmConfiguration( )**
- Alarm reporting behavior for the new alarms is the same as the behavior on Springware boards

Details on these enhancements are provided next.

1.60.1 New E1 Alarms

The following table lists new alarms for E1 technology. These new alarms are non-blocking and not received by default. To change these default values, use the **gc_SetAlarmConfiguration( )** function.

Table 7. New Alarms for E1 Technology (DM3 Boards)

Alarm	Meaning	Default Threshold Value	Range
DTE1_BPVS	Bipolar violation count saturation	255	0 to 255
DTE1_CECS	CRC4 error count saturation	255	0 to 255
DTE1_ECS	Frame sync bit error count saturation	0	0 to 255

1.60.2 New T1 Alarms

The following table lists new alarms for T1 technology. These new alarms are non-blocking and not received by default. To change these default values, use the **gc_SetAlarmConfiguration( )** function.

Table 8. New Alarms for T1 Technology (DM3 Boards)

Alarm	Meaning	Default Threshold Value	Range
DTT1_BPVS	Bipolar violation count saturation	255	0 to 255
DTT1_ECS	Frame bit error count saturation	0	0 to 255
DTT1_FERR	Two out of four consecutive frame bits (F bit) in error	0	0 to 255
DTT1_OOF	Out of frame error count saturation	0	0 to 255

1.60.3 Modifying Default Threshold Values for New Alarms

If desired, you can change the default threshold value of a new T1/E1 alarm by adding a parameter in the CONFIG file that corresponds to the PCD file in use on your board. The change is made per span. After threshold parameters are added, generate an updated FCD and start system services on the board. If threshold parameters are not added, default threshold values are in effect. See the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* for more information on CONFIG files, PCD files, and FCD files.

To modify default threshold values for E1 alarms, add one or more of the following parameters (sample value of 100 shown) to the [lineAdmin.x] section of a CONFIG file:

```
SetParm=0x1639,100 ! BPVS threshold range 0 - 255, default 255
SetParm=0x163c,100 ! ECS threshold range 0 - 255, default 0
SetParm=0x163d,100 ! CECS threshold range 0 - 255, default 255
```

To modify default threshold values for T1 alarms, add one or more of the following parameters (sample value of 100 shown) to the [lineAdmin.x] section of a CONFIG file:

```
SetParm=0x1639,100 ! BPVS threshold range 0 - 255, default 255
SetParm=0x163a,100 ! OOF threshold range 0 - 255, default 0
SetParm=0x163b,100 ! FERR threshold range 0 - 255, default 0
SetParm=0x163c,100 ! ECS threshold range 0 - 255, default 0
```

1.60.4 Support for New Alarms in GCAMS Functions

The new alarms are supported by the GCAMS API functions as documented in the *Dialogic® Global Call API Library Reference* in the Dialogic® System Release 6.0 PCI for Windows® bookshelf.

1.61 Telecom Subsystem Summary Tool (its_sysinfo)

The Telecom Subsystem Summary Tool (its_sysinfo) provides a simple way to collect information about systems built using Dialogic® products. The its_sysinfo tool collects data from the system on which you execute it and provides information about the system environment: the operating system, computer architecture, System Release software, and operational logs.

With the Service Update, the *its_sysinfo.htm* file now includes a Windows® Package Info section at the beginning of the file. For example:

```
                              WindowsPackageInfo

Active System Release

Dialogic(R) System Release 6.0 PCI for Windows Build 125 (System Release)

Build Type: System Release
Install Location: C:\Program Files\Dialogic
Install Date: 2-20-2007 at 15:52:30
Installed By: Computing Customer

Installed Features
     Devel
     Runtime

Previously Installed System Release

Dialogic(R) System Release 6.0 PCI for Windows Build 123 (System Release)

Build Type: System Release
Install Location: C:\Program Files\Dialogic
Install Date: 1-15-2007 at 15:29:40
Installed By: Computing Customer

Installed Features
     Devel
     Runtime
```

For detailed information about the its_sysinfo tool, see the *Dialogic® System Software Diagnostic Guide*.

1.62 Windows® Hardware Quality Labs (WHQL) Certification

Note: WHQL certification for Dialogic® System Release 6.0 PCI for Windows® Service Update is not currently valid. The product is getting recertified.

1.63 Single Echo Canceller Convergence

The Service Update allows you to set single echo canceller convergence on Dialogic® DMV160LP Media Boards, which reduces the number of false barge-ins and incorrect speech recognitions occurring in speech-enabled applications. A new channel parameter, ECCH_CONVERGE, provides this capability. Use **ec_setparm( )** with the ECCH_CONVERGE channel parameter to switch from continuous to single echo canceller convergence.

Dialogic® D/41JCT-LS Media Boards, and all other JCT Boards that use a continuous speech processing firmware load, function with single echo canceller convergence by default. Therefore it is not necessary to use the ECCH_CONVERGE channel parameter with these boards.

1.63.1 Feature Description

Speech-enabled applications that re-enable continuous speech processing and have loud prompts experience bursts of excessive echo in the streamed audio on each play file, causing application malfunction. These excessive echo bursts are caused by continuous echo canceller (EC) convergence, which is re-initialized and re-converges on each new **ec_stream( )** function. This results in several hundred milliseconds of excessive play echo at the beginning of each play file, which confuses the host-based recognizer, degrading the operation of the application system.

Single EC convergence can help with this problem. With single EC convergence, the addressed echo canceller will converge once, after the first **ec_stream( )** function is issued, and from then on the convergence coefficients are saved for the subsequent **ec_stream( )** functions. The echo canceller should be set to re-converge on the first call to **ec_stream( )** of each new phone call. This provides consistent echo cancellation and optimized barge-in performance.

The following sections describe how single EC convergence works on DMV160LP and JCT Boards.

1.63.1.1 Single EC Convergence for DMV160LP Boards

Using the new channel parameter, ECCH_CONVERGE, you can switch from continuous to single EC convergence. ECCH_CONVERGE can take a value of ON or OFF. The default value is ON (continuous EC convergence), which means that the echo canceller will re-converge, or retrain, on every new call to **ec_stream( )**. The ECCH_CONVERGE parameter supports applications that issue a new **ec_stream( )** function with each play file. This echo convergence mode setting is unaffected by the **ec_stopch( )** function.

Note: These parameter operations cannot be issued when any voice I/O function is active on that channel. A TDX_BUSY error will be returned if it is attempted.

Single EC convergence is set for DMV160LP Boards as follows:

1. At the beginning of each new phone call, the application must set ECCH_CONVERGE to ON to allow the echo canceller to adapt to the connected trunk's characteristics.

2. Immediately after the first **ec_stream( )** for the phone call, set the ECCH_CONVERGE to OFF.

3. Reset ECCH_CONVERGE to ON for the next phone call.

4. After the first **ec_stream( )**, set ECCH_CONVERGE to OFF.

5. Repeat for each phone call.

Note: Once the **ec_setparm( )** is issued with the ECCH_CONVERGE value set to OFF, the addressed voice channel's echo canceller will no longer track changes in the trunk characteristics. At the beginning of each new phone call, the application must set ECCH_CONVERGE to ON to allow the echo canceller to adapt to the connected trunk's characteristics. Failure to do this can result in poor echo cancellation, which affects important voice channel functions.

1.63.1.2 Single EC Convergence for JCT Boards

Dialogic® D/41JCT-LS Media Boards, and all other JCT boards using a continuous speech processing firmware load, function with single echo canceller convergence by default. Therefore it is not necessary to use the ECCH_CONVERGE channel parameter to obtain single EC convergence. The equivalent to setting the ECCH_CONVERGE parameter to ON for JCT Boards is to reset the echo canceller prior to every call to **ec_stream( )** by setting the DXCH_EC_TAP_LENGTH parameters.

Note: Calling the ECCH_CONVERGE parameter with JCT Boards will return an error indicating that this function is not supported.

JCT Boards will function with single EC convergence if you set all the CSP parameters once at the beginning of each new phone call prior to the first call to **ec_stream( )**.

If you need continuous EC convergence, set the parameter DXCH_EC_TAP_LENGTH prior to each call to **ec_stream( )**. Setting this parameter resets the echo canceller and forces the echo canceller to reconverge.

1.63.2 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about **ec_stream( )**, **ec_setparm( )**, **ec_stopch( )**, and other Dialogic® Continuous Speech Processing API functions and parameters, refer to the following documents:

- *Dialogic® Continuous Speech Processing API Programming Guide*
- *Dialogic® Continuous Speech Processing API Library Reference*

Note: The online bookshelf has not been updated for this feature, so the Continuous Speech Processing API documentation does not currently include information about the ECCH_CONVERGE parameter.

1.64 New Features in Dialogic® Global Call Protocols Package

A number of new features have been added to the Dialogic® Global Call Protocols Package, which is now part of the System Release software.

The following new protocols are supported:

- Bulgaria R2
- Croatia R2
- Kuwait R2
- Lithuania R2
- Uzbekistan R2
- Korea T1/R2
- Lebanon R2
- Poland R2
- Samsung PBX Lineside E1

There are also enhancements to existing protocols:

New parameters for Nortel Meridian Lineside E1 protocol
: New parameters have been added to specify whether the protocol will wait for IDLE, wait for ReleaseGuard, and wait for SEIZEACK.

Send blocking pattern when channel is put OOS
: A new parameter, CDP_BlockOnLOOS, has been added to the CDP files for several protocols to send a blocking pattern when a channel is put out-of-service. The protocols with this new parameter are:
    - Alcatel 4400 Lineside E1
    - Alcatel VPS 4x00 Lineside
    - Ericsson MD110 PBX Lineside E1
    - Korea GDS Lineside E1
    - Lucent Lineside E1
    - NEC Lineside E1
    - Nortel Meridian Lineside E1
    - T1 FXS Ground Start
    - United States T1 FXS/LS

Call transfer functionality
: The ability to transfer calls on switches using MELCAS Lineside protocol is now supported.

An updated version of the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* providing more detailed information about these new features has been added to the documentation bookshelf.

1.65 New Operating System Support

In addition to the supported operating systems listed in the Release Guide, the following operating system versions are now supported with this Service Update:

- Windows Vista® Business, 32-bit edition
- Windows Vista® Enterprise, 32-bit edition
- Windows Vista® Ultimate, 32-bit edition

  Note: See Section 1.4, "Support for Windows Vista® Operating System", on page 40 for information about running System Release 6.0 PCI Windows on Windows Vista, as opposed to running on other Windows® operating systems.
- Windows® XP SP2
- Windows Server® 2003 SP1 and SP2
- Windows Server® 2003 R2
- Windows Server® 2003 R2 SP2
- Windows® 2000 Update Rollup 1 for SP4

Note: Terminal Services Application Server Mode and Active Directory Application Server Mode are not supported on any operating systems.

1.66 New Station Interface Alarms

The Service Update provides the ability to monitor the communication links between a board and its associated Station Interface Box (SIB). If power to the SIB is lost or if any communication links between the board and the SIB are accidentally disconnected (e.g., cable is disconnected), an alarm event is sent to the application. With this new alarm event, the application can now be notified when a station interface is not online, so the application can stop sending calls to station interfaces that are no longer in service. The application can also be notified when the problem is corrected.

This feature is applicable to the following boards:

- Dialogic® HDSI/480, HDSI/720, HDSI/960, and HDSI/1200 Station Interface Boards
- Dialogic® DI0408LSAR2 Switching Boards
- Dialogic® DISI16R2, DISI24R2, and DISI32R2 Switching Boards

1.66.1 Feature Description

A new asynchronous event, MSEV_CHANSTATE, has been added to the Dialogic® Modular Station Interface (MSI) API. The event data for MSEV_CHANSTATE is:

MSMM_CS_ALARM
: Station interface failure, e.g., communication link disconnected.

MSMM_CS_IDLE
: Station interface online; sent when cable is reconnected, alarm is cleared, or station is powered up.

MSMM_CS_OUT_OF_SERVICE
: Loop current to station interface disabled, e.g., for maintenance purposes.

The MSEV_CHANSTATE event is disabled by default. Use the **ms_setevtmsk( )** function to enable events, for example, **ms_setevtmsk(msiB#C#, MSEV_CHANSTATE, MSMM_CS_ALARM | MSMM_CS_IDLE | MSMM_CS_OUT_OF_SERVICE, DTA_SETMSK)**.

Note: The enabling/disabling of the event is local to a process. If multiple processes are running on the same board, the event has to be enabled for each process.

Alarms are provided on a per station basis as opposed to a per link basis. This means that if a communication link is disconnected on an HDSI Board, for example, 30 alarms corresponding to the 30 stations on that link would be sent. It is up to the application to enable/mask the appropriate alarms to arrive at the desired number of alarms per link. The number of stations per link depends on the board you are using; for example, on the HDSI/960 Station Interface Board, link 1 = stations 1-30, link 2 = stations 31-60, link 3 = stations 61-90, and link 4 = stations 91-96.

Example

This example shows how to use the **ms_setevtmsk( )** function to enable a station to receive all three channel state events.

```
#include <msilib.h>

/* Enable channel state event */
void EnableCSEvents(int a_DevHdl)
{
     unsigned short t_SetBitMsk = MSMM_CS_ALARM | MSMM_CS_IDLE | MSMM_CS_OUT_OF_SERVICE;
     int t_Action = DTA_ADDMSK;
     unsigned short t_GetBitMsk = 0;
     if ( ms_setevtmsk(a_DevHdl, MSEV_CHANSTATE, t_SetBitMsk, t_Action) == -1 )
     {
          printf("ms_setevtmsk(dvh:%d, MSEV_CHANSTATE, bitmsk:0x%X, action:%d) failed \n",
a_DevHdl, t_SetBitMsk, t_Action);
          printf("Error Message = %s\n",ATDV_ERRMSGP(a_DevHdl) );
     }
     else
     {
          printf("ms_setevtmsk(dvh:%d, MSEV_CHANSTATE, bitmsk:0x%X, action:%d) success \n",
a_DevHdl, t_SetBitMsk, t_Action);
     }
     /* Verify the setting */
     if ( ms_getevtmsk(a_DevHdl, MSEV_CHANSTATE, &t_GetBitMsk) == -1 )
     {
          printf("ms_getevtmsk(dvh:%d, MSEV_CHANSTATE, bitmsk:0x%X) failed \n", a_DevHdl,
t_GetBitMsk);
          printf("Error Message = %s\n",ATDV_ERRMSGP(a_DevHdl) );
     }
     else
     {
          if ( (t_GetBitMsk & MSMM_CS_ALARM) == MSMM_CS_ALARM )
               printf("ms_getevtmsk(dvh:%d, MSEV_CHANSTATE, bitmsk:0x%X) - MSMM_CS_ALARM is
set\n", a_DevHdl, t_GetBitMsk);
```

```
          if( (t_GetBitMsk & MSMM_CS_IDLE) == MSMM_CS_IDLE )
               printf("ms_getevtmsk(dvh:%d, MSEV_CHANSTATE, bitmsk:0x%X) - MSMM_CS_IDLE is 
set\n", a_DevHdl, t_GetBitMsk);
          if ( (t_GetBitMsk & MSMM_CS_OUT_OF_SERVICE) == MSMM_CS_OUT_OF_SERVICE )
              printf("ms_getevtmsk(dvh:%d, MSEV_CHANSTATE, bitmsk:0x%X) - MSMM_CS_OUT_OFSERVICE 
is set\n", a_DevHdl, t_GetBitMsk);
     }
}

/* Process the SRL event */
long EventHandler (unsigned long temp)
{
     int devh = sr_getevtdev();
     long event = sr_getevttype();
     unsigned short * evtdata = (unsigned short*) sr_getevtdatap();
     switch(event)
     {

/*… */

          case MSEV_CHANSTATE :
               switch(*evtdata)
               {
                    case MSMM_CS_ALARM :
                         printf("MSEV_CHANSTATE(MSMM_CS_ALARM) is detected on devh:%d \n", 
devh);
                         break;
                    case MSMM_CS_IDLE :
                         printf("MSEV_CHANSTATE(MSMM_CS_IDLE) is detected on devh:%d\n", devh);
                         break;
                    case MSMM_CS_OUT_OF_SERVICE :
                         printf("MSEV_CHANSTATE(MSMM_CS_OUT_OF_SERVICE) is detected on 
devh:%d\n", devh);
                         break;
                    default:
                        printf("**Unknown EventData received...MSEV_CHANSTATE(Eventdata = 0x%x) 
on %d**\n", *evtdata, devh);
                         break;
               }     /* switch event data ends */
          break;
     }
     return 0;
}
```

1.66.2 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the Dialogic® MSI API, see the following documents:

- *Dialogic® Modular Station Interface API Programming Guide*
- *Dialogic® Modular Station Interface API Library Reference*

Note: The online bookshelf has not been updated for this feature. The following observations are worth noting:

- The descriptions of the **ms_setevtmsk( )** and **ms_getevtmsk( )** functions in the *Dialogic® Modular Station Interface API Library Reference* do not currently include information about the MSEV_CHANSTATE event.

- The Events chapter in the *Dialogic® Modular Station Interface API Library Reference*, and the Event Handling chapter in the *Dialogic® Modular Station Interface API Programming Guide*, do not currently include the MSEV_CHANSTATE event.

1.67 Support for ANI Category Digit Retrieval on Dialogic® DM3 Boards

The Service Update provides support for ANI category digit retrieval on Dialogic® DM3 Boards.

1.67.1 Feature Description

The **gc_GetCallInfo( )** function, which retrieves information associated with a call, can now be used to retrieve the category digit for DM3 Boards. Formerly, the **gc_GetCallInfo( )** CATEGORY_DIGIT parameter was supported on Dialogic® Springware Boards only.

The category digit is used to determine the origin or type of the calling party (for example, ordinary subscriber, pay phone) so that the application may choose to take a specific action based on the call's origin. The category digit is used mainly with E1 R2MF protocols, and the categories are determined by the protocol.

1.67.2 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about **gc_GetCallInfo( )** and other Dialogic® Global Call API functions, see the following documents:

- *Dialogic® Global Call API Programming Guide*
- *Dialogic® Global Call API Library Reference*

For features specific to E1 technology, see:

- *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*

Note: The online bookshelf has not been updated for this feature, so the *Dialogic® Global Call API Library Reference* and *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide* do not currently indicate that the CATEGORY_DIGIT parameter is supported on DM3 Boards.

1.68 New Media Load for Dialogic® DMV3600BP Boards

The Service Update provides a new media load, ML9B-LC, for the Dialogic® DMV3600BP Media Board. This new media load is a conferencing only media load, supporting large conferences. It provides 128 conferencing resources with echo cancellation and tone clamping. The maximum conference size without bridging is 64 parties. The maximum conference size with bridging is 126 parties per board.

Media load ML9B-LC is an addition to the ML9x series of media loads for conferencing. For example, media load ML9B can still be used for applications that require higher overall density without the need for large conference sizes. The conferencing features available with media load ML9B are: 160 conferencing resources with echo cancellation and tone clamping, maximum conference size of 16 parties without bridging, and maximum conference size of 142 parties with bridging.

1.68.1 Feature Description

Predefined sets of features for Dialogic® Boards are provided in media loads. A media load consists of a configuration file set (PCD, FCD, and CONFIG files) and the associated firmware that is downloaded to the board. See the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* for more information about media loads.

The features and channel densities provided by the new media load ML9B-LC for the DMV3600BP Board are as follows:

Features Supported	Rich Conferencing with Echo Cancellation and Tone Clamping	Maximum Conference Size without Bridging	Maximum Conference Size per Board with Bridging
Channel Density	128	64	126

For information about bridging, see the Conference Bridging chapter in the *Dialogic® Audio Conferencing API Programming Guide*.

Note: Voice resources are **not** included in the ML9x media loads.

1.68.2 Configuring the Software

The new media load can be selected by using the Dialogic® Configuration Manager (DCM). This procedure, which must be performed before the boards are started, is described in detail in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

1.68.3 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For detailed information about configuring DMV3600BP Boards, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Note: The online bookshelf has not been updated for this feature, so the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* does not currently include information about media load ML9B-LC.

For information about the Dialogic® Audio Conferencing (DCB) API, see the following documents:

- *Dialogic® Audio Conferencing API Programming Guide*
- *Dialogic® Audio Conferencing API Library Reference*

1.69 New Media Loads for Dialogic® DMV1200BTEP Boards

The Service Update provides new media loads for the Dialogic® DMV1200BTEP Media Board:

QSB-U3
: Provides rich conferencing (conferencing, echo cancellation, and tone clamping) with enhanced voice, FSK, transaction record, and fax.

QSB-ML10
: Provides rich conferencing (conferencing, echo cancellation, and tone clamping) with enhanced voice, FSK, and transaction record.

QSB-ML10-LC
: Provides rich conferencing (conferencing, echo cancellation, and tone clamping) with enhanced voice, FSK, and transaction record. Supports larger conferences than QSB-ML10.

QSB-U2
: Provides increased density for standard conferencing while also providing basic voice, FSK, and fax.

10b
: Provides rich conferencing (conferencing, echo cancellation, and tone clamping) while also providing full density basic voice with transaction record and FSK.

Note: For information about basic voice features and enhanced voice features, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

1.69.1 Feature Description

Predefined sets of features for Dialogic® Boards are provided in media loads. A media load consists of a configuration file set (PCD, FCD, and CONFIG files) and the associated firmware that is downloaded to the board. See the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* for more information about media loads.

Media Load QSB-U3

The features and channel densities provided by media load QSB-U3 are as follows:

Features Supported	Enhanced Voice	Transaction Record	FSK	Fax	Conferencing with Echo Cancellation and Tone Clamping
Channel Density	120	120	120	8	36

There are 120 total voice resources. Any combination of the voice features (enhanced voice, transaction record, and FSK) can be used up to a total of 120. In addition to these voice resources, 36 conferencing resources (with echo cancellation and tone clamping) and 8 fax resources can be used.

Notes: ***1.*** Conference size is limited to 18 parties without bridging. Conference bridging can be used to effectively expand a conference beyond the maximum size. Conference bridging consumes conferencing resources, reducing overall board conference density.

2. Although it is usually part of the enhanced voice media load features, TrueSpeech is not supported with media load QSB-U3.

3. QSB-U3 no longer supports CSP streaming to CT Bus. This frees up CT Bus time slots that can be allocated to other boards, allowing for higher system density. Applications that require CSP streaming to CT Bus can use QSB-ML10 or QSB-U1.

Media load QSB-U3 can be used with all protocols supported on the DMV1200BTEP Board, e.g., T1 ISDN, T1 CAS, E1 ISDN, E1 R2MF, and DPNSS/DASS2.

Media Load QSB-ML10

The features and channel densities provided by media load QSB-ML10 are as follows:

Features Supported	Enhanced Voice	Transaction Record	FSK	Conferencing with Echo Cancellation and Tone Clamping
Channel Density	120	120	120	54

There are 120 total voice resources. Any combination of the voice features (enhanced voice, transaction record, and FSK) can be used up to a total of 120. In addition to these voice resources, 54 conferencing resources (with echo cancellation and tone clamping) can be used.

Notes: ***1.*** Conference size is limited to 18 parties without bridging. Conference bridging can be used to effectively expand a conference beyond the maximum size. Conference bridging consumes conferencing resources, reducing overall board conference density.

2. Although it is usually part of the enhanced voice media load features, TrueSpeech is not supported with media load QSB-ML10.

Media load QSB-ML10 can be used with all protocols supported on the DMV1200BTEP Board, e.g., T1 ISDN, T1 CAS, E1 ISDN, E1 R2MF, and DPNSS/DASS2.

Media Load QSB-ML10-LC

Media Load QSB-ML10-LC is similar to QSB-ML10 but with larger conference sizes (38 parties instead of 18 parties without bridging). QSB-ML10 can still be used for applications that don't need larger conference sizes. To allow for higher system density, QSB-ML10-LC does not support CSP streaming to CT Bus.

The features and channel densities provided by media load QSB-ML10-LC are as follows:

Features Supported	Enhanced Voice	Transaction Record	FSK	Conferencing with Echo Cancellation and Tone Clamping
Channel Density	120	120	120	38

There are 120 total voice resources. Any combination of the voice features (enhanced voice, transaction record, and FSK) can be used up to a total of 120. In addition to these voice resources, 38 conferencing resources (with echo cancellation and tone clamping) can be used.

Notes: ***1.*** Conference size is 38 parties without bridging.

2. Although it is usually part of the enhanced voice media load features, TrueSpeech is not supported with media load QSB-ML10-LC.

Media load QSB-ML10-LC can be used with all protocols supported on the DMV1200BTEP Board, e.g., T1 ISDN, T1 CAS, E1 ISDN, E1 R2MF, and DPNSS/DASS2.

Media Load QSB-U2

The features and channel densities provided by media load QSB-U2 are as follows:

Features Supported	Basic Voice - FSK	Fax	Conferencing - Tone Clamping
Channel Density	120	12	120

Note: Echo cancellation is not supported with media load QSB-U2 and should not be enabled by the application.

Media load QSB-U2 can be used with all protocols supported on the DMV1200BTEP Board, e.g., T1 ISDN, T1 CAS, E1 ISDN, E1 R2MF, and DPNSS/DASS2.

Media Load 10b

The features and channel densities provided by media load 10b are as follows:

Features Supported	Basic Voice	Transaction Record	FSK	Conferencing with Echo Cancellation and Tone Clamping
Channel Density	120	120	120	120

There are 120 total voice resources. Any combination of the voice features (basic voice, transaction record, and FSK) can be used up to a total of 120. In addition to these voice resources, 120 conferencing resources (with echo cancellation and tone clamping) can be used.

Note: Conference size is limited to 20 parties without bridging. Conference bridging can be used to effectively expand a conference beyond the maximum size. Conference bridging consumes conferencing resources, reducing overall board conference density.

Media load 10b can be used with all protocols supported on the DMV1200BTEP Board, e.g., T1 ISDN, T1 CAS, E1 ISDN, E1 R2MF, and DPNSS/DASS2.

1.69.2 Configuring the Software

The new media loads can be selected by using the Dialogic® Configuration Manager (DCM). For DMV1200BTEP Boards, the **MediaLoad** parameter appears on the Trunk Configuration property sheet.

In addition to specifying the media load, the Trunk Configuration property sheet allows you to individually configure network trunks on the DMV1200BTEP Board with different T1 or E1 protocols. Based on your selections on this property sheet, DCM creates a composite configuration file set (PCD, FCD, and CONFIG files). This procedure, which must be performed before the board is started, is described in detail in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide.*

1.69.3 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For detailed information about configuring DMV1200BTEP Boards, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Note: The online bookshelf has not been updated for this feature, so the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* does not currently include information about media loads QSB-U3, QSB-ML10, QSB-U2, and 10b.

1.70 New Media Load for Dialogic® DMV600BTEP Boards

The Service Update provides a new media load, DSB-U2, for the Dialogic® DMV600BTEP Media Board. This new media load provides rich conferencing (conferencing, echo cancellation, and tone clamping) with enhanced voice, FSK, transaction record, and fax.

1.70.1 Feature Description

Predefined sets of features for Dialogic® Boards are provided in media loads. A media load consists of a configuration file set (PCD, FCD, and CONFIG files) and the associated firmware that is downloaded to the board. See the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* for more information about media loads.

The features and channel densities provided by media load DSB-U2 are as follows:

Features Supported	Enhanced Voice	Transaction Record	FSK	Fax	Conferencing with Echo Cancellation and Tone Clamping
Channel Density	90	90	90	6	48

There are 90 total voice resources. Any combination of the voice features (enhanced voice, transaction record, and FSK) can be used up to a total of 90. In addition to these voice resources, 48 conferencing resources (with echo cancellation and tone clamping) and 6 fax resources can be used.

Note: Conference size is limited to 16 parties without bridging. Conference bridging can be used to effectively expand a conference beyond the maximum size. Conference bridging consumes conferencing resources, reducing overall board conference density.

Media load DSB-U2 can be used with all protocols supported on the DMV600BTEP Board, e.g., T1 ISDN, T1 CAS, E1 ISDN, E1 R2MF, and DPNSS/DASS2.

1.70.2 Configuring the Software

The new media load can be selected by using the Dialogic® Configuration Manager (DCM). For DMV600BTEP Boards, the **MediaLoad** parameter appears on the Trunk Configuration property sheet.

In addition to specifying the media load, the Trunk Configuration property sheet allows you to individually configure network trunks on the DMV600BTEP Board with different T1 or E1 protocols. Based on your selections on this property sheet, DCM creates a composite configuration file set (PCD, FCD, and CONFIG files). This procedure, which must be performed before the board is started, is described in detail in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide.*

1.70.3 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For detailed information about configuring DMV600BTEP Boards, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Note: The online bookshelf has not been updated for this feature, so the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* does not currently include information about media load DSB-U2.

1.71 Call Transfer Support on the Dialogic® DMV160LP Board

The Service Update adds support for supervised call transfer on the Dialogic® DMV160LP Media Board.

1.71.1 Feature Description

Supervised call transfer is a feature that enables a controller (party A) already in a call with another party (party B) to transfer the call to a third party (party C). The end result is a call between party B and party C. This feature is a common requirement in IVR and voicemail applications.

1.71.1.1 Basic Call Transfer Scenario

The sequence of events in a supervised call transfer scenario is described below. It is assumed that party A and party B are already in a call.

1. Party A hookflashes party B, placing the call with party B on hold. This call is referred to as the “held” call.

2. Party A dials party C and waits for an answer.

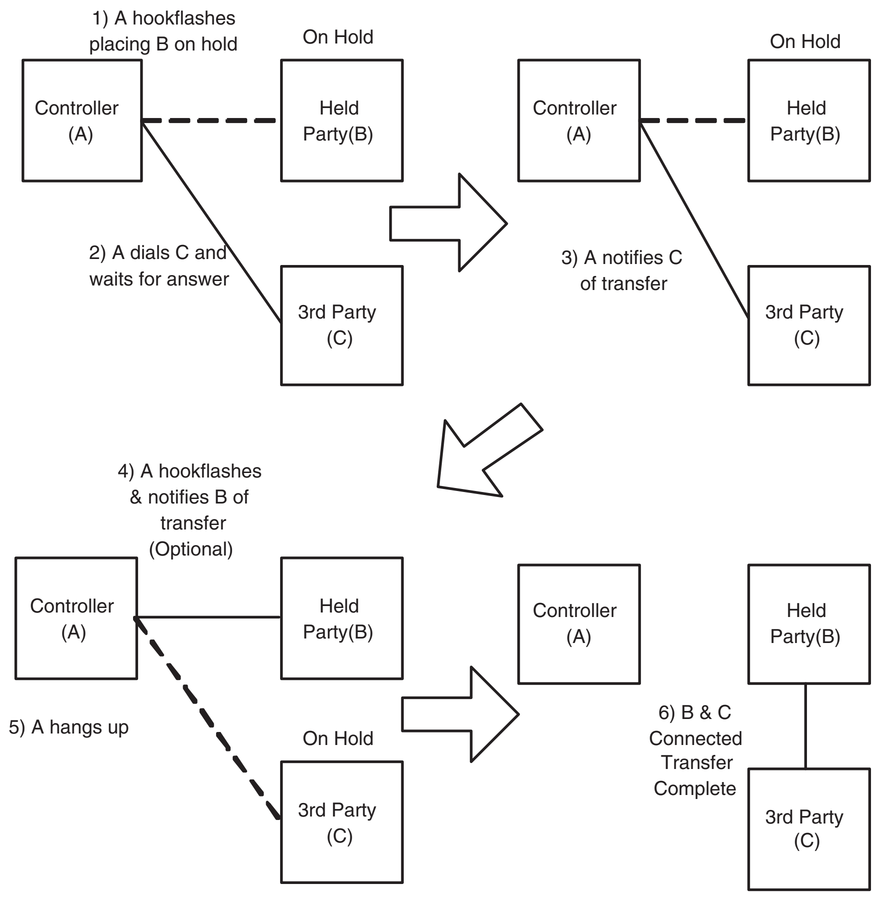
3. Party A notifies party C that the transfer is about to take place. This call is referred to as the “consultation” call.

4. Optionally, party A hookflashes party C and notifies party B of the transfer.

5. Party A hangs up.

6. Parties B and C are connected and the transfer is completed.

The sequence is shown diagrammatically in the following figure.



1.71.1.2 Call Transfer APIs

The supervised call transfer feature is provided by the following Dialogic® Global Call API functions:

gc_SetupTransfer( )
: initiates a supervised call transfer and allocates a CRN for the consultation call

gc_MakeCall( )
: used to make the consultation call

gc_CompleteTransfer( )
: used to complete the transfer and communicate to the CPE/CO equipment to connect the talk paths of the held call and the consultation call

gc_SwapHold( )
: communicates to the CPE/CO equipment that the talk path to the controller should be "swapped" from the held call to the consultation call. This allows the controller to swap

between the Held party and the Third party prior to the transfer. Once this API is completed, the roles of the held and consultation call are reversed.

Note: Depending on the PBX type and configuration, it may not be possible to use the **gc_SwapHold( )** function to swap between the held call and the consultation call. For non-US protocols, the **gc_SwapHold( )** function can operate correctly if the behavior of the protocol is similar to that of a US counterpart.

1.71.1.3 Application Development Notes

The following application development notes apply:

- When any of the parties involved in a transfer are dropped or remotely disconnected prior to calling **gc_CompleteTransfer( )**, all active calls (both consultation and held calls) must be dropped using **gc_DropCall( )** and the CRNs must be released using **gc_ReleaseCallEx( )**.
- The **gc_ResetLineDevice( )** function can be used to reset a channel and terminate all active calls when a transfer call scenario is active.
- When setting up a supervised call transfer, after the **gc_SetupTransfer( )** function is issued to obtain a CRN for the consultation call, a permanent signal timer (8 seconds) starts. If the consultation call is not made within the 8 second period, the timer expires and the application receives a GCEV_DISCONNECTED event.

1.71.1.4 PBX Testing

Note: The call transfer feature has been tested on PBX systems that have been configured to use US protocols only.

The basic call transfer scenario as described above has been tested on the following PBX systems:

- Siemens HiCom 150E Office Pro
- Mitel SX 200
- Ericsson MD110
- Alcatel Omni PCX 4400
- Panasonic Easa-Phone KX T30810
- NEC 2400

For the Siemens HiCom 150E, the following variations in the basic call transfer scenario have also been tested:

- The controller drops the consultation call before dialing is started
  - Party B calls party A
  - Party A hookflashes and then drops the call
  - Verify: Party B is connected back to Party A
- Blind transfer
  - Party B calls party A
  - Party A hookflashes (places call with party B on hold)

- Party A dials party C, then hangs up
- Verify: Party B and party C connected

- The held call is dropped by Party B
  - Party B calls party A
  - Party A hookflashes (places call with party B on hold)
  - Party A calls party C (consultation call)
  - Party C picks up
  - Party B hangs up
  - Party C hookflashes
  - Verify: Party C and party A connected
- The consultation call is dropped by Party C
  - Party B calls party A
  - Party A hookflashes
  - Party B calls party C
  - Party C hangs up
  - Verify: Party B and Party A are connected again

1.71.1.5 PBX Integration Issues

From a PBX perspective, call transfer is most often a sub-feature of Multi-Way Calling (MWC). MWC provides for several variations of conference call capability. Conference features are usually accessed via a flash hook followed by the dialing of an access code. The variation in the behavior of conference features needs to be taken into account when integrating a CT application on a PBX.

The following behavior needs to be considered:

Swapping between held and consultation calls
: If the PBX has conference capability enabled, issuing a second **gc_SwapHold( )** could cause a three-way call to be created. This call scenario can no longer be considered a call transfer scenario.

Remote party drop of consultation call
: There are a number of possible behaviors in this scenario. These include:
  - getting disconnect treatment on party C, or
  - automatically having the talk path connected back to party A (the held call)

Initiating a call transfer (using **gc_SetupTransfer( )**), then releasing the consultation call prior to issuing **gc_MakeCall( )**
: Various types of "ring-back" treatments can be applied by the PBX. A "ring-back" treatment occurs when the PBX generates the ring voltage on any of the parties involved in the transfer (or conference). The duration of the generated ring can be from 1 ring (approximately 6 seconds) to 6 rings (approximately 36 seconds).

1.71.2 Configuring the Software

The following configuration instructions apply when using this release update software with PBX systems:

- Updating the CONFIG File

- Running PBX Expert

1.71.2.1 Updating the CONFIG File

The following parameters require configuration in the DMV160LP Board CONFIG file:

- **Tone_SigId4** (Disconnect Tone Supervision) must be set to a value of 238113 (a fixed tone ID) to enable disconnect tone supervision. The default value is 0x0 (disabled).
- **BtStartTimeout** (Permanent Signal Planning) must be set to a value appropriate for the PBX system being used. The default is 8000 (8 seconds). This value may need to be changed if the PBX system has a shorter timeout prior to the start of the consultation call.

Whenever a CONFIG file has been modified, a new FCD file must be generated. This procedure is described in detail in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

1.71.2.2 Running PBX Expert

Use the PBX Expert utility (accessible from the Windows® Start menu) to detect and learn call progress tones.

1.71.3 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about generic Dialogic® Global Call API features, see the following documents:

- *Dialogic® Global Call API Programming Guide*
- *Dialogic® Global Call API Library Reference*

For features specific to Analog technology, see:

- *Dialogic® Global Call Analog Technology Guide*

Note: The online bookshelf has not been updated for this feature. The following observations are worth noting:

- The *Dialogic® Global Call API Library Reference* currently shows the call transfer functions, **gc_SetupTransfer( )**, **gc_CompleteTransfer( )**, and **gc_SwapHold( )** as not supported for Analog technology.
- The *Dialogic® Global Call Analog Technology Guide* does not provide any information about “call transfer” since this feature is not supported on any boards that include analog interfaces other than the DMV160LP Board as described here.

1.72 dx_reciottdata( ) Enhancements

The Service Update provides the following enhancements to the **dx_reciottdata( )** function:

- Initial silence compression
- Voice activity detector (VAD) with event notification

These enhancements are applicable to the following boards:

- Dialogic® DM/V, DM/V-A, DM/V-B, DM/VF, and DMV160LP Media Boards
- Dialogic® DM/IP Boards

1.72.1 Feature Description

The **dx_reciottdata( )** function, used to record voice data, has two new modes:

RM_VADNOTIFY
: generates an event on detection of VAD during the recording operation. The new event is TDX_VAD.

  Note: TDX_VAD is not an indication of function termination; it is an unsolicited event.

RM_ISCR
: adds initial silence compression to the VAD detection capability.

  Note: The RM_ISCR mode can only be used in conjunction with RM_VADNOTIFY.

To enable these modes, OR them to the **dx_reciottdata( )** function mode parameter. For example:

```
t_Return=dx_reciottdata(DevHandle, Iott, Tpt, &t_Xpb, EV_ASYNC|RM_VADNOTIFY);

t_Return=dx_reciottdata(DevHandle, Iott, Tpt, &t_Xpb, EV_ASYNC|RM_VADNOTIFY|RM_ISCR);
```

When these two modes are used together, no data is recorded as output until voice activity is detected on the line. The TDX_VAD event indicates the initiation of voice. The output file will be empty before the VAD is detected, although some initial silence may be included as specified in the FCD file.

Initial silence is the amount of silence on the line before VAD is detected. When using RM_ISCR, the default value for the amount of allowable silence is 3 seconds. Any initial silence longer than that will be truncated. This default value can be changed by modifying a parameter in the CONFIG file for the board and then generating a new FCD file. See Section 1.72.2, “Configuring the Software”, on page 255.

Supported Coders

These enhancements to the **dx_reciottdata( )** function are supported for the following encoding methods and sampling rates:

- OKI ADPCM, 6 kHz with 4-bit samples (24 kbps) and 8 kHz with 4-bit samples (32 kbps), VOX and WAVE file formats
- Linear PCM, 8 kHz sampling 64 Kbps (8 bits), 8 kHz sampling 128 Kbps (16 bits)
- G.711 PCM, 6 kHz with 8-bit samples (48 kbps) and 8 kHz with 8-bit samples (64 kbps) using A-law or mu-law coding, VOX and WAVE file formats
- G.721 at 8 kHz with 4-bit samples (32 kbps), VOX and WAVE file formats
- G.726 bit-exact voice coder at 8 kHz with 2-, 3-, 4-, or 5-bit samples (16, 24, 32, 40 kbps), VOX and WAVE file formats

1.72.2 Configuring the Software

Predefined sets of features for Dialogic® Boards are provided in media loads. A media load consists of a configuration file set (PCD, FCD, and CONFIG files) and the associated firmware that is downloaded to the board. See the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* for more information about media loads.

The initial silence compression and VAD event notification features are available in any media load that has voice functionality.

To change the default value for the amount of allowable silence when using RM_ISCR, you must add a new parameter in the CONFIG file that was selected for your board. The parameter is **0x416**, and must be added in the [encoder] section of the config file. The initial silence value for the parameter is specified directly in seconds, for example:

```
[encoder]
SetParm=0x416,6
```

This sets the maximum amount of allowable silence to 6 seconds. Any initial silence longer than that will be truncated.

Whenever a CONFIG file has been modified, a new FCD file must be generated. This procedure is described in detail in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

1.72.3 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the Dialogic® Voice API, see the following documents:

- *Dialogic® Voice API Programming Guide*

- *Dialogic® Voice API Library Reference*

Note: The online bookshelf has not been updated for this feature. The following observations are worth noting:

- The description of the **dx_reciottdata( )** function in the *Dialogic® Voice API Library Reference* does not currently show the RM_ISCR and RM_VADNOTIFY modes.
- The Events chapter in the *Dialogic® Voice API Library Reference* does not currently include the TDX_VAD event.
- The *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* does not currently include information about the **0x416** parameter.

1.73 Trunk Preconditioning

The Service Update provides the capability for trunk preconditioning, which allows boards to be placed in an alarm state during board initialization.

These enhancements are applicable to the following boards:

- Dialogic® DM/V, DM/V-A, DM/V-B, and DM/VF Media Boards
- Dialogic® DM/N Digital Telephony Interface Boards
- Dialogic® DM/IP Boards

1.73.1 Feature Description

While Dialogic® Boards are starting up and are connected to network trunks, there is a period where the digital network interface begins transmitting frames and idle CAS signaling. This state can exist for a minute or more before the board and application program are prepared to handle calls. During this time, a service provider (CO) may begin alerting (ringing) for inbound calls, but the calls cannot be answered because the board or application has not finished initializing. This results in lost calls.

A new configuration parameter, referred to here as the Initial Alarm State parameter, allows you to place trunks in an alarm state while the board is being initialized. This prevents the service provider from sending calls. The alarm clears and the trunks go in-service as soon as the first **gc_OpenEx( )** (or **gc_Open( )**) function for a trunk is executed in the application. (For T1 trunks, alarms clear after a 15-second delay to verify valid signaling.)

The possible values for the new Initial Alarm State parameter are:

Value	Description
0	Default - No alarm is transmitted on the trunk; all trunk time slots signal Out of Service.
1	TransmitAIS - An Alarm Indication Signal (AIS) alarm is transmitted on the trunk.
2	TransmitRAI - A Remote Alarm Indication (RAI) alarm is transmitted on the trunk.

Note: The default behavior also applies if the Initial Alarm State parameter is not used. Behavior is the same in both ISDN and CAS environments.

For more detailed information about configuring the Initial Alarm State parameter, see Section 1.73.2, "Configuring the Software", on page 257.

The Initial Alarm State parameter setting applies only upon board initialization. After the initial alarm state is cleared (by **gc_OpenEx( )** or **gc_Open( )**), trunks do not return to the initial alarm state unless you restart the board. Stopping the board or unloading the application does not return a board to its initial alarm state.

Note: An RAI alarm could result from a response to a loss of sync from the network side. If the Initial Alarm State parameter is set to 2, but a loss of sync (or similar condition) persists even after the board is initialized and **gc_OpenEx( )** or **gc_Open( )** is invoked, the RAI will continue to be transmitted until the network condition is cleared.

A board could transmit other alarms, as a response to a network condition, that are unrelated to this parameter. Those alarms will persist until the network condition is cleared.

1.73.2 Configuring the Software

Predefined sets of features for Dialogic® Boards are provided in media loads. A media load consists of a configuration file set (PCD, FCD, and CONFIG files) and the associated firmware that is downloaded to the board. See the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* for more information about media loads.

In order to use the Initial Alarm State parameter, it must be manually added to the CONFIG file that was selected for your board. The hexadecimal parameter number is **0x1626**, and must be added in the [lineAdmin] section for each trunk. For example:

```
[lineAdmin.1]
SetParm=0x1626,1   ! InitialAlarmState (None=0, AIS=1, RAI=2)

[lineAdmin.2]
SetParm=0x1626,1   ! InitialAlarmState (None=0, AIS=1, RAI=2)

[lineAdmin.3]
SetParm=0x1626,1   ! InitialAlarmState (None=0, AIS=1, RAI=2)

[lineAdmin.4]
SetParm=0x1626,1   ! InitialAlarmState (None=0, AIS=1, RAI=2)
```

Note: The lineAdmin section for each trunk can specify its own trunk preconditioning.

Whenever a CONFIG file has been modified, a new FCD file must be generated. This procedure is described in detail in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

1.73.3 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about **gc_OpenEx( )** and other Dialogic® Global Call API functions, see the following documents:

- *Dialogic® Global Call API Programming Guide*
- *Dialogic® Global Call API Library Reference*

Note: The online bookshelf has not been updated for this feature, so the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* does not currently include information about the Initial Alarm State **(0x1626)** parameter.

1.74 Extended Board Management API Support for Dialogic® DM3 Boards

The Dialogic® Board Management API provides runtime fault monitoring and management of boards. For example, in the event of an application or host crash, channels can be set out-of-service to prevent the switch in the CO from sending calls to a board if there is no application to process them. This prevents the acceptance of unwanted calls and the potential of being unnecessarily tariffed.

Formerly, the Board Management API was supported only for T1 North American ISDN protocols (4ESS, 5ESS, DMS100, DMS250, and NI2). The Service Update extends support of the Board Management API to DM3 Boards using E1/T1 CAS (PDK protocols), additional T1 ISDN (NTT and QSIG-T1), E1 ISDN (NET5 and QSIG-E1), DPNSS, and DASS2.

These enhancements are applicable to the following boards:

- Dialogic® DM/V, DM/V-A, DM/V-B, and DM/VF Media Boards
- Dialogic® DM/IP Boards

1.74.1 Feature Description

The **brd_SendAliveEnable( )** function enables host fault monitoring on the specified board. When enabled, the board monitors the host computer for the presence of a repeated “heartbeat,” or “ping.” The heartbeat is sent to the board by the **brd_SendAlive( )** function from an application on the host computer. If the board does not receive the “heartbeat” or “ping” message within the required parameters defined in the

brd_SendAliveEnable( ) function, the board treats it as a host failure. When this occurs, the board takes its network interface out-of-service, thus preventing the network from offering calls to the failed system. The board also releases/drops all active calls and frees associated memory.

The network interface is taken out-of-service by sending an Alarm Indication Signal (AIS) toward the network. This is the ITU recommended mechanism for informing the CO or network that the trunk is not available. In addition, for the T1 ISDN protocols that support it, the Q.931 maintenance message SERVICE (Out-Of-Service) is also used to inform the network that the channels are no longer available. The AIS alarm (and SERVICE message when applicable) are cleared automatically when the trunk is put in-service using **gc_OpenEx( )** or **gc_Open( )** on the trunk device (dtiBn) or a channel (dtiBnTm) in any given trunk, following the host or application crash.

In the event that an AIS alarm was being transmitted on some other trunks prior to the crash, then the AIS alarm on those trunks will not be cleared when the other trunks are put back in-service. In this case, the application needs to clear the alarm using the Global Call Alarm Management System (GCAMS) functions; see the Alarm Handling section in the *Dialogic® Global Call API Programming Guide* for information.

1.74.2 Documentation

The online bookshelf provided with Dialogic® System Release 6.0 PCI for Windows® contains information about all system release features including features for application development, configuration, administration, and diagnostics.

For more information about the Dialogic® Board Management API, see the *Dialogic® Board Management API Library Reference*. There are no API changes because of this feature; the only changes are:

- Support for E1/T1 CAS and E1 ISDN protocols in addition to T1 ISDN on DM3 Boards
- Sending an AIS alarm for all protocols rather than a protocol-specific out-of-service condition

For more information about GCAMS, see the following documents:

- *Dialogic® Global Call API Programming Guide*
- *Dialogic® Global Call API Library Reference*

1.75 New Boards Supported

The following new boards are supported in Dialogic® System Release 6.0 PCI for Windows® with this Service Update:

- Dialogic® D30E1P Voice Board
- Dialogic® D/4PCIU4S Media Board
- Dialogic® D/42-NE2 PCI PBX Integration Board

The D/4PCIU4S is a 4-port analog loop start media processing board that can support either voice with CSP, or voice with fax, based on the firmware file selected.The firmware file is specified in DCM using the **FirmwareFile** parameter on the **Misc** property sheet. The default firmware file is d4u.fwl for voice with fax. For voice with CSP, select the d4ucsp.fwl firmware file.

Note: The D/4PCIU4S Board is displayed as D/4PCIU in DCM.

The D/42-NE2 PCI PBX Integration Board was supported in older system releases and is now supported in Dialogic® System Release 6.0 PCI for Windows®. The following documents have been added to the online bookshelf to support the use of this board:

- *Dialogic® D/42 Series Software API Reference*
- *Dialogic® D/42 Series User's Guide*

Release Issues

The table below lists issues that can affect the hardware and software supported in Dialogic® System Release 6.0 PCI for Windows®. The following information is provided for each issue:

Issue Type

This classifies the type of release issue based on its effect on users and its disposition:

- Known – A minor hardware or software issue. This category includes interoperability issues (i.e., issues relating to combining different Dialogic® products in the same system) and compatibility issues (i.e., issues that affect the use of Dialogic® products in with third-party software or hardware). Known issues are still open but may or may not be fixed in the future.
- Known (permanent) – A known hardware or software issue or limitation that will not be fixed in the future.
- Resolved – A hardware or software issue that was resolved (usually either fixed or documented) in this release.

Defect No.

A unique identification number that is used to track each issue reported via a formal Change Control System. Additional information on defects may be available via the Defect Tracking tool at http://membersresource.dialogic.com/defects/.
Note that when you select this link, you will be asked to either LOGIN or JOIN.

PTR No.

Number from problem tracking system used prior to March 27, 2006. For customer convenience, both the PTR number and the corresponding defect number are shown. For issues reported after March 27, 2006, this column contains "--" and only the defect number is used to track the issue.

SU No.

For defects that were resolved in a Service Update, indicates the Service Update number. For defects that were resolved when the base release was generally available (before any Service Updates), a "--" is shown. For non-resolved issues, this information is left blank.

Product or Component

The product or component to which the issue relates, typically one of the following:

- A system-level component; for example, Host Admin
- A hardware product; for example, Dialogic® DM/V Boards
- A software product; for example, the Dialogic® Global Call library

Description

A summary description of the issue. For non-resolved issues, a workaround is included when available.

The following table lists all issues that relate to this release, sorted by Issue Type. For other sort orders, use the following links:

- View issues sorted by Service Update Number
- View issues sorted by Product or Component
- View issues sorted by Defect Number

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows®

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known	IPY00022362	19779		Clocking	When setting up mixed board configurations involving both SCbus and CT Bus, errors may arise as the result of the placement of the clock master.
Known				Clocking	In the DCM, use the TDM Bus device to configure clocking in either SCbus or CT Bus mode. You will no longer find the SCbusClockMaster and SCbusClockMasterSource parameters. Instead, use the following parameters when the bus is configured in SCbus mode: • Specify the SCbus clock master via the “Primary Master FRU (User Defined)” parameter. • Specify the SCbus clock master source via the “Derive Primary Clock From (User Defined)” parameter. A drop-down list will be provided so that you can select between an internal oscillator (equivalent to INDEPENDENT in older implementations of SCbusClockMasterSource) and a specific trunk to derive clock from (equivalent to LOOP in older implementations of SCbusClockMasterSource).
Known	IPY00022145	23509		Conferencing (DCB)	On Dialogic® DM3 Boards, the conference notification tone (a tone that is generated when a party enters or exits a conference) is enabled by default; whereas on Dialogic® Springware boards, the default was to disable conference notification tone. The two problems with enabling this tone by default are: • This is the opposite of what Springware provides. • There is no API method available to indicate that tone notification should be disabled when establishing a conference. While the MSCA_NN bit mask exists to disable notification to receive-only parties or monitors, this parameter doesn't apply to all party types. Additionally, it is suspected that the notification tone may be contributing to PT 23506 against noises generated when parties enter or exit a conference (in addition to any notification tone). As a result, if the suggestion is that tone notification should be disabled for better conferencing quality, by default the feature should be disabled. Workaround: Update the FCD/CONFIG files with a parameter to disable conference tone notification by default.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known	IPY00008091	31432		Conferencing (DCB)	The maximum value for the **dcb_estconf( )** function **numpty** parameter is 24. If you attempt to establish a conference with more than 58 parties initially by using the **dcb_estconf( )** function with **numpty** parameter set to greater than 58, rather than giving an API error, the driver will fail. Workaround: To create a conference with greater than 24 parties, use the **dcb_estconf( )** function to create a 24-party conference, and then use the **dcb_addtoconf( )** function to add parties one at a time. You can have as many parties in a conference as is supported by your particular board and its media load.
Known	IPY00008269	29634		CPI Fax	When a tiff header analyzing utility (tiffhdr.exe) is used to analyze a tiff file that's received on a Dialogic® CPI Fax Board running in MH mode (TIFF Group 3 1-D), it generates an error saying "error 2905 on line 138". Workaround: Use another encoding than MH.
Known	IPY00006204	34764		CSP	If the ECCH_CONVERGE parameter is set on a Dialogic® DM3 Board that does not support this feature, the board fails with error "device busy" and "SP 2 NOT RESPONDING." (The only board that supports the ECCH_CONVERGE parameter is the Dialogic® DMV160LP Board.)
Known				D/120JCT-EURO	The Dialogic® D/120JCT-EURO Board is unable to detect a standard continuous DISCONNECT tone (350 Hz, 440 Hz). Workaround: The DISCONNECT tone can be detected if FREQUENCY RES is set to "LOW" (125 Hz value). Therefore, select Euro-21 as country for D/120JCT boards; this has FREQ-LOW as the default frequency resolution.
Known	IPY00022110	22030		D/120JCT-LS	Routing LSI resources from the Dialogic® D/120JCT Boards results in heavy noise (echo problem).
Known	IPY00007346	29275		D/120JCT-LS	When configuring PCM encoding on a system containing both Dialogic® Springware Boards and Dialogic® DM3 Boards, the Dialogic® D/120JCT-LS Boards default to mu-law. When attempting to reconfigure from configured devices (top level) to A-law, the D/120JCT-LS Board does not change; it remains mu-law. Workaround: Configure PCM encoding for each board manually.
Known				D/120JCT-LS	The Dialogic® D/120 Rev2 Board supports Japan Caller ID, but the EC_RESOURCE must be set to ON. This reduces the density of the board to 8 channels. The D/120 Rev1 Board does not support Japan Caller ID.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known	IPY00008850	31245		D/480JCT-2T1 D/600JCT-2E1	When Global Level PCMEncoding is not set to “Automatic”, which is the default, it must be set to match the Board Level PCMEncoding for the Dialogic® D/480JCT-2T1 and D/600JCT-2E1 Boards.
Known	IPY00021906	21049		D/82JCT-U	Call Progress test fails 100% on Dialogic® D/82JCT-U Boards. **Explanation:** The D/82JCT-U voice resources cannot be used to provide CPA capability for other boards.
Known	IPY00021905	21029		D/82JCT-U	If a voice resource of a Dialogic® D/82JCT-U Board is listening to a front end other than the default (its own), it may return a disconnected result. **Explanation:** The D/82JCT-U Board supports the call progress analysis feature of **dx_dial( )**, only when the D/82JCT-U is using the default TDM routing.
Known	IPY00021909	21082		DCM	When using Dialogic® Global Call SS7, Dialogic® SS7 Signaling Boards (PCCS6) can be added manually via DCM, but DCM doesn't allow the SS7 board to be used as the SCbus clock master.
Known	IPY00022112	22090		Device Naming	**dx_open( )** causes an application crash with access violation when the size of name is equal to or more than 31 characters. **Explanation:** There is a limitation on the length of a device name. It can be no longer than 30 characters.
Known	IPY00026634	22919		DI Boards	**dx_playtone( )** can only play up to 35 unique tones.
Known	IPY00020991	20150		DI Boards	**ms_setvol( )** does not affect the volume level on the Dialogic® DI/SI32 or DI/0408-LS-A Board.
Known	IPY00022220	25318		Dialogic System Service	When starting the Dialogic® System Service (DSS) via the Windows® Service Control Manager, a Progress dialog box may display the following error message: “Error 1053: The service did not respond to the start or control request in a timely fashion” This Progress message appears when any Windows service started via Service Control Manager takes more than 3 minutes. The message is misleading because the DSS may still be in a start pending state and start completion should occur within 7 additional minutes. The 3 minute limit is hard-coded by Microsoft in Windows 2000. Workaround: Start the DSS using the DCM tool. **Note:** Refer to the Microsoft Knowledge Base, Product Support Services documentation for further details about Error 1053.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known	IPY00030640	23783		DM/IP Boards	There is no interoperability between a Dialogic® DM/IP Board and a Siemens IP phone in Fast Start mode. When working in Fast Start mode with the DM/IP Board as the originator, the DM/IP Board sends facility with reason startH245 which the Siemens IP phone does not support. Workaround: Work in Slow Start mode.
Known	IPY00028247	32107		DM/IP Boards	IP calls cannot return to voice session after T.38 fax transmission. Workaround: Must drop the call and reconnect to return to voice session.
Known	IPY00027310	29279		DM/IP Boards	Customer application may see intermittent failures on fax over IP.
Known	IPY00026608	28884		DM/IP Boards	The **ipm_SetParm( )** IPM_RFC2833MUTE_AUDIO parameter is not supported on Dialogic® DM/IP Boards.
Known	IPY00022155	23727		DM/IP Boards	When Dialogic® DM/IP Boards with 100Base-T NICs are used and the boards are properly configured, the boards will echo back a response if they are pinged from within the subnet. However, if they are pinged from outside the subnet, they do not echo any response. **Explanation:** This is a configuration issue. DM/IP boards can be configured for a board-specific Network Configuration using the DCM Network tab. In the Network tab, HostIPAddress and GatewayIPAddress are used to add a network route to the subnet of HostIPAddress by gateway GatewayIPAddress in the DM/IP board. There are two types of possible configurations: • Configuration 1. For example, if board IPAddress =192.50.50.20, HostIPAddress=192.50.49.12, SubnetMask=0xFFFFFF00, and GatewayIPAddress= 192.50.50.250. A route will be added in the board where 192.50.50.250 will act as the gateway to subnet 192.50.49.0 and there is no default gateway configured. In this case, if a Host resides in a subnet other than 192.50.49.0 will not be able to communicate with the board. • Configuration 2. For example, if board IPAddress=192.50.50.20, HostIPAddress=0.0.0.0, and GatewayIPAddress=192.50.50.250. A route will be added where 192.50.50.250 will act as default gateway to internet. In this case, a host residing anywhere in the network can communicate with board as long as board and host has network connectivity. You can choose to configure the routing table to either set up default gateway or only talk to specific subnet (which may serve security purposes).

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known	IPY00019225	25596		DM/IP Boards	All test files *.c and *.h in the iptmail_r4.dsp and gateway_r4.dsp have an extra carriage return CR added: 0D0D0A (CRCRLF) instead of 0D0A (CRLF) which prevents the Microsoft Development Studio's editor from proper alignment of the source code with debug info. Workaround: Use VisualStudio ver 6.0 instead of ver 5.x to compile the demo files. Doing that, you will get a dialog box stating that there is a problem with the format of the files and asked to approve conversion of the files to the Windows® environment. By choosing OK, the conversion will be run smoothly and the compilation will be successful.
Known	IPY00008999	31419		DM/IP Boards	Tone clamping is insufficient for GSM 20ms 3FPP. Bleed through tones are apparent in 1.125%. Workaround: Applications should not mix the out-of-band and in-band DTMF transfer techniques. Must use one or the other, not both.
Known	IPY00008656	31404		DM/IP Boards	While in RFC2833 mode, two of the same DTMFs sent consecutively are detected as only one DTMF. A string like: “112233445566778899” will be received as: “112334455667899” Workaround: The RFC2833 transmission itself is fine. Improved performance may be realized if the end point ensures the inter digit off time of 75ms or larger.
Known	IPY00008451	31342		DM/IP Boards	Marker bit in RFC2833 packet is not set properly on G.723 and G.729 codecs at multiple frames per packet rate and with G.711 codec at frame sizes greater than 10 ms.
Known	IPY00008332	29455		DM/IP Boards	**gc_Extension( )** API does not support IPPARM_DTMF_RFC_2833 parameter for generating DTMF digits using RFC 2833. An application should use a dxxx device to generate dual tones which will be encoded as RFC 2833 digits by the associated ipm device when the DTMF transfer mode is set the RFC2833. Workaround: Use the Voice Device to dial/receive digits in a call that uses RFC2833 as DTMF transfer mode.
Known	IPY00007957	31871		DM/IP Boards	IP calls cannot return to voice session after T.38 fax transmission. Workaround: Must drop the call and reconnect to return to voice session.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known	IPY00007879	31659		DM/IP Boards	The default nfrm value for user-defined tones on Dialogic® DM/IP Boards is 10. In situations where the DM/IP Board firmware is being used in a PSTN environment, this value may need to be lowered if you are seeing cadence detection failures. PSTN products (QVS) use a value of 3, but preliminary testing shows that a value of 5 is sufficient to reduce errors to 0 for DM/IP Boards.
Known	IPY00007701	31594		DM/IP Boards	RFC2833 failures occur when using low bit rate coders at 3-4% failure rate when tone duration is longer than 2 seconds.
Known	IPY00007555	29164		DM/IP Boards	The error message (RTP Recv: in media_RTPUnpack( ) SSRC failed) may occasionally be seen in DebugAngel when running Dialogic® DM/IP Board configurations. Workaround: Ignore the message; it does not affect the application.
Known	IPY00007322	29280		DM/IP Boards	Call progress over IP will fail intermittently in applications. If the application performs call analysis to find out how it was connected (e.g., PAMD, fax) it won't be able to do it consistently.
Known	IPY00006725	31651		DM/IP Boards	The multicast transmit address can be set as addresses other than the reserved multicast addresses. No error checking is available to prevent or warn against this. Workaround: Ensure that only valid IP addresses (244.0.0.0 to 239.255.255.255) are used.
Known	IPY00006467	29048		DM/IP Boards	When the host application crashes or is abnormally terminated (e.g. killed in Task Manager), Exit Notification does not cancel all events in the Dialogic® DM/IP Board firmware. The boards must be re-downloaded or else the application will fail to initialize. Workaround: Re-download boards after a host application crash.
Known	IPY00006466	29008		DM/IP Boards	**ipm_GetSessionInfo( )** returns all zeros instead of valid information. **Explanation:** No information will be returned if an RTP session has not been created, or if called before an RTCP report has been received during a session.
Known				DM/IP Boards	Echo when using Dialogic® DM/IP Boards with Microsoft NetMeeting. Workaround: Disable the residual speech flag by setting prmECResSpFlagEnableDisable (parameter number 0x1b65) to 0 in the config file, then restarting the Dialogic® System Service.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known				DM/IP Boards	To select a coder when using Dialogic® DM/IP Boards with Microsoft NetMeeting, invoke the coder list (Tools>Options>Audio>Advanced) to bring your preferred coder to the top of the priority list. For best results, Microsoft NetMeeting and the DM/IP component should be configured to use the same coder.
Known				DM/IP Boards	Microsoft NetMeeting does not support VAD when using Dialogic® DM/IP Boards. Workaround: Do not set the VAD parameter in the relevant demo configuration file (for example, gateway_r4.cfg).
Known				DM/IP Boards	A connection to Microsoft NetMeeting will fail if the T.38 fax coder is included in the capability list. Workaround: Do not include the T.38 fax coder in the capability list when establishing a connection with NetMeeting.
Known	IPY00023901	29646		DM/V Boards	TSC_MsgSetChanStateCmplt error occurs on ml1b ISDN load. Received following error message when running QsigE1 on Dialogic® DM/V-B Board during download. Error was seen in DebugAngel. `QHostFailed message 13:47:47.113| 002:CP1:[0xffff0000]TSC.000 Error - tscSetChanStateCmplt( ) - line 1 replyCount 0.` The appearance of this message does not affect normal operation.
Known	IPY00021432	30972		DM/V Boards	QERROR_WARNING warning may show up in DebugAngel window for basic call control or voice tests for Dialogic® DM/V(A) and DM/V(B) Boards. Such warnings are generated due to extreme load/stress conditions for CP pools while running certain test applications and are taken care of by Kernel CP Pools resize safety net.
Known	IPY00021419	30224		DM/V Boards	QERROR_WARNING warning may show up in DebugAngel window for basic call control or voice tests for Dialogic® DM/V(A) and DM/V(B) Boards. Such warnings are generated due to extreme load/stress conditions for CP pools while running certain test applications and are taken care of by Kernel CP Pools resize safety net.
Known	IPY00020482	30167		DM/V Boards	QERROR_WARNING warning may show up in DebugAngel window for basic call control or voice tests for Dialogic® DM/V(A) and DM/V(B) Boards. Such warnings are generated due to extreme load/stress conditions for CP pools while running certain test applications and are taken care of by Kernel CP Pools resize safety net.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known	IPY00007354	31617		DM/V Boards	When setting CODESET_7_IE or CODESET_6_IE together with USER_USER_IE in the setup message, neither IE is delivered to the remote side. However, when setting USER_USER_IE only, it is delivered to the remote side. Currently, Dialogic® DM3 Boards only support sending non-codeset 0 messages within supplementary services, such as FACILITY and NOTIFY.

Known				DM/V Boards	**R4 Compatibility Flag issue**: All systems running R4 applications on high density Dialogic® DM3 Voice and Network Interface Boards must make sure that the R4 Compatibility Flag parameter in the [CHP] section of the CONFIG files is set to 1 (this is the default value).
					You should enable the R4 Compatibility Flag in order to:
					• Delay reporting Offered call state until DNIS and ANI call information is available. (Only the timing is affected.)
					• Report ISDN Q.931 Cause values instead of Call State reasons in Call State events.
					• Support Q.931 Cause values as reasons in DropCall and ReleaseCall messages (ISDN protocols only).
					Set the R4 Compatibility flag in the [CHP] section as follows:
					`SetParm=0x1310,1       ! R4 Compatibility Flag`
					`                        ! 0-default, 1-enable,`
					`                        ! 2-disable`
					If this parameter is not set to 1, update the CONFIG file. After you make this modification, start the Dialogic® System Service for the change to take effect.

Known	IPY00024003	32882		DM/V160-LP	When setting up a supervised call transfer, after the **gc_SetupTransfer( )** function is issued to obtain a CRN for the consultation call, a permanent signal timer (8 seconds) starts. If the consultation call is not made within the 8 second period, the timer expires and the application receives a GCEV_DISCONNECTED event. The reason associated with the GCEV_DISCONNECTED event is “Event Caused by Protocol Error”. The reason should indicate that the disconnect is a result of the timer expiring.
Known	IPY00006684	31690		DM/V160-LP	Debug messages are being printed in DebugAngel when opening devices. Workaround: Ignore the messages. Despite the warning messages all functionality should work.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known	IPY00007996	32189		DM/V-B Boards	During high density, over 700 channels per chassis, an illegal termination of the system may result in a conditional lockup or hang state. Workaround: Either re-download or reboot the Dialogic Boards if the application terminates abnormally.
Known	IPY00022125	22545		DM3 Fax	Multi-page ASCII fax function always fails. This is a known limitation of the product. Workaround: Multiple pages can be sent by generating 1 iott per page (use the same handle, but define io_offset/io_length). However, most fax devices allow receipt of a “long page”.
Known	IPY00005994	33137		DM3 Fax	While running fax load tests on Dialogic® DM3 E1 fax boards, faxes stop transmitting after several hours.
Known	IPY00035574	--		DM3 Firmware	There is an interoperability issue when using Dialogic® DM/IP Boards with Dialogic® DM/V-B Boards (in PCI or PCI Express form factor) in a Dell PowerEdge 6850 chassis. It has been observed with certain high-load regression test applications run, a QERROR_KILLTASK error is generated and reported in the DebugAngel, and the DM/IP firmware crashes. This issue occurs when running voice functionality on the DM/IP and DM/V-B Boards simultaneously; the failure is on the first voice operation on the DM/IP board. Workaround: Disable direct memory access (DMA) transfers on the DM/IP Board by setting the **doDMA** parameter to 0 (off) in DCM.
Known	IPY00006407	36806		DM3 Firmware	Error Message “Warning: PDK SetProtocolOutOfService timed out” is generated intermittently on Dialogic® DMV1200BTEP Boards (with UL3 or ML10-LC R2MF firmware) when the application is stopped and started without re-downloading boards.
Known	IPY00006393	36758		DM3 Firmware	Random digit(s) missed in DNIS string on random Inbound calls when running with CAS media loads.
Known	IPY00006353	36792		DM3 Firmware	Error message “Warning: PDK SIG_AlarmOff timed out” is generated on Dialogic® DM/V1200A-4E1-PCI Boards (with ml1b_qsa_r2mf firmware) when downloading Dialogic® DMV1200BTEP Boards connected to them.
Known	IPY00023900	28301		DM3 Network	Errors occur when trying to use the phone application's dial pad: Error: Error 0x500 Description {No Description Available} Data {ErrorMsg 0xa0002 ErrorCode 0x500 Data {0x0 0x0 0x0 0x0}} Workaround: Ignore the errors.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known	IPY00008411	32133		DM3 Network	Under heavy load with 100% CPU utilization, if the host application were to exit abnormally (i.e. application error, memory exception), specific real-time tasks running on the board may assert. The only way to recover would be to restart the Dialogic® Services. Workaround: To prevent this, avoid running at or near 100% CPU utilization.
Known	IPY00007385	29991		DM3 Network	Missing completion event from **gc_DropCall( )** if called right after **gc_Attach( )** in SYNC mode on Dialogic® DM3 Boards. Workaround: Call **gc_Attach( )** in ASYNC mode and wait for completion event.
Known	IPY00007234	23614		DM3 Network	When a trunk receives AIS, the LineAdmin utility only displays the GREEN and RED LEDs, and *not* the YELLOW one.
Known	IPY00008208	30489		DM3 Voice	On Dialogic® DM3 Boards, call progress using **dx_dial( )** does not return a result of CR_FAXTONE when a CNG tone is played. Note that CED tones are correctly interpreted as CR_FAXTONE.
Known	IPY00022317	28582		DMV160LP	GSM voice coder issue. Workaround: To use the GSM voice coder for recording on the Dialogic® DMV160LP Board, you must disable the DMA option in the Dialogic® Configuration Manager (DCM) using the following steps: 1. Make sure the board is in the stop state. 2. Go to the Driver tab. 3. Modify the setting for the “Do DMA” parameter to 0 and click Apply. (This parameter is set to 1 by default.) 4. Re-download the board. In certain environments, such as in a system with five or more boards with all channels active, you may experience an increase in CPU utilization of 4% to 6% when using GSM.
Known	IPY00022006	29073		DMV160LP	Disconnect tone supervision. You may need to adjust the definition for the disconnect tone depending on the PBX system you are using. To do so, modify its definition in the dmv160lp.config file. Use one of the following values: Fast Busy (reorder): 19938 Dialtone: 41571 Busy: 19937

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known	IPY00021978	28889		DMV160LP	DTMF termination condition issue. On the Dialogic® DMV160LP Board, when the record operations are set to be terminated on a DTMF, the recorded file may contain between 30 to 50 msec of the terminating DTMF.
Known	IPY00023977	31550		Global Call	For an object intending to use **gc_MakeCall( )**, all call temporal parameters need to be settable via the make call block. Workaround: Continue to use **gc_SetUserInfo( )** before **gc_MakeCall( )** to set parameters.
Known	IPY00022395	20083		Global Call	On Dialogic® DM3 Boards, MakeCall is not successful after placing lines in service. **Explanation:** When running ISDN through R4 on DM3, it is important to ensure the line has been put in service and the D channel has come up prior to making calls on any channels. Workaround: There are three possible ways to ensure this happens: • Open up the board device (e.g., dtiB1) and wait for the event GCEV_D_CHAN_STATUS (with reason E_LINKUP | ERR_ISDN_LIB) prior to placing any calls. • Prior to running the application, run the QSCRIPT tool "lineadmin -board x" where "x" is the number of the board and put all of the lines in service. • Add a 3 second delay into the application between the time the channels are opened and the first attempt to issue a MakeCall.
Known	IPY00022135	23048		Global Call	Exception while receiving GCEV_OFFERED event. **Explanation:** Customers should not use **sr_putevt( )** to send any Dialogic-specific event codes. The Dialogic® library that is usually sending this event may need to change its internal state and may go out of sync with other libraries that will also receive this event using high priority handles.
Known	IPY00021408	29419		Global Call on IP	Applications running Dialogic® Global Call Software on host-based SIP or H.323 stacks and issuing **gc_WaitCall( )** before every call produce errors at higher call rates. Workaround: Applications should issue **gc_WaitCall( )** only once for a given channel at the beginning of application or after **gc_ResetLineDev( )** has been completed (GCEV_RESETLINEDEV has been received).

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known				Global Call on IP	When developing applications that use IP technology, Dialogic® Global Call Software does not support synchronous programming models. Unless explicitly stated in the *Dialogic® Global Call IP Technology User's Guide*, Global Call functions should be called in asynchronous mode.
Known	IPY00030585	25536		Global Call on ISDN	When using Dialogic® Global Call Software with ISDN, if the inbound call is disconnected while the application was trying the answer the call, depending on the timing, the application might get a GCEV_TASKFAIL with error code 0x10f (BADSTATE). Workaround: The application should restart the time slot using **gc_ResetLineDev( )** to handle this glare.
Known				Global DPD	Speed control for the Dialogic® D/160SC-LS-IDPD, D/240SC-T1-IDPD, D/300SC-E1-75-IDPD, D/300SC-E1-120-IDPD, and D/320SC-IDPD Boards cannot be used while the Global DPD feature is enabled. If any speed control adjustments are attempted while Global DPD is enabled, the function will return with a -1, indicating failure. Workaround: You can adjust the speed before or after placing or receiving a call that uses the Global DPD feature.
Known				Global DPD	The Global DPD feature must be implemented on a call-by-call basis. For the Global DPD feature to work correctly, each time an incoming or outgoing call is initiated, Global DPD must be initialized by using the **dx_setdigtyp( )** function with the D_DPDZ flag. Refer to the *Dialogic® Voice API Library Reference.*
Known	IPY00028199	29038		HDSI Boards	If you start the HDSI demo on a Dialogic® DI/SI32 Rev2 Board with a phone in the off-hook state, the off-hook state will not be detected. The application will report only on-hook on the MSI channel and an application crash will occur quickly afterwards.
Known	IPY00008842	33625		HDSI Boards	Cannot download Dialogic® HDSI-960-PCI Board using us_ and at_hdsi_96_play_rec.pcd and .fcd files.
Known	IPY00006017	33633		HDSI Boards	Multiple play/record tests fail; an extra digit (usually a 0) is at the end of the string.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description

Known				HDSI Boards	A Station Interface Box (SIB) firmware upgrade may be required to support the 25 Hz and 50 Hz ring frequencies provided in the country-specific configuration files. You can use the kernelver utility to retrieve the Dialogic® HDSI Board and SIB firmware versions.
					To retrieve the firmware versions, start the HDSI Board in DCM and then run kernelver twice (once for the HDSI Board firmware version and once for the SIB firmware version) using the following syntax examples.
					To retrieve the firmware version of the SIB, run:
					`kernelver -bxx -p1 -s`
					where xx is the LogicalID of the HDSI Board attached to the SIB. The LogicalID for a given Dialogic® DM3 Board can be determined through DCM. For details, refer to the DCM Online Help.
					To retrieve the firmware version of the HDSI Board, run:
					`kernelver -bxx -p1`

Known	IPY00013365	19303		Host Admin	DCM fails to detect any Dialogic® DM3 Boards when one unrecognized board is present.
Known	IPY00013282	17053		Host Admin	DCM fails to detect any Dialogic® DM3 Boards when one unrecognized board is present.
Known	IPY00013252	17052		Host Admin	DCM fails to detect any Dialogic® DM3 Boards when one unrecognized board is present.
Known	IPY00009150	28379		Host Admin	The Error code in *NCMApi.h* says "NCME_RELEAS_TIMESLOT" instead of "NCME_RELEASE_TIMESLOT".
Known	IPY00008501	31050		Host Admin	Listboards will report an error on the screen if it is run after a "single stop" operation has been performed on some of the boards. The problem is not seen as long as all of the boards are running (i.e., no boards have been stopped). Workaround: Run listboards with the "-l2" option, which runs the updated version of listboards.
Known	IPY00008491	31206		Host Admin	When DSS is started or stopped programmatically via the NCM API and while the DCM GUI is already running, the service status is not updated. Hitting the refresh button will refresh the GUI and show the correct service status. Alternatively, if the GUI is opened after the service is stopped or started, it reflects the correct status.
Known	IPY00007797	31695		Host Admin	Data backup and migration will not work with Terminal Services.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known	IPY00007510	32342		Host Admin	CTBB event logs are being made without descriptions appearing in the Application log panel. CTBB event logs should not even appear in the Application log panel. CTBB event logs should only appear in the System log panel with descriptions.
Known	IPY00006547	33730		Host Admin	After the initial install, DCM detects boards in a different order after Restore Device Defaults is run.
Known	IPY00006036	31669		Host Admin	Using Terminal Server to control a GDK machine is not supported. If needed, remote access to a GDK machine can be easily done through other means (e.g. remote desktop connection using XP, Netop, etc). Workaround: Don't use remote access to the GDK machine. It is not a requirement to run a demo on a GDK machine remotely (through Terminal Server).
Known	IPY00006011	31701		Host Admin	When using the Event service consumer object to subscribe for events, sometimes the application hangs while exiting on Windows® XP systems. Workaround: A sleep of 1 second at the end of the application causes this hang to go away.
Known	IPY00005982	31435		Host Admin	dlgsnapshot utility is not functional on Dialogic® DM/V-B Boards.
Known	IPY00024819	31099		Host Install	When uninstalling the Dialogic® Software, a warning message stating “Error encountered after attempting to launch” may appear if an error was encountered after the setup program attempted to launch a utility. Workaround: Follow these steps if you encounter this error: 1. Click OK to continue with the uninstall. 2. Follow the documented procedure to run the clean-up utility. The procedure can be found at: *http://www.dialogic.com/support/helpweb/dxall/tnotes/legacy/2000/tn020.htm* 3. Reboot.
Known				Host Install	The PDKManager tool, which downloads Dialogic® Global Call protocol modules and country dependent parameters to Dialogic® DM3 Boards, can be set up to run automatically when DCM is started. However, after performing an update install, PDKManager no longer runs automatically. Workaround: PDKManager must be rerun manually after an update install. For further information about PDKManager, see the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known	IPY00023978	31618		Host Library	Applications utilizing a Dialogic® HiZ board may take 15-20 minutes to shut down. Workaround: Tearing down channels in parallel will greatly reduce the shutdown time (less than 30 seconds).
Known	IPY00023708	27287		Host Library	When using **ms_genRingCallerId( )** on a Dialogic® HDSI or DI/SI32 Board, with cadence MS_RNGA_SHORTLONG, the phone rings with the specified cadence, but does not show the caller id.
Known	IPY00009177	31642		Host Library	SRLGetSubDevicesOnVirtualBoard returns 0 sub devices for Dialogic® D600JCT-2E1 Boards.
Known	IPY00008960	31060		Host Library	Applications using cached prompts will crash or be abnormally terminated if they do not issue a **dx_close( )** before exiting the application. If this is not adhered to, the cached prompt API will fail on the next execution and the board will have to be re-downloaded. Workaround: Once cached prompt is downloaded, close the physical board with **dx_close( )** API before exiting from application.
Known	IPY00007762	31734		Host-Based H.323 Protocol Stack	Dialogic® Global Call Software applications that use the host-based H.323 protocol stack may generate the following error message in the *gc_h3r.log* file if the application enables the stack to send the PROCEEDING message automatically. `! 22:22:19.393 ! M_SIGNAL ! L_ERROR ! 1 ! << SIGNAL::sendProcceding: RV cmCallSendCallProceeding Failed : [-996]` Workaround: Ignore this error message. The PROCEEDING message is actually sent.
Known	IPY00017747	29044		IP CCLIB	For host-based configurations (using Dialogic® Global Call or IPML API), if QoS lost packets alarms (QOSTYPE_LOSTPACKETS) have been enabled and out of band signaling is being used (either H.245 UII or RFC2833), then QOSTYPE_LOSTPACKETS alarms will be generated whenever information is sent out of band. Workaround: Ignore the alarm (which is for information purposes anyway) or disable the alarm event.
Known	IPY00022147	23574		ISDN	**cc_SetParmEx( )** fails when changing parameters that require numerical values (e.g. BC_XFER_RATE). This is not applicable to the parameters that require character string values (e.g. DIRECTORY_NUMBER). Workaround: When using **cc_SetParmEx( )** to set a parameter that requires a numerical value, set the length field in the PARM_INFO structure to 1 and ensure that only the first byte in the parmdata field contains the value.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known	IPY00022142	23256		ISDN	**cc_SetChanState( )** is only supported on North American T1 PRI protocols. It is not supported on NTT (Japanese) or European E1 protocols since the specs of these protocols do not support SERVICE/SERVICE_ACK messages.
Known				Modular Station Interface (MSI)	Multi-threaded applications must enclose all MSI calls in a critical section or applications run the risk of getting intermittent errors. Reports “No error”, firmware does not assert but reports “cnt q overflow”.
Known				Modular Station Interface (MSI)	If the coach speaks before any conversation has taken place between the client and the pupil, the client will hear some background noise for a fraction of a second. Under most circumstances, this will not be a problem since the coach usually will not need to speak before some conversation has taken place between the client and the pupil.
Known				Modular Station Interface (MSI)	**ms_setcde( )** fails to return a valid error message when an invalid chan_attr is assigned.
Known				Modular Station Interface (MSI)	Setting the board parameter MSCB_ND through use of the **setbrdparm( )** function fails to configure the volume, tone, and duration of the notify-on-add tone.
Known	IPY00022053	20344		NCM API	All Dialogic® Boards in system must be detected using either **NCM_DetectBoards( )** or **NCM_DetectBoardsEx( )** before using **NCM_StartDlgSrv( )**.
Known	IPY00021901	20949		NCM API	All Dialogic® Boards in system must be detected using either **NCM_DetectBoards( )** or **NCM_DetectBoardsEx( )** before using **NCM_StartDlgSrv( )**.
Known	IPY00006533	31731		PBX Expert (previously called PBXpert)	When using PBX Expert with the Dialogic® DMV160LP Board, for disconnect tone supervision to work, it must be enabled in two places. The **DisconnectTone** parameter in DCM must be enabled by selecting Yes, and the **Tone_SigId4** parameter in the board’s CONFIG file must also be enabled. For information about enabling the **Tone_SigId4** parameter, see the [CHP] Analog Voice Variant Definitions section of the CONFIG File Parameter Reference chapter of the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known				Programming Using External References	Please refer to the following technotes on the support website for information regarding how to program using external references: *http://www.dialogic.com/support/helpweb/dxall/tnotes/legacy/dlsoft/tn254.htm* *http://www.dialogic.com/support/helpweb/dxall/tnotes/legacy/dlsoft/tn253.htm* Note that it is not applicable to Dialogic® DMV2400A-PCI, DMV1200BTEP, DMV600BTEP, and DMV3600BTEP Boards. For the DMV2400A-PCI Board, if using Transaction Record, the maximum number of concurrent transaction records per physical board is 120, even though the board supports up to 240 channels of standard record.
Known	IPY00007185	29270		SNMP	SNMP fails when run on loopstart boards.
Known	IPY00014335	26956		Springware Firmware	CPU usage remains high after program stops. The CPU usage only returns to normal after drivers are stopped.
Known	IPY00007598	25289		Springware ISDN	When using network-side ISDN firmware, **gc_MakeCall( )** does not fail and does not produce any error message when the data link is down. The function returns successfully, but no subsequent call control events are received on the channel.
Known	IPY00022133	23032		Springware Voice	Playing and recording a 21-second file using 176 kHz Linear Coder and SCR. When viewing the file in CoolEdit, the coder is recording the 21-second file into a file over a 1000 minutes long. **Explanation:** If a file is opened in “write only” mode, the library is unable to update the header because the code that does this uses a generic manner that also works when the file already existed and only the data is replaced. Workaround: When you do a WAVE recording, your application should open the files in the mode specified below: Change: dx_fileopen(fname, O_WRONLY|O_CREAT|O_BINARY, _S_IWRITE) To: dx_fileopen(fname, O_RDWR|O_CREAT|O_BINARY, _S_IWRITE|_S_IREAD)

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known	IPY00022082	21385		Springware Voice	Cannot create more than 10 user-defined tone templates on one voice channel. **Explanation:** This is a limitation of the firmware. The firmware can recognize a total of 20 terminating conditions for an IO event, however only 10 of them can be tones. So when the user sets up 11 tones as terminating conditions, the firmware issues an error that is captured at the driver level and the play is stopped immediately. Workaround: Enable tone events while setting up the tones. When that particular event is received, the play can be stopped.
Known	IPY00006637	32339		Springware Voice	Async polled mode demo does not pick up the call after “Close Channel” button is used. The application does a **dx_unlisten( )** during cleanup, but it fails to do a **dx_listen( )** during init. This means that the front end and voice timeslots are now permanently disconnected. **Explanation:** This demo was designed originally for older Dialogic® Springware Boards that are now obsolete. These Springware boards required that the analog and voice timeslots be routed by the application. However, the newer Dialogic® JCT boards do not require this. Workaround: Stop and start the board in DCM for the demo to work again.
Known	IPY00040086	--		Windows Vista	DCM does not have help files that are compatible with Windows Vista®. After opening DCM and going to the contents item under the help menu, there is a message “Failed to execute DCM online help (config.hlp).” Workaround: Refer to Section 1.4, “Support for Windows Vista® Operating System”, on page 40; in particular, see WinHlp32.exe Not Included in Windows Vista.
Known	IPY00040083	--		Windows Vista	When running Dialogic® Diagnostics Software (UDD) with User Account Control enabled, error messages occur. Workaround: Refer to Section 1.4, “Support for Windows Vista® Operating System”, on page 40; in particular, see UDD Must Be Set to “Run as Administrator”.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known (permanent)	IPY00006127	33837		Board Download	The UK country parameter file for the Dialogic® D/41JCT-LS Board is missing from the release. If a D/41JCT-LS Board is configured for use in the United Kingdom (DCM Country parameter is set to United Kingdom), the system will start, but the system event log will report the following error: Error downloading file. The uk_41j.prm file cannot be found. Workaround: To correct this problem, locate the file *eu_41j.prm* in the *Program Files\Dialogic\data* directory and rename this file to *uk_41j.prm*.
Known (permanent)	IPY00028668	36716		Conferencing (CNF)	When trying to add an ipm device (e.g., ipmB1C2) to a conference, the **cnf_AddParty( )** function times out after 30 seconds. Around the same time as the time-out, an UNLISTEN event, followed by a LISTEN event, is received for the ipm device being added. The conference connection does appear to connect; audio does get passed through. But a time-out error is always returned.
Known (permanent)	IPY00026331	28279		CPI Fax	GFXHEADER does not work with .fls file expansion.
Known (permanent)	IPY00022296	19492		D/120JCT-LS	Older Dialogic® D/120JCT-LS Boards may experience a problem when trying to increase the amplitude by more than 5 dB. This is not a problem with the newer versions of the board.
Known (permanent)	IPY00010221	35118		D/600JCT-2E1	The second trunk of a Dialogic® D/600JCT-2E1 Board cannot be used as a clock source in SCbus mode. Setting the DCM parameter **DerivePrimaryClockFrom** to Front_End 2 causes download errors. Workaround: Use Front_End 1 as the clock source.
Known (permanent)	IPY00008602	30950		D/600JCT-2E1	The Dialogic® D/600JCT-2E1 Board may fail to clear alarms when the cable is manually removed and reconnected quickly many times, because the PMC chip on these boards may incorrectly report that the alarm is still present. Workaround: To clear the condition after the lockup, remove the cable and reconnect one more time.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known (permanent)	IPY00022342	15033		D/82JCT-U	"Parm not set correctly" error for board and channel parameters. The problem occurs when the **d42_setparm( )** or **d42_getparm( )** function is called simultaneously on multiple threads for the same channels on the Dialogic® D82JCT-U Board. Very infrequently, the function call will fail. Workaround: There are several workarounds for the problem. • Re-issue the **d42_setparm( )** or **d42_getparm( )** function when the function returns this failure. • Limit calls to **d42_setparm( )** or **d42_getparm( )** to a single thread in the application. • Implement a semaphore in the application to serialize calls to these functions in a multithreaded application.
Known (permanent)	IPY00022390	19978		DCM	The Terminal Server program cannot auto detect boards on a Windows® 2000 machine. **Explanation:** The Remote DCM is not designed to work via Terminal Server for security reasons.
Known (permanent)	IPY00022293	19308		DCM	The *spandti.prm* file is not downloaded by default. Workaround: Specify this parameter in the "ParameterFile" in DCM in order for it to take effect.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known (permanent)	IPY00029958	36722		DM3 Driver	DCM may not detect all boards in systems with D865GBF series motherboard. This problem occurs only with some revisions of the BIOS. Workaround: This problem can be corrected by updating the BIOS to version **P25**. First, determine if your system has the D865GBF series motherboard and check the BIOS version as follows: • From the Windows® Start menu, select Run, type MSinfo32, and click OK. • In the System Information window that is displayed, check the System Model and BIOS Version/Date values. For systems with a D865GBF series motherboard, the System Model is D865GBF. The BIOS Version/Date will be something similar to this: BF86510A.86A.0075.P24. (This shows BIOS version **P24**.) If your system has the D865GBF series motherboard and an earlier BIOS than version P25, update the BIOS to version P25 as follows: • Go to the following website for the Intel Desktop Board D865GBF: *http://downloadfinder.intel.com/scripts-df-external/Product_Filter.aspx?ProductID=948&lang=eng* • Follow the instructions provided at that website. Be sure to read the Release Notes and special instructions to be followed prior to installation.
Known (permanent)	IPY00031563	36612		DM3 Firmware and Host Runtime Library	A quick execution of **gc_Listen( )**, **dx_listen( )** to the same time slot, followed by **dx_unlisten( )** and **gc_UnListen( )**, results in an error in the RTF logging. Workaround: When calling **unlisten( )**, the application should implement a guard time (i.e., sleep) of 100 msec if **listen( )** has been called for the same time slot. (Calling **unlisten( )** for a different time slot does not require the guard time.) The application must still unroute time slots in reverse order (i.e., voice then network).
Known (permanent)	IPY00022290	19115		Global Call	In US_MF_O protocol, if **gc_DropCall( )** is called soon after **gc_MakeCall( )**, the line gets stuck in blocked state. Workaround: Redesign the application's state machine so that it does not call **gc_DropCall( )** within a few seconds of **gc_MakeCall( )**.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known (permanent)	IPY00022262	27343		Global Call	When using the us_mf_io ICAPI protocol (and possibly other ICAPI protocols) with Dialogic® Springware Boards and using the dx_ method of call analysis, fax tones may be misinterpreted if the ca_pamd_spdval field is set to PAMD_FULL in the DX_CAP structure. Workaround: Setting the ca_pamd_spdval field to PAMD_ACCU always gives the correct fax tone detection.
Known (permanent)	IPY00021424	30376		Global Call	This release does not support the ANAPI library or protocols. Customers should migrate to the equivalent Analog PDK protocol.

Known (permanent)	IPY00022261	27289		Global Call on ISDN	When using Dialogic® Global Call APIs and ISDN call control APIs in the same application, the user application must include *gclib.h* before *cclib.h*:
					`#include <gclib.h>`
					`#include <cclib.h>`

Known (permanent)	IPY00028395	35879		Host Admin	During system startup, if the system is configured to use DHCP, network connectivity problems may cause the DHCP service to respond slowly or not at all. When this occurs, DCM may fail to start the Dialogic® Boards. This may happen even if Remote DCM is not used. Workaround: The problem may be resolved either by correcting the network connectivity problem or, if you are not using Remote DCM, by adding the following line: **ooc.iiop.host=127.0.0.1** in the section titled "Settings For All Servers" in the *dlgadmin.config* file located in the dialogic/cfg directory. This will cause Remote DCM to not operate, but DCM will now be able to successfully start the boards locally.
Known (permanent)	IPY00008157	32588		Host Admin	CTBusBroker posts a warning message in the event log saying that FRU doesn't support common media type. Workaround: Ignore the message, as it doesn't affect system operation.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known (permanent)	IPY00006578	34616		Host Admin	Due to security enhancements implemented in Windows® XP SP2 and Windows Server® 2003 SP1, Remote DCM will no longer work with these operating systems. Workaround: In order to allow Remote DCM to work again, you have to revert the security settings to the pre-service pack states for the machine being accessed remotely by modifying the following two Windows® settings: First Setting: • Go to Control Panel -> Administrative Tools -> Component Services. • Go into the Properties page of Console Root -> Computers -> My Computer. • Under the COM Security tab, click on Edit Limits… button for both Access Permissions as well as Launch and Activation Permissions. • For Access Permissions, make sure “ANONYMOUS LOGON” has local as well as remote access. • For Launch and Activation Permissions, make sure “Everyone” has all local as well as remote permissions. Second Setting: • Create/modify the registry value “HKEY_LOCAL_MACHINE\SOFTWARE\Policies\Microsoft\Windows NT\RPC\RestrictRemoteClients”. It is a DWORD value that has to be set to 0 in order for Remote DCM to work. After these settings are applied, reboot the machine and the machine should be ready to be remotely managed through DCM again. Refer to the Microsoft support website for additional information on the security enhancements in new service packs.
Known (permanent)	IPY00032264	36119		Host Drivers	Under high load of **cc_PlayTone( )**, the device doesn’t return an event to indicate the end of the **cc_PlayTone( )**.
Known (permanent)	IPY00037706	--		Host Install	‘MERCCONFIG - DLGCMPD driver failed to start’ error is shown in the Windows® Event Viewer when only Dialogic® Springware Boards (no Dialogic® DM3 Boards) are installed. Workaround: The appearance of this error message on a **Springware-only system** does not indicate an error; it does not affect system use and can be ignored.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known (permanent)	IPY00028427	36079		Host Install	DCM fails to download the board when the cdp file name is wrong in pdk.cfg. An error message appears in the Event Log, but it doesn't adequately describe the error condition. Workaround: Check the pdk.cfg file for accuracy and check pdk.log in ..\Dialogic\log for errors.
Known (permanent)	IPY00028358	33991		Host Install	After installation, the operating system does not have all the drivers loaded for all the boards. Workaround: Take the following actions. **Use Case 1:** When users see the New Hardware Wizard pop-up: 1. Click “Next”. 2. Go to the “Advanced option”. 3. Point to the location where Dialogic software is installed, “…\dialogic\driver\” directory. 4. Click “Next”. 5. Click “Finish”. 6. Repeat this process for all the boards for which you see the pop-up. **Use Case 2:** Installation is complete and system rebooted; not all boards are detected by DCM: 1. Go to the “Device Manager”. 2. Expand the “DM/HDSI” and see if any board has a Yellow “!”. If yes, continue with the following steps. 3. Double click on this device. 4. Click update driver / reinstall driver. 5. Follow the same steps as above (Use Case 1). **Use Case 3:** Installation is complete and system rebooted; not all boards are detected by DCM: 1. Go to the “Device Manager”. 2. Expand the “PCI Devices” and see if any board has “Unknown PCI Device”. If yes, continue with the following steps: 3. Double click on this device. 4. Click update driver / reinstall driver. 5. Follow the same steps as above (Use Case 1).
Known (permanent)	IPY00016036	27709		ISDN Q.SIG Protocol	The Q.SIG E1 network-side firmware allows the host to send a SETUP_ACK message even when the incoming SETUP message contains a SENDING COMPLETE information element. This causes a glare condition between two DISCONNECT messages, followed by a glare condition between two RELEASE messages. The RELEASE_COMPLETE message is not sent.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known (permanent)	IPY00007501	30091		JCT Boards	A single 400 Hz tone can be erroneously detected as a dual tone defined as 459±40 Hz, 330±40 Hz because the DSP cannot detect dual tones that are less than 60 Hz apart. It is recommended that the dual tone be defined as 459±20 Hz, 330±20 Hz. This restriction is only for certain call progress tones including disconnect, extra dial tone, local dial tone, international dial tone, ringback, busy, and fax. **Rationale:** The DSP cannot reliably detect dual tones that are 60 Hz apart. In order to compensate, the firmware sets up a “twin” tone. A twin is a single frequency tone that will be reported to user applications as the dual tone. The frequency range of the twin tone is based on the frequency of the dual tone. For example, if the disconnect tone is set up as 330±40 Hz and 459±40 Hz, the maximum low frequency is 330 + 40 or 370 Hz and the minimum high frequency is 459 - 40 or 419 Hz. As these two frequencies are only 49 Hz apart, a twin frequency will be set up. In this case, the twin will have a lower bound of 359 Hz and an upper bound of 430 Hz. If the channel is presented with a single tone in this frequency range, it will be reported to the user application as a disconnect tone.
Known (permanent)	IPY00022130	22957		PBX Integration	The Dialogic® PBX Integration Board (D/82JCT or D/42JCT) MUST be the Primary Master in the system deriving reference from its Frontend. Any other configuration will result in frame slips which will cause the board and PBX to run out of synch and data to be lost. This is dictated by the PBX hardware design.
Known (permanent)	IPY00022119	22259		PBX Integration	The Dialogic® PBX Integration Board (D/82JCT or D/42JCT) MUST be the Primary Master in the system deriving reference from its Frontend. Any other configuration will result in frame slips which will cause the board and PBX to run out of synch and data to be lost. This is dictated by the PBX hardware design.
Known (permanent)	IPY00021449	31707		Protocols	When running R2MF PDK protocols on Dialogic® DM3 Boards under a flexible routing configuration, as you increase the density across multiple boards, connect failures may increase when an off-board voice/media resource is used for call control. These failures can be reduced by using a voice/media resource from the local board that is doing the call control and not from any other board in the system.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known (permanent)	IPY00032742	--		Runtime Trace Facility (RTF)	ERR1 in RTF log can be ignored when using Dialogic® DISI Switching Boards. When you use DISI Boards in your system, the following prints will appear with an ERR1 label in the RTF log files: `04/13/2006 17:54:50.386   3348        3352 Cheetah            ERR1      Dm3Player dxxxB28C1 ----- DOES NOT EXIST` This is not considered an error for DISI Boards. DISI Boards have no player or recorder components and that is why these messages are getting printed. System performance is not affected significantly. However, although these are not errors on DISI Boards, the same print could be an error for other types of boards. These prints are useful in determining if a board doesn't have any normal components and when debugging firmware load issues and OAM device enumeration issues.
Known (permanent)	IPY00032735	--		Runtime Trace Facility (RTF)	ERR1 in RTF log can be ignored when fax resources are involved. The following prints might appear as ERR1 in the RTF log when the Dialogic® Fax API is used: `04/14/2006 17:11:57.201   2508        1484 Cheetah            ERR1      Dm3Stream dxxxB143C4 ----- Data received after first EOF on stream 17` These errors can be ignored when fax resources are involved. Performance is not altered significantly. These prints have been kept in the RTF logs because they help in debugging stuck channels when voice resources are involved (dx_rec, ec_stream).
Known (permanent)	IPY00032730	--		Runtime Trace Facility (RTF)	ERR1 in RTF log that appears at teardown of a process when Dialogic® DM3 Conferencing/MSI Libraries are involved can be ignored. The following prints may appear with an ERR1 label in the RTF log during the teardown of a process when DM3 Conferencing/MSI Libraries are involved: `04/17/2006 11:42:17.844   3948        2376 Cheetah                ERR1 Dm3MsgDispatcher      ERROR: No client attached to QComp (0:2:1:21:33)` In such a case, this is not an error. But this would be considered an error when any of the DM3 Voice Media Libraries are involved. These prints do not alter performance significantly. They are helpful in the debugging process and they only appear during the process teardown time.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known (permanent)	IPY00030548	24937		Springware Voice	If you call the **dx_reciottdata( )** function and you specify a DX_MAXSIL termination condition, **dx_reciottdata( )** ignores this condition and will eventually trigger on another one (DX_MAXTIME) even if there is silence on the line. Workaround: Set the EC_RESOURCE feature in DCM to ON.
Known (permanent)	IPY00022341	14325		Springware Voice	lineDial does not generate DTMF tones A, B, C, and D if specified dial string is in uppercase during inband dialing. Workaround: You must use lowercase letters a, b, c, d in a dial string that contains an escape string. The firmware only accepts lowercase letters a, b, c, d. The TSP provides the conversion for out-of-band dialing (default), but does nothing when an escape string is present in the dial string (in-band dialing). Therefore, any uppercase letters A, B, C, D used in a dial string that contains an escape string will be ignored by the firmware.
Known (permanent)	IPY00022151	23673		Springware Voice	DE_RINGS is received after the TDX_RECORD if the onhook recording begins while the ring is generated on the line. **Explanation:** When using analog Dialogic® Springware Boards, if a ring is generated on the line when you begin onhook recording, the DE_RINGS event will be received by the application after the TDX_RECORD event, i.e. after the recording is finished. The firmware was designed to process the rings in this manner and suspends the ring debouncer on issuance of commands like play and record. The debouncer is resumed only on completion of the play or record. Since the ring debouncer is suspended, the firmware cannot send a DE_RINGS event to the application. On resumption, the debouncer goes back to its last state and will send the ring event to the host.
Known (permanent)	IPY00022100	21760		Springware Voice	Calling **dx_setevtmsk( )** while a **dx_play( )** is running will cause the play to return with a TDX_ERROR, EDX_SH_BADCMD. **Explanation:** Calls to **dx_setevtmsk( )** on a channel that is currently playing a file causes an error to be returned and the channel to get stuck in an unstable state that requires the application to be shutdown and restarted to recover. Workaround: Call **dx_setevtmsk( )** initially with DM_DIGITS to enable DTMF digit events. DM_DIGITS should not be specified when making subsequent calls to **dx_getevtmsk( )**, for instance, when turning on/off silence events. This requires the application to be modified so that the DM_DIGITS flag is only passed to the function at initialization time.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Known (permanent)	IPY00037015	--		UDD	Dialogic® Diagnostics Software (UDD) reports download errors when multiple boards are installed in the same system. Workaround: When using UDD to test multiple Dialogic® Springware Boards in the same system, use DCM to disable all boards except the one being tested.
Resolved	IPY00032793	--	113	Board Deselection	For Dialogic® DM3 PSTN boards, DCBU switchover fails when the primary and standby D-channels are configured on different boards.
Resolved	IPY00039538	--	166	Board Detection	Error messages are seen in the Windows® Event Viewer indicating that the RTF server is not running. This is occurring because none of the Dialogic® services have a dependency configured on the RTF service.
Resolved	IPY00037356	--	148	Board Detection	DCM assigns the same physical slot ID to two boards (in different physical slots).
Resolved	IPY00033013	--	108	Board Detection	If the customer installs build 104 or later, and if they have Dialogic® CPI2400_1_T1 and CPI3000_1_E1 boards in their system, and they do a 'Restore Default' from the DCM GUI, the name will be changed to DF240_1_T1 and DF2000_1_E1, respectively. The change is only in the name displayed in DCM GUI, and there is no functionality difference. The same thing can happen if the customer uninstalls the existing build and installs the build 104 or later.
Resolved	IPY00040874	--	181	Board Download	Dialogic® DMV1200BTEPE Board fails to start on Service Update 155.
Resolved	IPY00038946	--	162	Board Download	Dialogic® JCT Media Boards download failed.
Resolved	IPY00038792	--	162	Board Download	Slow download times for Dialogic® JCT Media Boards on high-end machines.
Resolved	IPY00038074	--	154	Board Download	The OAMSYSLOG component reports multiple "DM3FDSP - GetOverlappedResult()[2] timeout for board 5, Error= 121" entries in RTF logs during load test.
Resolved	IPY00033228	--	154	Board Download	Cannot route voice device if it is not on the same board as the digital frontend device.
Resolved	IPY00035350	--	134	Call Control	While sending NonStandard Control data in an H.323 message, if the input string contains a byte with value 0x00, all the data after this byte will not be sent.
Resolved	IPY00034079	--	125	Call Control	After **gc_SwapHold( )** function successfully returns, both Global Call call states were reported as GCST_CONNECTED.
Resolved	IPY00028500	35390	63	Call Logging	The **cl_open( )** function is not working; it doesn't give a value to the parameter errno and always returns zero.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00007803	31782	18	Call Logging	The **cl_ReleaseTransaction( )** function has no effect on non-call-related transactions; memory keeps increasing.
Resolved	IPY00036423	--	142	Conferencing	Sometimes a noise is generated when a party leaves a conference; the noise disappears when a party is added to the conference.
Resolved	IPY00028633	35748	142	Conferencing	Sometimes a noise is generated when a party leaves a conference; the noise disappears when a party is added to the conference.
Resolved	IPY00009499	33932	142	Conferencing	A loud scratch/click sound occurs when entering a conference when 1-2 parties are already in the conference.
Resolved	IPY00007470	32437	142	Conferencing	A sharp noise occurs when changing conference resource mode to MSPA_MODERECVONLY.
Resolved	IPY00006707	33803	142	Conferencing	Sometimes a noise is generated when a party leaves a conference; the noise disappears when a party is added to the conference.
Resolved	IPY00021218	30986	--	Conferencing (CNF)	When using the Asynchronous programming model in a CNF application, API timeouts may be observed under certain heavy load conditions, particularly if the user is adding or removing parties or creating/tearing down conferences at a high rate.
Resolved	IPY00022229	25660	--	Conferencing (DCB)	When calling **dcb_setcde( )** to set the attribute of a conferee, the value MSPA_MODEFULLDUPLX cannot be ORed with any other MSPA_ value and has to have its own **dcb_setcde( )** called for it. This was resolved by updating the *Dialogic® Audio Conferencing API Library Reference.*
Resolved	IPY00038235	--	154	Configuration	The **dcb_dsprescount( )** function returns an incorrect value. It returns double the resources.
Resolved	IPY00036073	--	133	Configuration	Dialogic® DM/V and DM/V-A Boards cannot be configured for R2 protocol or any PDK protocol through the PDK Config property sheet in DCM on Service Update 118.
Resolved	IPY00035875	--	134	Configuration	**gc_Start( )** fails when an application was compiled.
Resolved	IPY00032796	--	108	Configuration	A blue screen occurs with a mini dump whenever the Dialogic® D/600JCT Board is configured with CTR4 protocol.
Resolved	IPY00006348	36782	103	Configuration	ML5BC on Dialogic® DMV3600BP Board incorrectly shows up as ML5B in DCM.
Resolved	IPY00038499	--	160	CSP	When using **ec_stream( )**, a completion event is never triggered back when using .wav recording (based on Win32 programming model).

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00034678	--	125	CSP	For Dialogic® D/41JCT-LS Board CSP application, **dx_playiottdata( )** fails after **ec_listen( )** to route any timeslot.
Resolved	IPY00031529	36814	108	CSP	The mapping of CSP channels is not being handled correctly on the Dialogic® D/120JCT Board during Earth Recall processing, which can lead to an assert in dslac_dd.c.
Resolved	IPY00011782	29318	--	CSP	TEC_VAD event does not occur when DM_VADEVTS is set by **ec_setparm( )**.
Resolved	IPY00010776	35105	63	CSP	The **ec_stopch( )** function does not always return a TEC_STREAM event.
Resolved	IPY00009001	34393	39	CSP	Echo canceled data transmitted over the SCbus to another channel using the CSP ExtraTimeslot feature still contains echo.
Resolved	IPY00008046	32435	18	CSP	There is no CSP extra timeslot assigned even though ExtraTimeslot is set to ON in DCM with the CSP firmware selected. As a result, **ec_getxmitslot( )** returns 0 and there is no way to share the echo canceled data with another channel through the SC/CT Bus.
Resolved	IPY00006862	36830	98	CSP	**ec_stream( )** returns -1 when running with a CAS protocol.
Resolved	IPY00030909	35327	70	CSP Demo	Running the VoiceDemo after running the CSPAuto demo gives the error message "Unexpected event received 0x89, error 96" after selecting play Vox.
Resolved	IPY00030605	25864	--	CSP Demo	For CSPdemoDM3 demo, the "wait for ring/wink" option (-w) does not work for Dialogic® DM/V-A Boards.
Resolved	IPY00020943	24719	--	CSP Demo	Typos in printf statements of cspdemo code may mislead troubleshooting.
Resolved	IPY00009423	32858	18	CSP Demo	CSPAuto demo fails to return TEC_STREAM event if more than one process is run per board.
Resolved	IPY00033410	--	111	D/120JCT-LS	When using five Dialogic® D/120JCT-LS Boards (CSP firmware), Dialogic® service will not start after a restart.
Resolved	IPY00032244	36750	108	D/240JCT-T1	A Dialogic® D/240JCT-T1 Board that is running NTT protocol incorrectly accepts the next incoming call while the previous call is not released by host.
Resolved	IPY00031535	36852	108	D/240JCT-T1	ISDN channel hang occurs when Dialogic® D/240JCT-T1 Board receives a STATUS message in a particular ISDN call state.
Resolved	IPY00009017	32209	18	D/240JCT-T1	When dualcall feature is enabled, firmware selects channel 1 for incoming call while the channel is being used for outbound call.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00008476	32065	18	D/240JCT-T1	With high call load, a firmware crash occurs on Dialogic® D/240JCT-T1 Boards running NTT protocol.
Resolved	IPY00012124	28389	--	D/300JCT-E1	Dialogic® D/300JCT-E1 Board crashes with an unstable E1 line.
Resolved	IPY00012757	27544	--	D/480JCT-2T1	When trying to configure the Dialogic® D/480JCT-2T1 Board for SoftFax capability, the *spfax.fwl* firmware file option is listed as available for “FirmwareFile” (span 1) category but not for “FirmwareFile2” (span 2) in their respective pull-down menus.
Resolved	IPY00011986	28826	--	D/480JCT-2T1	**dx_stopch( )** does not terminate call progress analysis when using DX_CAP parameter: ca_intflg = DX_PVDOPTNOCON, or DX_PVDENABLE, or DX_PVDOPTEN.
Resolved	IPY00030589	25101	--	D/600JCT-2E1	Windows® Device Manager fails to install Dialogic® D/600JCT-2E1 Board.
Resolved	IPY00030882	36057	108	D/82JCT-U	Dialogic® D/82JCT-U Board is not reporting rings to the application consistently.
Resolved	IPY00008169	31242	18	D/82JCT-U	Dialogic® D/82 firmware is not reading proper integration data from the 2-line display when integrated with vectors instead of hunt groups on a G3 switch. It only reads the second line of the display.
Resolved	IPY00008405	32026	18	D30EP	The Dialogic® D30EP Board should respond with REL COM message after receiving a SETUP message including the IBCAP message block.
Resolved	IPY00030597	29237	--	DCM	Dialogic® DM/V480-2T1 and DM/V960-4T1-PCI Boards are not detected by DCM on a Windows® 2003 system (.NET).
Resolved	IPY00028248	33718	154	DCM	The board and protocol descriptions for *ml10_dsa_net5.pcd* are incorrect in the DCM Assign Firmware File dialog box.
Resolved	IPY00019208	23343	--	DCM	ML2_120 is displayed in DCM instead of ML6_120 when configuring an E1 QS A with Media Load 6.
Resolved	IPY00019147	29041	--	DCM	Default setting of DISI32_R2_UK.config is mu-law even though UK uses A-law.
Resolved	IPY00014213	28818	--	DCM	When the RTF is activated and the DCM is stopped, the size of rtflog.txt generated becomes 0 KB.
Resolved	IPY00014103	28004	--	DCM	The DCM cannot load certain .tsf files properly to detect certain tones, especially when a tone is noisy and the volume is low.
Resolved	IPY00012759	27658	--	DCM	Under the Interface tab in DCM, users are allowed to put a blank in the place of ISDNProtocol and ISDNProtocol2 instead of “None” and this leads customers to wrongly believe that this is the method to download non-ISDN firmware.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00008135	29106	--	DCM	More than five boards cannot be downloaded and started through the DCM in some dual-processor chassis.
Resolved	IPY00028262	33199	56	Demos	**gc_GetCallInfo( )** fails to return the caller ID on a Dialogic® D/4PCIU Board when using the gc_basic_call_model demo.
Resolved	IPY00019091	29917	--	Demos	The demos under demos\voice directory have problems compiling.
Resolved	IPY00009633	33596	18	Demos	Ansrmt demo code does not include copyright notice, and execution of the demo prints “Dialogic” on the output screen.
Resolved	IPY00008676	33200	22	Demos	The Voice Demo fails with **gc_GetMetaEvent( )** failure when going offhook with an analog board, such as a Dialogic® D/120JCT Board.
Resolved	IPY00015573	28376	--	DI Boards	The number of tone templates defined in the config/fcd files should be less than 88 to guarantee no runtime problems. If more than 88 are defined at download, care should be taken to not exceed 128 total during runtime.
Resolved	IPY00009130	32103	56	DI Boards	The **ms_setvol( )** function fails intermittently on the Dialogic® DI/SI32 Board.
Resolved	IPY00008909	32265	56	DI Boards	When placing an outbound call on a Dialogic® DI0408LSAR2 Board trunk to an invalid number (operator intercept), a GCEV_CONNECTED event with positive answering machine detection (PAMD) is received, rather than a disconnect with SIT.
Resolved	IPY00008826	32458	56	DI Boards	In the following config/FCD files for the Dialogic® DI0408LSAR2 Board, the impedance setting required for the German stations is improperly set: disi*_r2_de.config, disi*_r2_de.fcd
Resolved	IPY00008283	32979	22	DI Boards	**gc_Stop( )** returns 0 (GC_SUCCESS) before TCP ports are effectively closed.
Resolved	IPY00007277	31912	56	DI Boards	When a Dialogic® DI0408LSAR2 Board is in a call and is the station party to hang up first, the POTS party hears a loud squeal for 2-3 seconds before the call is disconnected.
Resolved	IPY00014097	27655	--	DI/0408-LS-A	R4 High Performance libraries fail to return NAME or DATE for CallerID info on Dialogic® DI/0408-LS-A Boards.
Resolved	IPY00010139	33782	25	DI/0408-LS-A	When opening a resource, on connect a file is played (.vox), but when pressing a DTMF the play is not terminated. The DTMF is ignored and the file continues to play.
Resolved	IPY00010514	35342	154	DI0408LSAR2	**ms_genringex( )** fails to ring stations on Dialogic® DI0408LSAR2EU Board with UK ML3, and MSI device is left in a bad state.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00041178	--	178	Diagnostics	When Dialogic® System Release 6.0 PCI for Windows® is installed on a partition other than C, the d82diagutil application cannot locate the *voxcfg* file because it is looking for it on the C drive instead of the drive where the Dialogic® System Release is installed.
Resolved	IPY00037708	--	148	Diagnostics	The its_sysinfo tool, which is used to collect data including a PCI firmware dump, is not collecting a full memory dump.
Resolved	IPY00037643	--	160	Diagnostics	Using Visual Studio (V6 or 2005) to attach to a running process causes an access violation in LIBRTFMT.DLL.
Resolved	IPY00008420	32765	18	Diagnostics	Under certain call scenarios and when the application first starts, GetCallid errors are logged to DebugAngel. If not regularly checked, this file will grow. If this message is benign, it should not be logged.
Resolved	IPY00012765	28171	--	Dialogic System Service	Memory leaks with every Dialogic® System Start/Stop iteration. Repeatedly stopping and restarting eventually consumes all memory.
Resolved	IPY00041078	--	178	DM/IP Boards	Unknown audio or DTMF is being sent from a Dialogic® DM/IP Board at the beginning of a SIP call, which precedes the expected audio to be heard from the file played.
Resolved	IPY00038391	--	174	DM/IP Boards	Dialogic® DM/IP Board stops returning events due to a DSP failure.
Resolved	IPY00038190	--	166	DM/IP Boards	When running high volume load tests with Dialogic® DM/IP Boards to test SIP call control and media activity, the DM3 firmware reports data access exceptions from "Task:0x1993418 StatesTask" in DebugAngel logs. During this time, all active calls get suspended and performing media activity is not transmitted across the network to other end point.
Resolved	IPY00031560	36801	113	DM/IP Boards	When calling a Dialogic® DM/IP Board using G729a codec, the volume coming from the board will decrease when DTMFs are sent with RFC2833.
Resolved	IPY00031550	36859	105	DM/IP Boards	RFC2833 digits sent continuously from Dialogic® DM/IP Board.
Resolved	IPY00022013	29211	--	DM/IP Boards	For ipvs_evr_isdn_net5_307, using the IPML test series, DTMF/tone tests have about 80% failure rate in detecting the digits/tones accurately.
Resolved	IPY00021448	31633	22	DM/IP Boards	When using Dialogic® DM/IP Boards, the Type Of Service (TOS) byte cannot be set dynamically using either the Dialogic® Global Call API or the Dialogic® IP Media Library API.
Resolved	IPY00020968	28273	--	DM/IP Boards	An ARRAY FULL failure can occur on the PQ-II processor (SP13) on Dialogic® DM/IP Boards in a bulk SIP-call environment when more than 80% of the channels are being used on each board.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00020965	27459	--	DM/IP Boards	Application returns "synchronization object time out" error when calling IPML functions.
Resolved	IPY00020954	26090	--	DM/IP Boards	An IP call to a non-existent IP address returns "normal clearing".
Resolved	IPY00020951	25615	--	DM/IP Boards	In the iptmail_r4.dsp parser in the apppars.c module cannot handle 0D0D0A (CRCRLF) in the last line of the iptmail_r4.cfg, which causes iptmail_r4 to crash in **In_mailInitialization( )** function.
Resolved	IPY00020557	28561	--	DM/IP Boards	**gc_GetCTInfo( )** wrongly reports bus encoding as mu-law when called on a Dialogic® DM/IP Board configured for A-law operation.
Resolved	IPY00020556	28282	--	DM/IP Boards	When attempting a stop procedure using **gc_Stop( )**, the handlers are trying to be disabled without checking whether they were enabled in the first place.
Resolved	IPY00020545	29167	--	DM/IP Boards	Error messages in GC_H3R logs while running fax over IP (Non T.38).
Resolved	IPY00014105	28272	--	DM/IP Boards	When DMA is enabled on Dialogic® DM/IP Boards and a host to board or board to host DMA transfer is taking place, there is a chance that the transfer might not complete and the board would hang waiting for the completion to occur.
Resolved	IPY00012521	29111	--	DM/IP Boards	Downloading IPVSC firmware creates iptBxTy in the registry.
Resolved	IPY00010565	35077	56	DM/IP Boards	The Dialogic® DM/IP601-2E1-100 Board cannot start; it fails with error.
Resolved	IPY00009062	19233	--	DM/IP Boards	Trying to change media type on the TDM Bus from mu-law to A-law or vice versa with a T1 or E1 Dialogic® DM/IP Board in the system, produces error message "error calling CTBB_UserApply( )".
Resolved	IPY00009042	31632	22	DM/IP Boards	When using Dialogic® DM/IP Boards, the Type Of Service (TOS) byte cannot be set dynamically using either the Dialogic® Global Call API or the Dialogic® IP Media Library API.
Resolved	IPY00008839	33389	18	DM/IP Boards	RTP data sent over IP immediately after a fax CED call progress tone may be delayed and cause choppy audio at a receiving device. In a back to back configuration, the fax CED call progress tone is not always detected when using 30 ms G.711 coders.
Resolved	IPY00007640	30390	30	DM/IP Boards	When using Microsoft NetMeeting to call into the iptmail_r4 demo, connections fail or DTMFs dialed from NetMeeting are not being detected when using different coders on the Tx (local) and Rx (receive) sides.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00007472	28620	22	DM/IP Boards	Applications should not set the TOS bits through Dialogic® Global Call Software (Parameter = IPPARM_CONFIG_TOS) or Dialogic® IPML Software (Parameter = PARMCH_TOS). This may cause a blue screen.
Resolved	IPY00006801	33501	22	DM/IP Boards	The gc_h3r error entry for failing to retrieve a presentationRestricted should be changed to a level of WARNING.
Resolved	IPY00006731	31661	22	DM/IP Boards	For Dialogic® DM/IP Boards, if an error is returned when attempting to set TOS via Dialogic® Global Call Software, the error message is unclear. Error states: “Received IPMEV_Error during ipm_setParm on device ...: No Error.” Setting of TOS field works correctly. Only in cases where API fails (for any reason), the error message is not reported correctly.
Resolved	IPY00032019	25211	--	DM/V Boards	Out-of-Service messages are not sent out on a Dialogic® DM/V960-4T1-PCI Board in NFAS configuration.
Resolved	IPY00016138	27431	--	DM/V Boards	Dialogic® DM/V Board fails to download if call progress is set to “y”.
Resolved	IPY00016062	28905	--	DM/V Boards	Doing a **gc_close( )** on the 120th channel using R2MF, it may occasionally take approximately 8 seconds to go out of and back into service.
Resolved	IPY00015557	27160	--	DM/V Boards	Using Dialogic® DM/V960 (qs_ protocol) Board, **dx_rec( )** does not return immediately after the actual recording finishes.
Resolved	IPY00008962	32275	18	DM/V Boards	MAXSIL with CSP on Dialogic® DM/V Board is not working using the CSP stream.
Resolved	IPY00007390	31844	18	DM/V Boards	ISDN protocols mixed with DPNSS protocol in a single system will not work if the NetCRV feature is enabled for DPNSS through the registry setting.
Resolved	IPY00014898	26665	--	DM/V480A-2T1	Operator intercept outcomes are falsely detected as positive answering machine when doing call progress analysis with Dialogic® DM/V480A-2T1 Boards.
Resolved	IPY00019232	25922	--	DM/V-A Boards	The board type for the media load ml1_pcires.pcd, ml10_pcires.pcd, ml9b_pcires.pcd are identified as DM/V1200A-PCI although these are pcd files for Dialogic® DM/V2400A-PCI Boards.
Resolved	IPY00016183	28779	--	DM/V-A Boards	Channels are not set in service upon setting channel state when parm 1312 is set to 0 (default) in ml2_qsa_4ess.config.
Resolved	IPY00016180	28553	--	DM/V-A Boards	D channel will not recover if line taken out-of-service and put back in-service when the channel is connected.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00015512	29922	--	DM/V-A Boards	Download failures of *ml2_qsa_net5.pcd* and *ml2_qsa_qsige1.pcd* files on Dialogic® DM/V1200A-4E1-PCI Boards.
Resolved	IPY00015507	29486	--	DM/V-A Boards	**dx_playiottdata( )** does not return any events. This happens with Dialogic® DM/V960A and DM/V480A Boards.
Resolved	IPY00028649	36416	90	DM3 Admin	Blue screen crashes occur in dlgcmpd when using Dialogic® DM/V480A-2T1-PCI Boards on Windows® 2003.
Resolved	IPY00041233	--	178	DM3 Call Control	When a call is terminated in the GCST_DETECTED state, a fake GCEV_OFFERED event should not be generated if the application has enabled the GCEV_DETECTED event.
Resolved	IPY00041209	--	178	DM3 Call Control	GCEV_UNBLOCKED event doesn't arrive for individual channels, even though GCEV_BLOCKED was delivered to individual channels, after AIS alarms occur and are then cleared.
Resolved	IPY00037507	--	154	DM3 Call Control	Event API fails to deliver an event when the T1 is configured for CAS and the cable is unplugged.
Resolved	IPY00036504	--	144	DM3 Call Control	Calling **gc_MakeCall( )** causes a SETUP message to be sent. If the first response from the other side is CONNECTED, the board responds with CONNECT_ACK, but GCEV_CONNECTED is not sent to the application. The problem only occurs if the board is set to Network End; if the board is set to User End, GCEV_CONNECTED is sent.
Resolved	IPY00034857	--	148	DM3 Call Control	When performing call progress analysis via the Dialogic® Global Call media detection method, if the media detection occurs before the out-of-band CONNECT message is received, GCCT_UNKNOWN is returned as a result.
Resolved	IPY00039068	--	166	DM3 Conferencing	The **dcb_addtoconf( )** function returns failure, and ATDV_ERRMSGP shows the error message as "Timed out waiting for reply from firmware."
Resolved	IPY00037861	--	154	DM3 Conferencing	If one conferee goes on mute, other conference participants hear buzzing noise. **Note:** A documentation update to Section 6.7, [0x3b] Parameters (parameters 0x3b03 and 0x3b04) has been added in the Documentation Updates section for the Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide. Please refer to it for information relevant to this defect resolution.
Resolved	IPY00037817	--	154	DM3 Conferencing	When playing background music through the telephone set, music cuts are heard when party A speaks.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00037396	--	148	DM3 Conferencing	Static background noise trails voice in conferences with more than six parties.
Resolved	IPY00037373	--	154	DM3 Conferencing	In a conference with two parties, if party A keeps speaking while party B starts speaking, party B hears breaks from party A while party B is speaking.
Resolved	IPY00038317	--	155	DM3 Configuration	The Dialogic® DM/F240-T1-PCI Board is incapable of running T1 robbed bit protocols.
Resolved	IPY00006345	36788	98	DM3 Configuration	All *cas*.config files for T1 CAS protocols that come by default or get generated (for Dialogic® DM/V-B Boards) have all spans set for D4/B8ZS. However, most T1 lines in the field are configured for D4/AMI or ESF/B8ZS. This adds an extra configuration step because the default doesn't match either of the likely cases. **Resolution:** Based on feedback concerning standard configuration of switches, the default layer1 line coding parameter (0x1603) for the CAS loads was modified to AMI (from B8ZS). This change does not affect any of the other protocols. **Old:** SetParm=0x1603,**7** ! Coding (B8ZS=7, AMI=8) **New:** SetParm=0x1603,**8** ! Coding (B8ZS=7, AMI=8)
Resolved	IPY00041421	--	181	DM3 Fax	Fax channels may hang when a stop is issued at the end of a send fax page.
Resolved	IPY00041079	--	181	DM3 Fax	The **fx_rcvfax( )** function returns -1 error after the system is running for several days, and the system is not able to receive faxes.
Resolved	IPY00039661	--	174	DM3 Fax	**ATFX_RESLN( )** sometimes returns 0, which is an invalid value. (According to the documentation, the only valid values are 98 and 196.) **Note:** A documentation update has been added in the Documentation Updates section for the Dialogic® Fax Software Reference. Please refer to it for information relevant to this defect resolution. There are additional return values for **ATFX_RESLN( )**, and the values passed to **fx_rcvfax( )** and **fx_sendfax( )** have more options. (The defect number associated with the documentation update is IPY00040796.)
Resolved	IPY00039476	--	171	DM3 Fax	Stuck fax channels during inbound calls.
Resolved	IPY00038407	--	160	DM3 Fax	**ATFX_RESLN( )** sometimes returns 0, which is an invalid value. (According to the documentation, the only valid values are 98 and 196.)
Resolved	IPY00037467	--	154	DM3 Fax	Dialogic® DM3 fax channel hangs. DebugAngel reports two errors: "QERROR_KILLTASK" and then "QERROR_WARNING".
Resolved	IPY00037166	--	154	DM3 Fax	After an inbound fax call, the fax resource cannot go back to idle after **fx_stopch( )**.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00037161	--	154	DM3 Fax	With fax on Dialogic® DM/F240-1T1-PCI Boards, when the badscanline percentage exceeds the default value, the library and firmware both seem to send out RTN messages to request page re-send and retraining, but the RTN messages never get to the line. The result is that there is never a re-send, and the received image is distorted.
Resolved	IPY00032797	--	154	DM3 Fax	The fax sender cannot wait to receive retry of digital identification signal (DIS) message, and gets Phase E status (EFX_COMMERRTX) transmit communication error.
Resolved	IPY00031596	36840	98	DM3 Fax	A TIFF file will be faxed HI/LO resolution based on the TIFF header and not by the sndflag parameter for **fx_sendfax( )**.
Resolved	IPY00028607	36356	84	DM3 Fax	When using ml5_qsa_5ess firmware, a kill task occurs if the remote side tries to send a page with 24x32 width. When **fx_rcvfax( )** is called, it returns -1 with a TFX_FAXERROR, and the kill task occurs around 2 seconds later.
Resolved	IPY00028375	35507	63	DM3 Fax	When you implement a send fax and receive fax in one call by using turnaround polling, the polling bit is not updated when the receiving fax contains more than one page. This causes the **fx_rcvfax( )** function to complete with TM_POLLED instead of TM_FXTERM.
Resolved	IPY00028361	36091	70	DM3 Fax	There is a problem when sending a multi-page fax to a Dialogic® DM/V-B Board with UL3. It seems like the fax was received successfully from the sending side, but when opening the received tif file you only see the first page of the fax. However the size of the file is almost the same as the tif file sent.
Resolved	IPY00028349	35898	70	DM3 Fax	Dialogic® DMV1200BTEP Board fails to receive faxes sent from a Sharp UX-510A fax machine. The faxtrace utility reports an invalid image detected error, and a TFX_FAXERROR is reported to the user application and logged in RTF trace.
Resolved	IPY00028336	35991	70	DM3 Fax	There is a problem when sending a multi-page fax to a Dialogic® DMV600BTEP Board with UL1. It seems like the fax was received successfully from the sending side, but when opening the received tif file you only see the first page of the fax. However the size of the file is almost the same as the tif file sent.
Resolved	IPY00028326	34858	70	DM3 Fax	Intermittently, fax channels get stuck during fax reception. The DebugAngel log file reports this condition with the following error: “Stream Id 0 Data Size 0 Flags 4 Discarded.”

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00028278	35423	84	DM3 Fax	Fax sender (Dialogic® DM/F240 Board) cannot wait to receive retry of DIS message and gets Phase E status (EFX_COMMERRTX).
Resolved	IPY00028196	36256	84	DM3 Fax	TIFF (with valid tags) fails on Dialogic® CPi/2400 and DM/F Boards.
Resolved	IPY00016159	27903	--	DM3 Fax	When using the Dialogic® DM3 Fax E1 board to receive fax, the signal gain is too high. The DM/F300 Board fails to receive fax from some old models of fax machines. This does not happen with new models of fax machines.
Resolved	IPY00014110	28684	--	DM3 Fax	When sending a fax from a Dialogic® DM/F240 Board through a Dialogic® JCT front-end board, some fax pages show reversed data.
Resolved	IPY00009922	33056	18	DM3 Fax	When sending a multi-page fax to a Dialogic® DMV1200BTEP Board running UL1, it appears that the fax was received successfully from the sending side. However, when opening the received tiff file, you see that it only received the first page of the fax.
Resolved	IPY00033584	--	113	DM3 Firmware	Double ringback tone in China causes false cadence break on Dialogic® DM/V-A and DM/V-B Boards.
Resolved	IPY00031561	36775	108	DM3 Firmware	Intermittent blue screens occur when trying to shut down or reboot the OS with Dialogic® Boards downloaded.
Resolved	IPY00028658	36606	90	DM3 Firmware	While processing calls on ISDN lines with Dialogic® DM/V-B Boards, memory pool corruptions occur that make the firmware reject all incoming calls on the span with Circuit Not Available cause code.
Resolved	IPY00028557	36302	90	DM3 Firmware	**gc_BlindTransfer( )** fails after several days of normal operations. The system answers calls successfully, but blind transfer fails.
Resolved	IPY00028549	35901	154	DM3 Firmware	QERROR_WARNING messages appear in Dm3StdErr log, and then all channels lock up.
Resolved	IPY00028430	36333	88	DM3 Firmware	After running for several days, the driver side goes out of sync and the DM/V side reports AIS (blue alarm) error. The issue is seen only when **brd_SendAlive( )** feature is enabled.
Resolved	IPY00028417	35650	71	DM3 Firmware	ml2_qsa media loads (for example, ml2_qsa_5ess.pcd) do not support exit notification properly.
Resolved	IPY00028373	35431	62	DM3 Firmware	DCM doesn't start when using media load 10b and E1CC (clear channel) mode (ml10b_qsb_4_e1cc.pcd) on Dialogic® DMV1200BTEP Board.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00028345	35148	64	DM3 Firmware	D-channel fails to come up on a back to back setup. Two spans of NI2 are connected back to back and the D-channel comes up. When the cable is pulled out, the network side doesn't report the D-channel going down. When the cable is reconnected, the D-channel doesn't come up.
Resolved	IPY00028299	35430	84	DM3 Firmware	ANI is not returned when Screening Indicator is set under DMS protocol.
Resolved	IPY00028295	34503	56	DM3 Firmware	Application starts getting TDX_ERROR events after **dx_playiottdata( )**.
Resolved	IPY00028273	32842	70	DM3 Firmware	SETUP message without channel ID is not accepted by the firmware on Dialogic® DM/IP Board when the board is configured as Network side.
Resolved	IPY00028244	34159	84	DM3 Firmware	The 0x3925 parameter does not appear in the *ml9b_pcires.config* file.
Resolved	IPY00021322	29294	--	DM3 Firmware	Central Processor Failure is returned in the event log after making a few calls.
Resolved	IPY00010545	35967	74	DM3 Firmware	Qsig firmware does not send IE information to the application when the IE length is longer than normal.
Resolved	IPY00010472	34532	74	DM3 Firmware	When all 95 channels are enabled in NFAS back-to-back systems using gc_basic_call_model, the system hangs.
Resolved	IPY00010418	35572	63	DM3 Firmware	The **dx_stopch( )** function does not return TDX_PLAY(0x81).
Resolved	IPY00009611	33998	56	DM3 Firmware	Sometimes **dx_stopch( )** fails to terminate the voice activity (play or record), leading to a player or recorder stuck in a stopping state.
Resolved	IPY00009597	32651	18	DM3 Firmware	GCEV_ANSWER events are missing when using NI2 (ml2_qsa_ni2).
Resolved	IPY00009588	34915	74	DM3 Firmware	With pdk_us_mf_io protocol, using immediate start with wait for dial tone option, 25% of outbound calls fail with protocol error.
Resolved	IPY00009103	33425	18	DM3 Firmware	Static-sounding background noise trails voice in conferences with more than six parties.
Resolved	IPY00009080	33144	18	DM3 Firmware	GCEV_DROPCALL event is not always returned from **gc_DropCall( )** after call glare has occurred.
Resolved	IPY00008779	34575	56	DM3 Firmware	Dialogic® DM/V160LP firmware crashed running a load test.
Resolved	IPY00007819	33173	84	DM3 Firmware	Host ISDN state machine gets out of sync with switch after 4ESS RESTART messages.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00006647	36598	98	DM3 Firmware	There is memory corruption on the Dialogic® DMV1200BTEP Board when using ISDN firmware. The memory corruption is caused by any of the following conditions: • Facility IE in the inbound SETUP > 128 bytes • Several call scenarios causing a memory leak: - Incoming SETUP message with an active call on the specified bearer channel - Incoming SETUP message while the channel is BLOCKED (i.e., application has not issued **gc_WaitCall( )**) - SETUP glare condition - Disconnect glare
Resolved	IPY00011347	34397	56	DM3 Hardware	The general network interface alarm LED on the Dialogic® DMV1200BTEP and DMV600BTEP Boards is always on after DCM is started. Even after unplugging the E1 cable, there is no change; the LED is still on.
Resolved	IPY00038998	--	160	DM3 Host Runtime Library	Bipolar violation alarms are reported in LineAdmin, but are not reported programmatically via GCAMS.
Resolved	IPY00030905	34640	56	DM3 Host Runtime Library	When Service Update 30 is installed on the System Release 6.0 PCI Windows base release, if **ms_open( )** is called, it causes Microsoft Visual C++ Runtime Library errors. This does not happen when Service Update 30 is installed on a clean system.
Resolved	IPY00010478	33053	18	DM3 Host Runtime Library	During a hold glare scenario, the application never receives a GCEV_HOLDREJ event.
Resolved	IPY00009554	32913	18	DM3 Host Runtime Library	The application doesn't receive a GCEV_UNBLOCKED event after an alarm is cleared.
Resolved	IPY00008893	32725	18	DM3 Host Runtime Library	An ERR entry for circular buffer underruns occurs when the streaming to board feature is used on a Dialogic® DM/V960A-4T1 Board. This causes the rtflog file to roll over more quickly, thus losing valuable data.
Resolved	IPY00008422	33443	18	DM3 Host Runtime Library	The **dx_stopch( )** function does not return when called synchronously.
Resolved	IPY00006779	34516	56	DM3 Host Runtime Library	When **sr_putev( )** is used to add an event to the run-time library, if the parameter **evtlen** is set bigger than 512, the application receives an exception error.
Resolved	IPY00012770	28838	--	DM3 Install	Windows® device manager (WDM) unable to load dlgcdm3_nt4.inf for Dialogic® DI/0408-LS-A and DI/SI32 Rev 2 Boards.
Resolved	IPY00032239	36769	98	DM3 IP	There is a problem when using IPPARM_SIP_HDR to set call ID.
Resolved	IPY00031791	36793	98	DM3 IP	**gc_InvokeXfer( )** fails to send a SIP REFER message, and no event or failure indication is returned to the application.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00010914	36429	84	DM3 IP	If application uses both H.323 and SIP, **gc_ResetLineDev( )** called before **gc_WaitCall( )** locks up channel on Dialogic® DM/IP241-1T1 Board,
Resolved	IPY00010900	36349	98	DM3 IP	The PARM_TX_ADJVOL_ and PARM_RX_ADJVOL_ parameters in the *ipmlib.h* header file are defined via #defines but have semicolons after them, which make it hard to compile applications built using it.
Resolved	IPY00010760	36647	98	DM3 IP	When a call is placed to an IP address that does not exist or to a valid IP address that does not have a SIP phone active, you cannot call **gc_DropCall( )** to disconnect the call; you have to wait for the 64-second INVITE timer to expire before you receive a GCEV_DROPCALL.
Resolved	IPY00008651	32111	18	DM3 Media Span	When running high density Dialogic® DM3 Media Span systems, the Windows® Event Viewer may become populated with SRAM corruption errors. There will be no other system impact.
Resolved	IPY00031590	36755	154	DM3 Network	**gc_BlindTransfer( )** is not working on the Dialogic® DMV160LS Board.
Resolved	IPY00028555	36110	154	DM3 Network	With **ms_SetMsgWaitInd( )**, if the user of the Dialogic® DI Board station picks up prior to the function returning, it renders the device useless for up to 30 seconds.
Resolved	IPY00028516	35001	154	DM3 Network	Hook flash is sometimes not detected on Dialogic® DI Boards when it is issued from its station interface during the ring cycle.
Resolved	IPY00028444	35763	124	DM3 Network	GCEV_PROGRESSING message not sent to application.
Resolved	IPY00028408	35117	92	DM3 Network	Board crashes when using **gc_SetInfoElem( )** to add "Display Name" IE prior to calling **gc_AnswerCall( )** when connected to Nortel switch that has been configured for 5ESS.
Resolved	IPY00028406	35210	56	DM3 Network	PDKManager is encountering problems in registering the protocol for Dialogic® DM3 Boards, but the problems are not reported to the user and there are failures after the application starts.
Resolved	IPY00028293	35281	62	DM3 Network	The **gc_BlindTransfer( )** function is not working properly when using pdk_us_ls_fxs protocol with CSP_WaitDialToneEnabled = 1 and CSP_DialToneWaitTime =5000 (Default). If the blind transfer fails because dial tone is not available on the line, then **gc_BlindTransfer( )** should return a GCEV_TASKFAIL event after 5 seconds. However, this is not happening; the channel hangs without TASKFAIL.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00016070	29869	--	DM3 Network	After re-download of remote side, the local Layer2 still thinks that Layer1 is down and does not issue SABME request.
Resolved	IPY00010627	35339	74	DM3 Network	Using Dialogic® DMV1200B Boards with NFAS group of 5 spans, you cannot make calls from the 5th span when the NFAS group is on two boards.
Resolved	IPY00010593	35619	74	DM3 Network	Using the default *pdk_us_ls_fxs_io.cdp* file, the application is not able to detect the DISCONNECT tone that is defined in the .cdp file.
Resolved	IPY00009790	34269	56	DM3 Network	The **gc_AlarmSourceObjectNameToID( )** function doesn't return pass or fail when used with Dialogic® DM3 Boards; it throws an exception that the application either catches, or, if not programmed to catch, an application error is generated.
Resolved	IPY00009660	35169	74	DM3 Network	On Dialogic® DM/V-A and DM/V-B Boards with pdk_us_ls_fxs protocol and CDP_WaitDialToneEnabled =1, when blind transfer is initiated but no dial tone is available on the line, the **gc_BlindTransfer( )** function does not return and the channel hangs.
Resolved	IPY00009494	33772	30	DM3 Network	Outbound calls fail with "Out Of Order" error on Dialogic® DM3 Fax Board running GDK load.
Resolved	IPY00009300	34862	56	DM3 Network	With NET5 protocol, firmware does not send GCEV_PROGRESSING event to application upon reception of a PROGRESS message with unknown but syntactically correct event. Unless the message is incorrectly formatted, the event should always be generated.
Resolved	IPY00008659	32773	18	DM3 Network	The **gc_SetInfoElem( )** function does not allow you to set MLPP IE (0x41) nor does it allow CodeSet shift 5 (0x95) when using 5ESS or 4ESS on Dialogic® DM3 Boards.
Resolved	IPY00008391	31850	18	DM3 Network	The Connected Number Information Element is ignored in CALL CONNECT ISDN message.
Resolved	IPY00007916	32554	18	DM3 Network	When a span is set as NET5 network end and an incoming SETUP message comes in without a channel ID IE, the SETUP is ignored.
Resolved	IPY00007844	27539	22	DM3 Network	If a call is received on Q.931 where there is no channel ID in the SETUP message, the call is rejected by the firmware.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00007686	31991	84	DM3 Network	When configuring inter-board NFAS where the Primary and Secondary D channel are on separate boards, NFAS trunks on the board with the Secondary D channel cannot make or accept calls. However, NFAS trunks on the Primary D channel board (intra-board NFAS) are not affected and calls can successfully be placed. If the Data Link on the Primary D channel is taken down, the Standby D channel does not sucessfully take over and now NFAS trunks on the both boards cannot make or accept calls.
Resolved	IPY00007370	27563	18	DM3 Network	When using the DMS100 protocol, after a RESTART message is received, all B-channels are put Out Of Service and all inbound calls are rejected with Cause code 44, channel busy.
Resolved	IPY00007288	27764	18	DM3 Network	Outbound calls fail when the ALERTING message contains a Non-Locking Shift IE.
Resolved	IPY00007269	32571	27	DM3 Network	In a glare condition, a local DISCONNECT message with reason WRONG_MSG_FOR_STATE was received. The reason should be some other Q.931 cause code.
Resolved	IPY00038533	--	160	DM3 Runtime Libraries	An internal parameter is not decremented correctly when a process exits, causing failures in opening devices.
Resolved	IPY00040832	--	174	DM3 Voice	TEC_STREAM event is not returned to the application when **ec_stopch( )** is called after **dx_unlisten( )** is performed on that voice channel.
Resolved	IPY00040685	--	174	DM3 Voice	**ATDX_TRCOUNT( )** returns the wrong value when playing a GSM 6.10 WAVE file on Dialogic® DM3 Boards.
Resolved	IPY00039586	--	166	DM3 Voice	ERROR_BROKEN_PIPE error internal message is reported in RTF logs during a streaming to board play.
Resolved	IPY00039412	--	166	DM3 Voice	TDX_PLAY is not generated to the application during streaming to board play; **dx_GetStreamInfo( )** is not returning correct information.
Resolved	IPY00038981	--	166	DM3 Voice	TDX_PLAY is not generated to the application during streaming to board play; **dx_GetStreamInfo( )** is not returning correct information.
Resolved	IPY00038611	--	160	DM3 Voice	When using the **dx_playtone( )** function with TONEON or TONEOFF as the terminating condition, when the TONEON or TONEOFF event occurs, the program gets a TDX_ERROR event instead of TDX_PLAYTONE event.
Resolved	IPY00038435	--	160	DM3 Voice	Channels hang and are not able to recover once in a CS_STOPD state.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00037493	--	155	DM3 Voice	When running high volume load tests (500+ voice channels) for performing records, the RTF log shows “Buffer is corrupted” errors in the Dialogic® DM3 StreamSink component.
Resolved	IPY00037262	--	155	DM3 Voice	Under certain corner conditions, host CPU utilization increases a large percentage (15% witnessed) after issuing a record on multiple voice channels, and remains that way even after the record completes.
Resolved	IPY00037183	--	148	DM3 Voice	When recording WAV 176 bps file (11 KHz, 16 bits per sample), **dx_mreciottdata( )** stops prematurely with EOD before recording all bytes specified in io_length field of DX_IOTT structure, if this field is set to some large value (in this case, 26 Mb). Other formats, such as 64 kbs PCM MuLaw, ALaw, Linear, and ADPCM did not exhibit this problem.
Resolved	IPY00036861	--	144	DM3 Voice	When attempting to run transaction recordings under rapid succession, sometimes the internal CT Bus routing fails and the record returns with a TDX_ERROR event with the result “Switching Handler is not Present.”
Resolved	IPY00031562	36766	100	DM3 Voice	After a transaction record, routing another voice resource to another channel fails.
Resolved	IPY00028576	36197	84	DM3 Voice	**ATDX_BDNAMEP( )** does not work properly with Dialogic® DM3 Boards. When you request **ATDX_BDNAMEP( )** on the handler of dxxxB1C1 (of a DM3 Board), you receive an empty string instead of dxxxB1. When you request **ATDX_BDNAMEP( )** on the handler of dxxxB1 (of a DM3 Board), you receive the string dxxxB1.
Resolved	IPY00028527	36129	84	DM3 Voice	**gc_BlindTransfer( )** failed to return with GCEV_BLINDTRANSFER after calling **dx_getdig( )** (if digits are received).
Resolved	IPY00028421	35417	70	DM3 Voice	When using Dialogic® DM/V600BTEP Board, glitch can be heard in recordings done by **ec_stream( )** when another recording is being done on a neighboring time slot.
Resolved	IPY00028372	34427	56	DM3 Voice	Channel gets stuck in a stopping state during record operation, and **dx_stopch( )** doesn't stop the channel.
Resolved	IPY00028258	33717	71	DM3 Voice	When selecting ml10_dsa_net5.pcd for a Dialogic® DM/V600A-2E1-PCI Board, DCM displays ML2_60. It should display ML10_60 instead.
Resolved	IPY00015578	28841	--	DM3 Voice	“Device busy” message on voice resource after calling quickly **dx_stopch( )** on **dx_TxRxIottData( )**.
Resolved	IPY00015574	28393	--	DM3 Voice	**dx_TxRxIottData( )** ignores TPT maxtime termination when receiving FSK data.
Resolved	IPY00015360	25285	--	DM3 Voice	**dx_dial( )** with “T” parameter does not set digits to DTMF.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00014104	28121	--	DM3 Voice	Need a method to improve call progress analysis on Dialogic® DM3 Boards similar to that for Dialogic® Springware Boards.
Resolved	IPY00011063	36799	92	DM3 Voice	When using Dialogic® DISI32R2 Board, calling **ms_estconf( )** causes the following error to be displayed: "Unhandled exception in application.exe (NTDLL.DLL): 0xC0000005: Access Violation."
Resolved	IPY00010660	33502	18	DM3 Voice	Board resets when multiple parties are added and removed from conferences.
Resolved	IPY00009683	33685	56	DM3 Voice	**dx_stopch( )** cannot stop voice channel if run in thread.
Resolved	IPY00009525	32827	30	DM3 Voice	If an application calls **dx_addspddig( )** to assign DTMF values to increase and decrease speed, then these DTMF values do not show up in the digit buffer again until the board is redownloaded, even if the conditions are cleared by **dx_clrsvcond( )**.
Resolved	IPY00009433	34878	63	DM3 Voice	**dx_playiottdata( )** ignores the data length and plays until EoF, which sometimes causes noise if there is additional data after "data chunk."
Resolved	IPY00009279	33694	18	DM3 Voice	**dx_reciottdata(ASYNC)** returns 0 but fails to return any event, and after that, the channel does not respond to any commands.
Resolved	IPY00009231	32953	56	DM3 Voice	Accuracy of call progress analysis (PVD, PAMD) when using the default CPA qualification values for Dialogic® DM3 Boards needs to be improved.
Resolved	IPY00008770	29169	--	DM3 Voice	Time out waiting on reply from firmware when ms_setstparm(MSSP_STPWR, MS_PWROFF or MS_PWON) command was issued.
Resolved	IPY00008559	32510	18	DM3 Voice	The fcdgen utility reports an error when changing the default PVD/PAMD qualification parameters.
Resolved	IPY00007872	33351	18	DM3 Voice	The **dx_playtoneEx( )** function stopped after 40 repetitions; it did not play as long as defined.
Resolved	IPY00019150	29550	--	DMV160LP	**dx_playiottdata( )** never returns any event.
Resolved	IPY00016174	28266	--	DMV160LP	Debug messages that shouldn't be there show up in the DebugAngel viewer.
Resolved	IPY00016064	28972	--	DMV160LP	Caller ID is not received reliably 100% of the time. Upon caller ID failure, the application receives an empty caller ID string.
Resolved	IPY00008722	29103	--	DMV160LP	PBXs that provide excessive leakage current within 300 msec after a disconnect may trigger a false ring indication.
Resolved	IPY00008233	28782	--	DMV160LP	**dx_getfeaturelist( )** indicates that CSP is supported on the 5th virtual board (which contains the fax resources), but CSP functions cannot be called on these resources.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00015228	31100	--	DMV160LPHIZ	The functions **cl_StartTrace( )**, **cl_StopTrace( )** and **cl_DecodeTrace( )** are not supported on Dialogic® Analog HiZ boards. If the buttons to "Start Trace", "Stop Trace" or "Decode Trace" are selected from the Sniffer menu bar of the SnifferMFC demo while running with the Dialogic® DMV160LPHIZ Board, error code "0x3006 function not supported" will be returned. This information will be added to the *Dialogic® Call Logging API Software Reference*.
Resolved	IPY00015226	31072	--	DMV160LPHIZ	In the SnifferMFC.EXE, the ability to select TSC as an argument for **cl_Open( )** was omitted. The problem can be resolved by recompiling SnifferMFC from source.
Resolved	IPY00015217	30280	--	DMV160LPHIZ	Fax tones can't be reliably detected if PVD and PAMD detection are disabled while call analysis is enabled. To ensure that fax tones can be detected for call analysis, make sure that PVD and PAMD are enabled.
Resolved	IPY00011464	30206	--	DMV160LPHIZ	Collection of dialed digits is not supported for outbound calls.
Resolved	IPY00008173	30959	--	DMV160LPHIZ	The **cl_GetMessageDetails( )** function returns an "<unknown>" message text for the **pszName** instead of displaying the proper message text in the cases of "CallInfo_CallerName" and "CallInfo_DateTime". This issue pertains to Dialogic® Analog HiZ Boards only.
Resolved	IPY00036280	--	142	Fax	When a Dialogic® VFX/41JCT-LS Board is receiving fax when the line quality is not good, sometimes the calls are terminated by error with ESTAT 193.
Resolved	IPY00034495	--	131	Fax	Firmware crash occurs when certain TIFF file is sent from one channel in MH, 9600 MSLT=10ms condition.
Resolved	IPY00034105	--	125	Fax	Dialogic® VFX/41JCT-LS Board channel is unable to send/receive fax after particular fax call scenario occurs.
Resolved	IPY00031534	--	139	Fax	When sending a fax, the Dialogic® VFX/41JCT-LS Board cannot establish phase B with some particular fax machine.
Resolved	IPY00010370	34054	27	GDK	GDK functionality is not operational. GDK channels cannot be detected.
Resolved	IPY00036418	--	139	Global Call	On Dialogic® DM3 Boards, the **gc_Open( )** function does not cause sabmr messages to be sent for the DPNSS protocol.
Resolved	IPY00034841	--	134	Global Call	While closing the H.323 channels, some may be stuck in an intermediate state causing the subsequent events to be directed to incorrect devices.
Resolved	IPY00032263	36681	92	Global Call	The **gc_SetupTransfer( )** function fails when calling far end, ring no answer.
Resolved	IPY00032022	55549	--	Global Call	Incorrect cause returned for event GCEV_BLOCKED.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00031588	36770	111	Global Call	Problem with **gc_HoldCall( )**, which sends the SUSPEND message to the network.
Resolved	IPY00028579	34569	70	Global Call	**gc_RetrieveCall( )** returns error “function not supported in this state” when using pdk_sw_e1_ntmd_io.cdp.
Resolved	IPY00028560	36335	84	Global Call	**gc_SetupTransfer( )** fails with invalid line device.
Resolved	IPY00028530	36371	87	Global Call	When **gc_Start( )** fails, **gc_ErrorInfo( )** cannot be used to retrieve the error code. **gc_ErrorInfo( )** fails with an error indicating that **gc_Start( )** has not been issued. The correct behavior is for **gc_ErrorInfo( )** to execute successfully and return the error code and description.
Resolved	IPY00028492	35458	62	Global Call	**gc_SendMoreInfo( )** is failing when using PDK Argentina on Dialogic® DM3 Boards.
Resolved	IPY00028478	35825	65	Global Call	GCST_ONHOLD state is not returned as documented after a successful **gc_HoldCall( )**; instead the incorrect GCST_CONNECTED state is returned.
Resolved	IPY00028446	35330	65	Global Call	Dialogic® Global Call Software does not have result values for certain SIT tone terminations when performing call progress analysis using PDK protocols.
Resolved	IPY00028416	35839	70	Global Call	Synchronous calls to **gc_WaitCall( )** cause access violation error upon exit of the function.
Resolved	IPY00028384	35875	65	Global Call	**gc_MakeCall(SYNC)** returns -1 with an undocumented error code when an operator intercept is received. The problem occurs when dialing a number whose results terminate with SIT.
Resolved	IPY00028207	36310	84	Global Call	**gc_CompleteTransfer( )** does not complete successfully. Error returns: “Function not supported in current state.”
Resolved	IPY00021487	24364	--	Global Call	Using **dx_deltones( )** in connected state causes **gc_DropCall( )** to hang. This was updated in the *Dialogic® Global Call Analog Technology Guide*.
Resolved	IPY00019243	25537	--	Global Call	**dx_close( )** fails if called after **gc_close( )** in a Dialogic® R4 Library application that uses **gc_attach( )**.
Resolved	IPY00019149	29335	--	Global Call	Running a modified gc_basic_call_model application (play/record features added), for the first time there are no problem. However, after ending the application and running it a second time, errors are seen in the DebugAngel log.
Resolved	IPY00016068	29758	--	Global Call	If an application exits ungracefully two times in a row, upon a third application execution, **gc_MakeCall( )** always returns with GCEV_DISCONNECTED.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00016066	29040	--	Global Call	GCEV_OFFERED event received on a board device (DTIB1) causes application to crash when using overlap receive.
Resolved	IPY00015514	30170	--	Global Call	A NON RECOVERABLE FATAL ERROR occurs if a call is dropped at the remote-end in the ACCEPTED state. After that, the port is dead and there is no way to feasibly recover.
Resolved	IPY00010364	35130	56	Global Call	The GCEV_DIALING event is being reported too early when using PDK CAS/R2 protocol on Dialogic® DM3 Boards.
Resolved	IPY00010129	34274	56	Global Call	Dialogic® Global Call Software does not provide a way to disable DISCONNECT TONE SUPERVISION with pdk_na_an_io.cdp.
Resolved	IPY00010035	35159	56	Global Call	Under certain conditions when a **gc_MakeCall( )** attempt times out, it incorrectly displays the result message as NORMAL CLEARING instead of timeout.
Resolved	IPY00009673	33519	18	Global Call	After many successful outbound calls, a **gc_MakeCall( )** fails with GCEV_DISCONNECTED (gc_msg=Event caused by protocol error, cc_msg=Pcikup). After that, all subsequent calls on that network device fail with the same reason. The problem gets cleared only after the device is closed and reopened.
Resolved	IPY00009517	33543	18	Global Call	GCAMS incorrectly reports the DCHAN_CFA alarm as a DTE1_LOS alarm when the D-channel is down.
Resolved	IPY00009462	34121	39	Global Call	GCEV_FATAL_ERROR events occur frequently when using R2 PDK protocols, and channels can be lost until the boards are redownloaded.
Resolved	IPY00009457	32846	18	Global Call	The **gc_DropCall( )** function fails to send a completion event when an alarm occurs right after the function is issued.
Resolved	IPY00009131	32810	18	Global Call	Running a Dialogic® Global Call program with both analog and digital line devices enabled will cause an error if the program also opens a digital board device. The following error message is generated: Insufficient number of functbls.
Resolved	IPY00009094	33816	56	Global Call	When disconnecting and reconnecting spans running the pdk_mx_r2_io protocol, GCEV_FATAL_ERROR events occur.
Resolved	IPY00009024	32014	18	Global Call	When performing call progress using the **gc_MakeCall( )** function on Dialogic® DM3 Boards, the TSP does not distinguish between the CaNoRingback and CaNoAnswer responses.
Resolved	IPY00008293	35190	56	Global Call	When **gc_MakeCall( )** is issued under PDK CAS, with CPA parameters specified (GC_PARM_BLK), an access violation occurs.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00008150	33069	18	Global Call	**gc_MakeCall( )** changes the CRN value after the permanent signal timer expires.
Resolved	IPY00007743	30029	--	Global Call	Sometimes the first **gc_MakeCall( )** under T1 ISDN with R4 will not complete successfully.
Resolved	IPY00006856	36800	98	Global Call	When a board device is closed while time slots are still open on that trunk, the board device fails to open when the application tries to open the board again. The **gc_OpenEx( )** to reopen the span fails with an Invalid linedevice error.
Resolved	IPY00006790	35137	131	Global Call	For outbound Dialogic® Global Call SS7 calls with dialstring *1234, the leading * is stripped and replaced with a trailing 0 (i.e., 12340) causing call to fail.
Resolved	IPY00006654	36085	84	Global Call	Using Dialogic® Global Call SS7 protocol, when ISUP sent SUSPEND and RESUME message, the Global Call library did not generate a GCEV_RETRIEVECALL event.
Resolved	IPY00007242	28807	--	Global Call on IP	Applications using Dialogic® Global Call over IP cannot switch from voice to T.38 fax after receiving CNG/CED tone. The remote side can initiate a switch from voice to T.38 by either sending a RequestMode for H.323 or a ReInvite for SIP. Currently, Global Call does not support the sending of RequestMode or ReInvite.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00014904	28798	--	Global Tone Detection	If you have trouble detecting tones and see “Can't allocate space in suspect list for GTD(x)” in the DebugAngel output, consider the following: • Seeing this message is not necessarily a problem. If you are properly detecting tones of interest, this message can be safely ignored. It is generated when too many tone frequency ranges overlap the signal actually being received. Because the commonly used, predefined tones (e.g. DTMF tones and Call Progress), are given precedence on the list, detection of standard tones is not compromised. • You may wish to review any user-defined tones to ensure that they do not duplicate predefined tones. While detection will not be impacted, the tracking of multiple definitions of the same tone is inefficient. If you cannot detect a user defined tone and are seeing the above message, it usually means the newly defined tone overlaps a frequency range that is already used by several existing tones. This can often be remedied by taking one or more of the following steps: • Check for duplication of tones. Are you adding tones that differ by only small values? Are you duplicating existing default tones? If so, try to consolidate. Use or redefine existing tones instead of creating new ones that differ only slightly. • Check for unnecessarily wide frequency tolerances. Try to use ranges only large enough to meet requirements. Very large ranges do not improve detection, and can negatively impact efficiency. • When you have discretion as to what frequency areas to use for your user defined tones, try to choose a range that is little used by standard tone definitions. The above suggestions will improve detection efficiency even when no errors are observed.
Resolved	IPY00037372	--	154	H.323 Call Control	An access violation/assert is seen in the Dialogic® Global Call IP Call Control Library if a RequestMode message for changing audio codecs is received.
Resolved	IPY00037351	--	154	H.323 Call Control	When the remote capabilities contain one audio codec and T.38 fax codec, the Dialogic® Global Call IP Call Control Library will incorrectly attempt to switch to fax.
Resolved	IPY00030591	25747	--	HDSI Boards	Rebooting a Dialogic® HDSI system with DCM set to autostart will occasionally fail to come up with fatal error in MSILineActivate.exe.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00019274	27450	--	HDSI Boards	Dialogic® System Release 5.1.1 does not include proper CONFIG/PCD/FCD files for configuration of MSI/1200 (HDSI) to be used with SIB units less than 120 ports.
Resolved	IPY00008337	33011	22	HDSI Boards	Modifications to the ring frequency in the HDSI CONFIG file do not take effect.
Resolved	IPY00031597	36527	110	Host Admin	Autodump leaves the board in an unknown state when it fails to download diagnostic firmware.
Resolved	IPY00030913	34816	71	Host Admin	**NCM_GetVersionInfo( )** reports incorrect values for the DSS version information.
Resolved	IPY00030886	31675	62	Host Admin	When PCI bus number of a board is 0, DCM shows the value as 0x00, but when the bus number is non-zero, DCM shows the value in decimal format. This is inconsistent. (The zero value is now shown as a decimal value as well to make them all consistent.)
Resolved	IPY00030885	35102	71	Host Admin	The computer screen goes “blank” when the Dialogic® service is starting. The machine cannot be operated from the local terminal. The blank screen cannot be recovered until the video mode is altered (using remote control software).
Resolved	IPY00030595	27307	--	Host Admin	The perfcctl program fails to start and prevents the enabling of Dialogic counters.
Resolved	IPY00028511	36316	84	Host Admin	Dialogic® DISI32R2 Board failed to start with DISI32_R2_UK and DISI_R2_AU FCD/PCD files.
Resolved	IPY00028442	35573	63	Host Admin	The **brd_SendAlive( )** API feature to allow for watchdog alarms on spans throws an exception when enabled.
Resolved	IPY00028407	35620	74	Host Admin	The **ATDV_SUBDEVS( )** function fails on the Dialogic® DI0408LSAR2 Board due to a device mapping issue. The application gets a “Timed out waiting for firmware” error message.
Resolved	IPY00020547	29189	--	Host Admin	Error message “rAddExt (VT Tree) Array Full (124725)” printed on the screen when running with Slow Start.
Resolved	IPY00020546	29188	--	Host Admin	Error messages in GC_H3R logs when trying to run with Slow Start.
Resolved	IPY00012764	28169	--	Host Admin	CTBB services assign two ports UDP and TCP that just listen. These port numbers are randomly assigned and that makes it hard to leave these port numbers open.
Resolved	IPY00010860	35438	62	Host Admin	After upgrading a PCI RAID controller, Dialogic® System Service does not start automatically. The user must re-detect and reconfigure hardware in order to start services.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00010226	36092	74	Host Admin	In the DCM Trunk Configuration tab for Dialogic® DM/V-B Boards, the Media Load and Trunk protocol values listed in the pull-down menu at the bottom do not show the currently selected value at the top when each is highlighted.
Resolved	IPY00009466	32615	18	Host Admin	Only administrators should have write access to the RTF config file.
Resolved	IPY00009426	32539	56	Host Admin	When performing a “Restore Device Defaults,” the service startup mode gets reset to automatic, which can cause problems with customers expecting the state to remain at Semi-Automatic or Manual.
Resolved	IPY00009305	34805	56	Host Admin	If you reboot the system without stopping the Dialogic® System Service, an error occurs when trying to start the system again.
Resolved	IPY00009263	33385	56	Host Admin	When a Dialogic® DM/V-A or DM/V-B Board is shut down in DCM, an error event is generated in the Windows® Event Viewer. The error message is “dwCheckPoint=6”. The error event can be ignored, since boards can be restarted without error and applications can be run without a problem.
Resolved	IPY00008881	33156	18	Host Admin	In certain systems, the transmit timeslot information is not correctly assigned for a few devices on a board. For example, **dx_getxmitslot( )** will return -1 on certain devices, or the output from devmapdump will not show the correct transmit timeslot information for certain devices, while other devices on the same board will have correct transmit timeslot information. There is a high probability that this problem occurs on the first few devices on a board.
Resolved	IPY00008308	32313	62	Host Admin	**NCM_GetVersionInfo( )** reports incorrect version information. This occurs when getting “About” information in DCM as well as when retrieving the information through the API.
Resolved	IPY00008243	35013	56	Host Admin	Using **dt_xmitwink( )** toggles abcd bits from 0000->1111 instead of 0000->1010 as in previous releases.
Resolved	IPY00007997	31583	30	Host Admin	When starting and stopping services, DCMOBJ.EXE memory usage and handle count increase continually and never get deallocated. This also happens when using the NCM API and polling to see if services have started up.
Resolved	IPY00007715	32343	25	Host Admin	In systems with Dialogic® Springware Boards only, a dialog box with an error message is displayed when accessing any option under Settings->System/Device Autostart of DCM. The error message states “Failed to set Device Autostart setting”, and “Requested data not found in NCM data storage.”
Resolved	IPY00007352	31530	18	Host Admin	Uninstall cleanup utility does not report whether it passed or failed.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00030671	32696	18	Host Drivers	After the Driver Verifier finds an obsolete call used in the device driver, a blue screen occurs.
Resolved	IPY00030669	32557	18	Host Drivers	A blue screen occurs during the boot verification tests performed by the Windows® kernel using Verifier.exe.
Resolved	IPY00028477	35170	62	Host Drivers	Blue screens occur after >2 hours of load test when more than one Dialogic® DMV1200B Board is installed.
Resolved	IPY00028390	35768	63	Host Drivers	A blue screen occurred after running an application for 2-3 hours with a Dialogic® DMV600BTEP Board.
Resolved	IPY00028354	34032	56	Host Drivers	When the Dialogic® D/480JCT Board is assigned an interrupt of 45, it causes genload to fail.
Resolved	IPY00009527	34921	56	Host Drivers	A blue screen occurred after stopping the system running on a Q10000 chassis.
Resolved	IPY00009015	32108	18	Host Drivers	The **dx_stopch( )** function does not return with a completion event on several channels.
Resolved	IPY00032258	36810	95	Host Install	Silent install is flagging itself as completed even though it still requires a reboot.
Resolved	IPY00030892	35704	65	Host Install	The cleanup utility does not remove the IPMedia service.
Resolved	IPY00028521	36081	70	Host Install	The QSB-U2 media load is not available in the pull down menu in the Trunk Configuration tab of DCM under the media load selections for the Dialogic® DMV1200BTEP Board.
Resolved	IPY00028506	36209	74	Host Install	Service Update 65 cannot be installed as an update install. When you Start Services, DCM gives the errors “Failed to Detect Boards”, “Error Configuring the TDM Bus”.
Resolved	IPY00028472	36043	70	Host Install	After upgrading from Service Update 62 to Service Update 64, the Registry keys and the DCM About dialog box still indicate SU 62.
Resolved	IPY00019757	31086	--	Host Install	When installing the Dialogic® System Release Software, pressing F1 for online help on any of the install screens results in an error. To access online help, go to the root directory on the CD and double-click on install.hlp. This error message does not negatively affect the rest of the software installation; all files are properly installed.
Resolved	IPY00012763	28122	--	Host Install	Uninstalling Dialogic® System Release 5.1.1 will cause an existing CT-Connect install to fail.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00010161	33665	18	Host Install	Installation of the Dialogic® System Release Software fails when the Windows® PATH environment variable is not defined. After making all install selections, and "Next" is clicked to begin the copy of files, a pop-up box marked "Severe" appears reporting the following error: "Failed to set the environment variable (path)". The install then aborts.
Resolved	IPY00010136	33019	56	Host Install	When Dialogic® System Release 6.0 PCI for Windows® is installed on a partition that is not drive C, the first thing the install script does is to delete any existing system release folders in drive C. It does not delete the existing system release in the specified install drive.
Resolved	IPY00008769	32441	18	Host Install	Files are left on the system after doing an uninstall.
Resolved	IPY00008078	33939	56	Host Install	If the user chooses to back up data during the uninstall, an error is generated and the data is not backed up. This issue only affects the Config file migration; DCM and GC migration are OK. The uninstall continues after being prompted for an action, and the uninstall completes.
Resolved	IPY00007781	32411	18	Host Install	Boards can't be started when UNC names are used during the install for directory paths.
Resolved	IPY00007620	31945	18	Host Install	During the install using Terminal Server, an error pops up for GDKInf.exe.
Resolved	IPY00007575	29790	--	Host Install	The installation screen has incomplete instructions and the help is incorrect.
Resolved			62	Host Install	When performing an **update install** (not a full install), if an INF file change comes in, the board's configuration in the registry is not updated. The binaries that use the new INF file are installed, but the new INF information is not updated to the registry. Error messages may pop up when trying to configure a board, and as a result the board configuration will fail.
Resolved	IPY00038849	--	160	Host Library	When opening channels asynchronously with **gc_open( )**, sequentially one after another channels fail to open.
Resolved	IPY00032265	36780	95	Host Library	The Dialogic® Standard Runtime Library (SRL) seems to get into a "hung" state, causing event and IO to stop.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00032262	36688	95	Host Library	**sr_waitevtEx( )** hangs in multi-threaded environment. For example, if you have an application that creates two threads, each thread monitors its own events via **sr_waitevtEx( )**. The first thread makes a call, synchronously. The second thread makes a call, synchronously. Before the second call is connected, the first call is disconnected. At this time you do not receive a CCEV_DISCONNECT for the first call. The **sr_waitevtEx( )** call just hangs for the first thread.
Resolved	IPY00032247	36698	95	Host Library	D-channel remains down after setting the Logical Data Link State.
Resolved	IPY00031587	36666	94	Host Library	**gc_GetLinedevState( )** can return the wrong state for the D-channel because at times the GCEV_D_CHAN_STATUS event gets posted before the D-channel state is updated in the library.
Resolved	IPY00030907	34175	56	Host Library	Calling **sr_getboardcnt( )** with DEV_CLASS_DCB returns 0 boards with conferencing load.
Resolved	IPY00028597	36108	84	Host Library	When the completion event for **gc_MakeCall( )** results in GCEV_TASKFAIL, the application then uses the CRN returned from the **gc_MakeCall( )** to issue a **gc_DropCall( )**, **gc_ReleaseCall( )**. Both of these functions fail with "invalid CRN".
Resolved	IPY00028592	36295	84	Host Library	RESTART messages change the maintenance state of a channel if the channel was IN SERVICE when the message arrived.
Resolved	IPY00028542	36633	92	Host Library	Access violation occurred with **sr_putevt( )/gc_GetMetaEvent( )**.
Resolved	IPY00028514	35412	62	Host Library	Setting MEDIA_TYPE_DETECT flag in PDK_MAKECALL_BLK on Dialogic® Springware Boards causes the GCEV_CONNECTED event to indicate GCCT_INPROGRESS as expected, but a GCEV_MEDIADETECTED event is never received. This prevents enabling/disabling call progress analysis on a call-by-call basis on Springware Boards.
Resolved	IPY00028452	35597	63	Host Library	Problems with libdtimt.dll cause the **cc_GetDLinkState( )** function to fail when the program is compiled for Service Updates after SU 58.
Resolved	IPY00028334	35134	62	Host Library	Transaction record occasionally doesn't return any data, even though the application is performing the same sequence of events as in a successful transaction record.
Resolved	IPY00016067	29539	--	Host Library	No event generated when CO sets B channel(s) out of service. Application keeps calling on blocked channels.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00010556	35157	56	Host Library	Calling **ATDV_SUBDEVS( )** on Dialogic® DM3 MSI stations while alarms are being processed on DM3 trunks on span cards can cause a deadlock condition in the application, which can hang the system for 8 seconds.
Resolved	IPY00009758	32759	63	Host Library	Calls to unsupported or unimplemented functions on Dialogic® DI/0408-LS-A Boards are resulting in an incorrect error code being generated.
Resolved	IPY00009004	32303	18	Host Library	An application may see a deadlock while using Streaming to Board. For Streaming to Board, an application calls **dx_PutStreamData( )** to put data in the Circular Stream Buffer. As a result of issuing this API, if the total size of the data in the Circular Stream Buffer happens to cross the high water mark set by the application, a process deadlock may occur between the application thread calling the **dx_PutStreamData( )** and an internal thread in the library.
Resolved	IPY00008151	33070	18	Host Library	Problems occur with rtf logging; enabling modules in 'RtfConfigWin.xml' does not log any information to the *rtflog.log* file.
Resolved	IPY00007995	32188	30	Host Library	Any ODI error in the *rtflog.txt* file that has an error code of 0x2801e can be ignored, since this is not a functional error. This error only indicates that the queried Dialogic® DM3 component does not exist. This is an expected error if the component being queried does not exist in the firmware due to the pcd file downloaded.
Resolved	IPY00037004	--	142	IP	IP trunks hang due to missing **gc_AnswerCall( )** event.
Resolved	IPY00032664	--	105	IP	RFC2833 DTMFs not detected by Dialogic® DM/IP241 Boards.
Resolved	IPY00011037	36677	98	IP Host	Inbound fax call fails. This happen when previous call on the same device is dropped and media devices are disconnected using **gc_SetUserInfo( )**.
Resolved	IPY00034413	--	134	IP Media Session Control (RTP)	Parameter checking behaves inconsistently when calling **ipm_StartMedia( )**.
Resolved	IPY00035506	--	131	ISDN	An ISDN call disconnects during the ACCEPT state. When this occurs the application does not get a CCEV_DISCONNECT event.
Resolved	IPY00030001	36796	118	ISDN	ISDN traces not functional for NI2 and QSIG on Dialogic® DM3 Boards.
Resolved	IPY00016178	28372	--	ISDN	Dialogic® DM3 ISDN firmware formats all outbound messages with the Interface ID bit set to 1. This has resulted in both incoming and outgoing calls being rejected.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00016168	28084	--	ISDN	When performing ISDN tracing on all spans in a bulk call environment, dual span boards can encounter firmware crashes with just one board running.
Resolved	IPY00007887	29513	--	ISDN	Under heavy load, firmware assert happens in vb_util.c when working with ISDN protocols.
Resolved	IPY00008535	31333	18	ISDN Protocols	The 5ESS protocol firmware on Dialogic® JCT Boards does not handle incoming STATUS messages with a cause value “Response to Status Enquiry” properly.
Resolved	IPY00009183	32625	18	ISDN Q.SIG Protocol	When the service provider sends a STATUS message carrying cause value 100 (invalid information element contents), the board responds with an immediate disconnect.
Resolved	IPY00037319	--	160	JCT Call Control	If a board running ISDN 4ESS receives a CALL PROGRESS message in which the LOCATION information element in the Progress Indicator is 1010 - Location (network beyond interworking point), it sends back a STATUS message to the switch with Cause Value 100 (Invalid Information Element Contents).
Resolved	IPY00028305	32144	74	Media Voice Library	When the system is under heavy load, it’s been observed that anywhere between 2-100 hours, a Voice channel fails to return a completion event TDX_RECORD while doing a Record operation. The application even on calling **dx_stopch( )** does NOT recover the channel and the channel is stuck. This problem happens on a heavily loaded system and only affects record operation on a Voice channel. Play operations work fine. If the application then calls **ATDX_STATE( )** to examine the state of the channel and for all subsequent calls, the value returned is 7 (“CS_STOPD”).
Resolved	IPY00040179	--	171	Modular Station Interface (MSI)	SRL_TIMEOUT_ERROR occurs after upgrading from SU 154 to SU 166. (Results in **ms_listen( )** / **ms_unlisten( )** failures.)
Resolved	IPY00038551	--	162	Modular Station Interface (MSI)	**ms_stopfn( )** causes two TSC_MsgReleaseCall messages to be sent to the Dialogic® DM3 Analog TSP.
Resolved	IPY00038433	--	160	Modular Station Interface (MSI)	The **ms_stopfn( )** function fails to stop the ringing on a Dialogic® DISI32R2 Board.
Resolved	IPY00028642	36548	92	Modular Station Interface (MSI)	The **ms_estconf( )** function is not working correctly in Service Update 74.
Resolved	IPY00019148	29058	--	Modular Station Interface (MSI)	**ms_genring( )** will cause an application to crash (in libdm3msi.dll) if you open the same MSI device twice and if you execute the **ms_genring( )** on the second device handler. Workaround: Don't open the MSI devices twice.
Resolved	IPY00019145	28944	--	Modular Station Interface (MSI)	Using MS_RNG_DEFAULT in the **ms_genringex( )** function generates “Invalid Ring Cadence” error.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00029931	36809	110	NCM API	Throws an exception when attempting to run application from debugger.
Resolved	IPY00012520	29075	--	NCM API	Attempting to add a new device using **NCM_AddThirdPartyDevice( )** fails with a name that is subset of an existing device name.
Resolved	IPY00008465	32590	30	NCM API	If an application calls **NCM_StartSystem( )** and then aborts while polling to see if services have started up via **NCM_GetSystemState( )**, then on reattempts at starting services downloads fail and can result in a crash of dlgc_srv.exe.
Resolved	IPY00022302	27992	--	NCM Documentation	**NCM_DetectBoardsEx( )** does not give list of PCD files for Dialogic® DM3 Boards. This was resolved by updating the NCM API documentation.
Resolved	IPY00040536	--	178	OA&M	While application is running, message is logged in the Windows® event log: Faulting application OAMEventService.exe, version 1.0.0.21,
Resolved	IPY00032271	36699	104	OA&M	There is a limitation to the amount of processes you can use because of a limitation of signals you can create in the operating system.
Resolved	IPY00038280	--	155	OA&M,	A non-OAMIPC based client was attempting connection to an internal software component, an OAMIPC-based server. This caused the internal OAMIPC-based server to crash when invoking the Dialogic® System Service startup or shutdown.
Resolved	IPY00039014	--	167	PBX Call Control	Adept display parser cannot handle large displays correctly. Displays larger than 24 characters (per line) do not parse correctly (regardless of rules created). Nortel PBXs can be configured to use displays larger than 24 characters per line (e.g., 32 characters). When the customer does so, the functions **d42_gtcallid( )** and **d42_gtcallidex( )** return invalid displays.
Resolved	IPY00006562	35636	108	PBX Call Control	When Mitel SX-2000 switch swaps CPU, there is a Loss of Carrier, but does not gain carrier back when finished.
Resolved	IPY00041082	--	178	PBX Expert (previously called PBXpert)	Manual mode is grayed out on PBX Expert. **Note:** Manual mode has been restored to PBX Expert in Dialogic® System Release 6.0 PCI for Windows®; it is applicable to Dialogic® Springware Boards only.
Resolved	IPY00013978	25586	--	PBX Expert (previously called PBXpert)	PBX Expert defined disconnect tone does not trigger LCOFF event in the application.
Resolved	IPY00007737	32060	22	PBX Expert (previously called PBXpert)	The Dialogic® D41JCT-LS Board fails to test the disconnect tone (affects all JCT boards). Problem appears to be in the Dialogic® Voice Library with using **dx_open( )** a second time in PBX Expert.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00016065	28992	--	PBX Integration	The Called side drops a connection when trying to connect from channel 1 to channel 2 through a PBX.
Resolved	IPY00014328	26154	--	PBX Integration	The Hardware Wizard continues to appear each time the machine boots and asks for the correct device driver when using either the Dialogic® D/82JCT-U or D/42NE-PCI Board.
Resolved	IPY00009506	28636	--	PBX Integration	A phantom ring event is generated on the Nortel Meridian.
Resolved	IPY00009268	29354	--	PBX Integration	Call id on the Dialogic® D/82 Board is inaccurate when using networked Meridian switches.
Resolved	IPY00009010	27123	--	PBX Integration	Dialogic® D/42JCT-U Board Mitel integration: When there is a T in the display, it is detected as a trunk call.
Resolved	IPY00008814	29442	--	PBX Integration	Setting or clearing MWIs too quickly causes a loss of ability to dial feature codes.
Resolved	IPY00008813	29388	--	PBX Integration	Forwarded calls that originate on a networked Mitel SX2000 switch are not being tagged with the correct callid strings.
Resolved	IPY00008445	29094	--	PBX Integration	Host application does not receive TD42_ASYNCCHSTATUS in Mitel 2000.
Resolved	IPY00037923	--	160	PDK	Using PDK protocols on a system with Dialogic® Springware and DM3 Boards, T1/E1 GC Alarm Condition: evt=0x832 occurs, causing a GCEV_BLOCKED event. The channel remains in a BLOCKED state.
Resolved	IPY00016176	28334	--	PDK	The num_rings parameter for **gc_AnswerCall( )** does not work on Dialogic® DM3 PDK, and always defaults to 2.
Resolved	IPY00016173	28222	--	PDK	PDK Manager takes a long time to load (several minutes per board).
Resolved	IPY00016167	28077	--	PDK	Single board start/stop fails on Dialogic® DM3 Boards with PDK protocols.
Resolved	IPY00016157	27800	--	PDK	pdk_us_mf_io fails to detect busy when used on Dialogic® DM3 Boards.
Resolved	IPY00015569	28209	--	PDK	The ml1_ds2_cas.pcd file will print the message “timerIdx = 61166" while running test application.
Resolved	IPY00031767	36021	84	Protocols	pdk_r2_io.psi sets a wrong channel state after a timer expires.
Resolved	IPY00030912	33334	18	Protocols	pdk_us_ls_fxs_io reports that the firmware detected a disconnect tone while there was no disconnect tone on the line. Calls are being falsely disconnected because of this.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00030679	33146	18	Protocols	When using the pdk_us_mf_io protocol, the GCEV_UNBLOCKED event is not generated when Layer 1 alarms are cleared after the channel is put OOS.
Resolved	IPY00028595	35538	70	Protocols	Basic call control using Korea GDS LS protocol does not work on Dialogic® Springware Boards.
Resolved	IPY00028584	35809	84	Protocols	A6 should be a terminating tone for pdk_in_r2_io.cdp protocol.
Resolved	IPY00028497	36042	84	Protocols	With South Africa with pdk_sw_e1_ac4400_io.cdp, a disconnect tone does not get detected.
Resolved	IPY00028454	36090	84	Protocols	When using Lucent Lineside E1 PDK protocol, the **gc_RetrieveCall( )** function failed to transfer the call state from GCST_ONHOLD to GCST_CONNECTED.
Resolved	IPY00028411	34284	84	Protocols	When using 5ESS protocol with Dialogic® Springware Boards, outbound calls fail with Cause Value 1100100, invalid information element, in response to the Proceeding and Progressing message received.
Resolved	IPY00028378	34586	56	Protocols	For inbound call, channel is blocked after the remote caller hangs up before sending DNIS, when using pdk_hk_dtmf_io.cdp.
Resolved	IPY00028363	36020	84	Protocols	Dialogic® Springware T1 Boards send incorrect “Interface ID present” to remote side when using T1 ISDN (DMS, 4ESS, 5ESS).
Resolved	IPY00028277	32444	18	Protocols	A GCEV_MEDIADETECTED event is not generated on the Dialogic® DM/V160-LP Board even though the call is answered at around 30 seconds.
Resolved	IPY00010746	35042	56	Protocols	When using the pdk_us_mf_io protocol, if CDP_OUT_Send_Alerting_After_Dialing = 1 and CPA is disabled, the user expects to get the GCEV_ALERTING event right after dialing. However, if the remote side answers the call too quickly, the GCEV_CONNECTED event is returned and the GCEV_ALERTING event never comes in.
Resolved	IPY00010664	34063	27	Protocols	When you configure a Dialogic® DM3 Board with ISDN 4ESS for USER and NETWORK side, the ANI cannot be extracted properly on an incoming call using **gc_GetCallInfo( )**.
Resolved	IPY00010621	34537	56	Protocols	When using the pdk_us_mf_io protocol in the Feature Group D configuration, ANI is missing the last digit when ANI is not terminated with the expected ST digit.
Resolved	IPY00010520	34048	27	Protocols	When you configure a Dialogic® DM3 Board with ISDN 5ESS for the NETWORK side, the ANI cannot be extracted properly on an incoming call using **gc_GetCallInfo( )**.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00010372	35035	56	Protocols	After sending CAS_HOOKFLASH, there should be some delay before sending DTMF in pdk_sw_e1_necls_io protocol, if CDP_WaitDialToneEnabled = 0 (i.e., do not wait for dialtone).
Resolved	IPY00010223	34985	56	Protocols	pdk_sw_e1_ermx_io.cdp can only accept one ringing signal (the internal ringing or the external ringing but not both). Defining CAS_RING_APPLIED (0001 -> 0xxx) solves the detection of the two ringing signals but causes problems with outgoing calls.
Resolved	IPY00010004	34685	56	Protocols	When using the pdk_us_mf_io protocol in the Feature Group D configuration, the protocol does not send a Disconnect signal when it times out waiting for DNIS and ANI. This occurs when the remote side is configured as Feature Group B and makes a call.
Resolved	IPY00009943	34160	62	Protocols	If pdk logs are enabled for TxRx bit information (by adding "ALL INTEGER_t PSL_TXRX_LOG_ENABLE=1" in the respective .cdp file), **gc_MakeCall( )** fails with GCEV_TASKFAIL.
Resolved	IPY00009887	34053	25	Protocols	When configuring a system to use ISDN NI2 protocol in conjunction with NT1, the D-channel does not come up.
Resolved	IPY00009837	35049	56	Protocols	There seems to be a hard-coded 30-second timeout on a Make Call when the call is made in Alerting mode, which will terminate the call if no one picks up the phone. The expected behavior is that the call will not be dropped automatically, so the phone will ring forever if no one picks up. This occurs on T1 CAS lines.
Resolved	IPY00009409	34663	56	Protocols	When using FXS protocol and calling a busy station using supervised transfer, you get a disconnect event for both the consultation CRN and transferred CRN.
Resolved	IPY00009407	32547	18	Protocols	The **gc_SetChanState( )** function fails to return a completion event or error indication when using the pdk_us_mf_io protocol on a Dialogic® DM/V960A-4T1 Board.
Resolved	IPY00009272	33981	27	Protocols	When using Qsig protocol, a DISCONNECT message is received 4 seconds of making a call into a cellular network. The cause code is 102 "Recovery on timer expiry".
Resolved	IPY00008963	32443	18	Protocols	A GCEV_MEDIADETECTED event is not generated when running FXS on a Dialogic® DM/V960A-4T1 Board and the call is not answered until after the CA_ANSWER time-out.
Resolved	IPY00008634	31778	18	Protocols	**cc_CallAck( )** sends incorrect IEs when used for sending a host controlled PROCEEDING message.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00008374	32601	18	Protocols	The **gc_SetChanState( )** function fails to return a completion event or error indication when using the pdk_us_mf_io protocol on a Dialogic® DM/V960A-4T1 Board.
Resolved	IPY00008220	34972	56	Protocols	When using the pdk_us_mf_io protocol, digits from the previous call are returned in ANI.
Resolved	IPY00008138	32978	25	Protocols	DPNSS firmware doesn't answer to an EEM(C) *89B# message (Add-On Validation for conference support).
Resolved	IPY00007955	17567	18	Protocols	With the Dialogic® D/240SC-T1 Board, the 4ESS protocol does not support multiple restart.
Resolved	IPY00007716	27336	18	Protocols	When using the ETSI option to drop a call by directly sending a RELEASE instead of a CALL DISCONNECT message, the response is “REL” but should be “REL COMPL”.
Resolved	IPY00007573	29445	18	Protocols	The **gc_MakeCall( )** function “timeout” parameter with pdk_us_mf_io times out after a maximum of 45 seconds when using wink start protocol.
Resolved	IPY00007327	30233	56	Protocols	With the pdk_mx_r2_io protocol, if the E1 cable is disconnected and reconnected, the application does not receive all the GCEV_UNBLOCKED events.
Resolved	IPY00006823	35851	70	Protocols	When using the pdk_us_mf_io protocol, the firmware crashes when CAS_Seize is similar to a wink signal.
Resolved	IPY00006811	36584	92	Protocols	The pdk_us_ls_fxo protocol fails to notify the PDK library that the disconnected call has been already released, which prevents the application from dropping the call when a new incoming call is pending.
Resolved	IPY00006809	34543	56	Protocols	When CDP_IN_DNIS_ST_Needed = 0, the pdk_e1_cas_io protocol should not issue timed-out error while waiting for DNIS.
Resolved	IPY00006804	34319	37	Protocols	If a board is configured using *pdk_us_ls_fxs_io.cdp* file and a call is abandoned after the first ring, the application is not receiving the GCEV_DISCONNECTED event that is expected.
Resolved	IPY00006771	34329	56	Protocols	Using Belgium R2 protocol, when configured in DTMF/MF mode, in the Offered state the **gc_ResetLineDev( )** function does not behave properly.
Resolved	IPY00006769	34478	39	Protocols	The default CDP_GrpA_TermToneMask3 for pdk_cn_r2_io.cdp should be 10 instead of 8.
Resolved	IPY00006762	34664	56	Protocols	When using E1 line side protocol and calling a busy station using supervised transfer, you get a disconnect event for both the consultation CRN and transferred CRN.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00006748	34587	56	Protocols	The PDK E1 CAS protocol cannot be downloaded on Dialogic® DM3 Boards, and Dialogic® Springware Board channels cannot be opened when using this protocol.
Resolved	IPY00006735	34344	56	Protocols	On Dialogic® DM3 Boards, when dialtone is enabled on Belgium R2 protocol, if the first DTMF/MF digit of DNIS sent is 1 then the DNIS digits received at the inbound side are not the same as sent by the outbound side.
Resolved	IPY00039179	--	162	PSTN Call Control	During a glare scenario, a GCEV_RELEASECALL event is incorrectly returned to the synchronous function **gc_ReleaseCall( )**. Events should only be returned from asynchronous functions.
Resolved	IPY00038979	--	160	PSTN Call Control	The pdk_sw_e1_fxs_io protocol does not forward the correct reason when a call is disconnected due to detection of a SIT. The reason should indicate that SIT was detected.
Resolved	IPY00038612	--	160	PSTN Call Control	When calling **gc_BlindTransfer( )** synchronously, the function sometimes returns -1 and takes approximately 30 seconds to return with this error.
Resolved	IPY00038494	--	160	PSTN Call Control	CP failure on Dialogic® DM/N960-4T1 Board.
Resolved	IPY00038244	--	154	PSTN Call Control	If **gc_MakeCall( )** is called with GC_PARM_BLK set to NULL, ERR1 is shown in the RTF log.
Resolved	IPY00038130	--	154	PSTN Call Control	A GCEV_FATALERROR occurs on Dialogic® D/480JCT-2T1 Board.
Resolved	IPY00037607	--	148	PSTN Call Control	If another call comes in between a **gc_DropCall( )** and **gc_ReleaseCallEx( )**, the call is not detected. The problem occurs when the drop call and release call are issued within 1-2 seconds of each other.
Resolved	IPY00036886	--	142	PSTN Call Control	The call type information is incorrectly being encapsulated in the METAEVENT's extevtdatap pointer in the GCEV_OFFERED event when using ISDN call control on Dialogic® DM3 Boards.
Resolved	IPY00036833	--	142	PSTN Call Control	When using NI2 protocol on Dialogic® JCT Boards, disconnect glare causes next call to be rejected with cause code 44, channel not available.
Resolved	IPY00036830	--	142	PSTN Call Control	The DPNSS cause “Network Termination” (NT=0x02) is not supported.
Resolved	IPY00036448	--	142	PSTN Call Control	With 5ESS ISDN on Dialogic® Springware Boards, call setup fails when the CALLED NUMBER TYPE is set to NETWORK_SPECIFIC (0x03).
Resolved	IPY00036347	--	142	PSTN Call Control	QERROR_WARNING messages appear in *Dm3StdErr log*. Eventually, **gc_SetChanState( )** fails on all channels, and all channels are blocked.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00036337	--	139	PSTN Call Control	5ESS for Dialogic® DM3 Boards did not support CALLED NUMBER TYPE in the NETWORK_SPECIFIC (0x03), IE.
Resolved	IPY00036248	--	139	PSTN Call Control	When using Dialogic® Global Call SS7, the 0xb and 0xc address signals, which were previously reported to the application as "b" and "c", are now getting reported as "#" and "*", thus breaking backward compatibility.
Resolved	IPY00036247	--	139	PSTN Call Control	A Dialogic® JCT Board running with the NT1 protocol receives an Alerting message with incorrect GCEV_PROCEEDING event instead of the expected GCEV_ALERTING on a channel.
Resolved	IPY00036101	--	139	PSTN Call Control	User program cannot obtain a large UUI information along with other IEs using **cc_GetSigInfo( )** on Dialogic® JCT Boards.
Resolved	IPY00036044	--	139	PSTN Call Control	Failures seen when invoking **gc_SetChanState( )** on Dialogic® JCT Boards.
Resolved	IPY00035451	--	131	PSTN Call Control	WinXP **gc_OpenEx( )** fails for device ":N_dkB1T1" for Dialogic® SS7 Board when configured for clear channel.
Resolved	IPY00035148	--	131	PSTN Call Control	The **gc_Unlisten( )** function has no effect when issued on "dk" devices using Dialogic® Global Call SS7.
Resolved	IPY00034816	--	131	PSTN Call Control	SIT tone not detected on Nortel Meridian protocol.
Resolved	IPY00034738	--	131	PSTN Call Control	Call progress analysis does not properly report fax tone when parameter All INTEGER_t CDP_OUT_ConnectType has a value of "1".
Resolved	IPY00034618	--	125	PSTN Call Control	**gc_DropCall( )** fails when responding to a GCEV_DISCONNECT event after a GCEV_BLOCKED event.
Resolved	IPY00034606	--	131	PSTN Call Control	While issuing a make call during a supervised transfer to a destination that is busy, **gc_ResultMsg( )** returns with PROTOCOL ERROR.
Resolved	IPY00034050	36636	116	PSTN Call Control	SIT tone operator intercept is incorrectly reported as Unknown SIT tone to the application on Dialogic® DMV160LP Board.
Resolved	IPY00034018	--	115	PSTN Call Control	SIT tone operator intercept is incorrectly reported as Unknown SIT tone to the application on Dialogic® DMV960A and DMV160LP Boards.
Resolved	IPY00033698	--	124	PSTN Call Control	The primary call cannot be re-transferred via **gc_SetupTransfer( )** when the transferred call is disconnected after SwapHold.
Resolved	IPY00033244	--	113	PSTN Call Control	Dialogic® DM/V1200BTEP Board is sending a RELEASE COMPLETE (with cause 0x22) after receiving a CALL PROCEEDING.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00033163	--	124	PSTN Call Control	Access violation occurs when running ISDN based application.
Resolved	IPY00033009	--	113	PSTN Call Control	NCAS does not work on mixed T1/E1 or mixed ISDN/CAS configurations.
Resolved	IPY00032875	--	116	PSTN Call Control	Cannot send Facility Message on CRN1 when CRN1 is put on hold with Dialogic® DM3 Boards.
Resolved	IPY00031559	36828	113	PSTN Call Control	Station sets of Dialogic® DISI Board always initialize to ONHOOK.
Resolved	IPY00014188	30005	--	R4 Libraries	Running Dialogic® R4 application under the VC++ 6.0 debugger results in 100% CPU utilization and missing events after closing and reopening Global Call devices
Resolved	IPY00039492	--	165	Runtime Trace Facility (RTF)	RTF logging has a memory leak and drops some log messages.
Resolved	IPY00038894	--	160	Runtime Trace Facility (RTF)	RTF logging corrupted device name in **dx_close( )**.
Resolved	IPY00038545	--	165	Runtime Trace Facility (RTF)	In RTFManager, the *RtfMatrix.xml* file was used to map the modules in the RTFConfig file to a family and technology group. But if any changes were made to the RTFConfig file outside of RTFManager, the configuration section would fail. **Note:** The mapping file was removed, and attribute tags were added to the RTFConfig file to define the mappings, making the configuration section of RTFManager more robust.
Resolved	IPY00038524	--	160	Runtime Trace Facility (RTF)	Multiple threads can be created in the RTF server for a single client when the system is heavily loaded. This leads to a build-up of threads in the server, which can lead to thread creation failures.
Resolved	IPY00037789	--	160	Runtime Trace Facility (RTF)	RTF logs are not generated if application is executed as a service and launched as user “Network Services.”
Resolved	IPY00036919	--	148	Runtime Trace Facility (RTF)	Unable to configure RTF trace capabilities using RTFManager because the selection is grayed out.
Resolved	IPY00036469	--	148	Runtime Trace Facility (RTF)	RTF 3.0 introduced increased memory usage of 7 MB in the client. So for each process running on the system that is directly or indirectly linked with RTF, an additional 7 MB of memory is used.
Resolved	IPY00038572	--	165	SIP Call Control	When running a Dialogic® Global Call IP-based application that enables notification of certain SIP messages through GCEV_EXTENSION events, the application is not able to determine the IPPARM_MGSTYPE value for incoming SIP messages. The message type value returns more bytes than expected, making the application unable to decipher which message was received.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00035831	--	134	SIP Call Control	Segmentation fault occurs in libipm_vsc.so when calling **gc_close( )** on Dialogic® Global Call (IP based) line device.
Resolved	IPY00035822	--	134	SIP Call Control	Dialogic® Global Call SIP application does not respond to 407 Proxy Authentication Required messages.
Resolved	IPY00035613	--	134	SIP Call Control	Fails to send a BYE message after dropping call on Avaya IP PBX.
Resolved	IPY00032715	--	110	SIP Call Control	SIP 3xx redirection and forward message: cannot set (or retrieve) more than one redirection address.
Resolved	IPY00033102	--	125	SIP Call Transfer	Supervised transfer fails on Party B getting GCEV_XFER_FAIL.
Resolved	IPY00036855	--	154	SNMP	When using MIB2 from RFC1213, Dialogic® SNMP agent fails to return valid information when a "get" command is issued.
Resolved	IPY00010060	34495	56	SNMP	SNMP service crashes periodically. The event log entry reads: "Faulting application snmp.exe, version 5.2.3790.0, faulting module CosNaming405.dll, version 0.0.0.0, fault address 0x00004e2e".
Resolved	IPY00009266	34050	56	SNMP	*dlgagent.log* under "c:" is created by SNMP and increases automatically when DCM starts.
Resolved	IPY00033492	--	131	Springware Boards	After repeating of network connection down and up while service/app is running, some channels cannot re-establish layer 2 connection, send or receive calls.
Resolved	IPY00021428	30443	--	Springware Boards	The listboards utility will not display the model number, configuration ID, or the driver state for Dialogic® Springware Boards.
Resolved	IPY00009171	29859	18	Springware Boards	A board configured for Layer 2 access should be able to bring down/up layer 2 on demand. When cycling a few times through the UP and DOWN states with the NE1 firmware on trunk 1 and CTR4 firmware on trunk 2, it eventually becomes impossible to activate layer 2.
Resolved	IPY00039427	--	166	Springware Call Control	If an outbound call is made (**gc_MakeCall( )**) and then a **gc_DropCall( )** is issued, a drop call event should be received. But instead, a disconnect event is returned.
Resolved	IPY00039331	--	165	Springware Call Control	When using DPNSS, the response to the setup message from the switch is incorrect; an incomplete Number Acknowledge Msg is returned.
Resolved	IPY00039249	--	162	Springware Call Control	When **gc_WaitCall( )** is issued after an incoming call is pending, the **gc_AcceptCall( )** fails even though the application receives the GCEV_OFFERED event.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00038539	--	160	Springware Call Control	Interface ID Present enabled in network setup message causes channel on Dialogic® D/480JCT Board using 4ESS protocol to reject call with invalid information element contents.
Resolved	IPY00037633	--	148	Springware Call Control	**gc_BlindTransfer( )** does not work when using the pdk_sw_e1_ssls_io protocol with Dialogic® Springware Boards.
Resolved	IPY00037318	--	148	Springware Call Control	Dialogic® Springware ISDN 4ESS protocol does not support LOCATION type 1010 in Progress Indicator.
Resolved	IPY00041426	--	181	Springware Fax	The **ATDV_ERRMSGP( )** function returns a "Null" string if the **fx_sendfax( )** function fails with **ATDV_LASTERR( )** error code 0x114 when attempting to send an invalid TIFF file. The error string returned by the former should reflect a valid string that relates to the error value from the latter.
Resolved	IPY00040798	--	174	Springware Fax	When enabling RTF logging (after modifying the *RTFConfigWin.xml* file), the Fax demo fails to start and exits with an exception. When the RTF trace is enabled with default settings, the Fax demo doesn't crash. However, when the *RTFConfigWin.xml* file is modified to trace application activities, the Fax demo crashes.
Resolved	IPY00039341	--	165	Springware Fax	The Dialogic® VFX/41JCT-LS Board sometimes fails to receive fax with ATFX_ESTAT() = 195 when using 14.4kbps/no ECM mode and multi-page signal (MPS).
Resolved	IPY00038836	--	160	Springware Fax	Fax error codes are not reported properly with Dialogic® VFX41JCT-LS Board.
Resolved	IPY00038419	--	160	Springware Fax	The **fx_sendfax( )** function never returns, and CPU utilization reaches 100%.
Resolved	IPY00038298	--	160	Springware Fax	When using Dialogic® VFX/41JCT-LS Board, multiple consecutive ECM fax receive calls failed and **ATFX_ESTAT( )** reported 198. Non-ECM fax receives by the same channel were successful.
Resolved	IPY00033122	--	108	Springware Fax	Firmware crash occurs when receiving particular FSK data during send or receive fax.
Resolved	IPY00031536	36637	108	Springware Fax	The entire Dialogic® VFX/41JCT-LS Board gets hung after some particular image is received.
Resolved	IPY00030908	34886	70	Springware Fax	When an ASCII text file is faxed from a Dialogic® VFX/41JCT-LS Board and the resolution is set to fine, the font size of the received document is reduced by about half of the sent document's font size.
Resolved	IPY00030906	36237	84	Springware Fax	**fx_open( )** causes memory leak on some analog boards.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00030890	32415	70	Springware Fax	When an ASCII text file is faxed from a Dialogic® VFX/41JCT-LS Board and the resolution is set to fine, the font size of the received document is reduced by about half of the sent document's font size.
Resolved	IPY00030883	32772	70	Springware Fax	Tiff header font size when sending fax is reduced on the received document when faxed from a Dialogic® VFX/41JCT-LS Board.
Resolved	IPY00030880	35634	84	Springware Fax	Fax reception fails when DF_ACCEPT_VRQ is set in the receive flag and the sending fax machine is PRI-MPS capable.
Resolved	IPY00028611	36204	84	Springware Fax	When a Dialogic® VFX Board is receiving fax from a particular fax machine, after it sends DIS, it sometimes cannot recognize DCS from the remote end and the call is disconnected with phase E status: “Excessive HDLC carrier” without retry.
Resolved	IPY00028599	35799	65	Springware Fax	**fx_rcvfax( )** does not terminate when fax call is disconnected during RNR/RR and there's busy tone.
Resolved	IPY00028578	36159	84	Springware Fax	During ECM receiving, the board sends an invalid PPR that is not requesting any frames for resend.
Resolved	IPY00028480	36640	95	Springware Fax	Dialogic® VFX/41JCT-LS Board randomly fails to receive multi-page inbound fax. The receiving side responds to the Multi-Page Signal (PPSMPS) message with a request to repeat last message (CRP). The sending fax machine repeats the PPSMPS message two more times, followed by a disconnect (DCN) message. The Dialogic® Fax Library returns error of EFX_DISCONNECT, and **ATFX_ESTAT( )** returns 127 (which is EFX_WHYDCNRX see faxlib.h) /* Unexpected DCN while waiting for DCS/DIS */.
Resolved	IPY00028479	35937	70	Springware Fax	Dialogic® VFX/41JCT-LS Board channel does not recover from **fx_rcvfax( )** operation when the remote fax is disconnected and there's busy tone.
Resolved	IPY00028360	33514	62	Springware Fax	An access violation occurs when sending a tiff file starting from a page number it does not contain. For example, if you try to send a tiff file beginning at page 3 that only contains 2 pages, the **fx_sendfax( )** function called will crash.
Resolved	IPY00028351	35775	65	Springware Fax	Part of sent image is sometimes missing from received TIFF file using ECM mode fax receive.
Resolved	IPY00028341	35790	108	Springware Fax	The fax tx modem signal level from a Dialogic® VFX Board changes (for both send and receive fax) after **dx_playiottdata( )** is used.
Resolved	IPY00011005	36213	84	Springware Fax	When using SoftFax, legal size documents sent from a Toshiba fax machine are intermittently split into two pages.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00010734	34753	56	Springware Fax	In ECM fax send operation, the Dialogic® VFX/41JCT-LS Board sets wrong frame count in PPS message when responding to PPR from remote fax machine.
Resolved	IPY00010365	35132	56	Springware Fax	The operating system hangs when trying to download Dialogic® D/4PCIUF Boards.
Resolved	IPY00009411	32855	70	Springware Fax	If a call is placed to a board, and a fax machine is not presented on the line, the fax log reports a CSID that appears to be from the last valid fax call on the line. This results in a Phase E status error 120 = "No fax machine present on the line".
Resolved	IPY00009099	33444	18	Springware Fax	When sending fax to a Xerox Able 1221 fax machine, Dialogic® VFX/JCT fax firmware does not respond to the NSF/CSI/DIS. This is because the NSF was 67 bytes long but the hdlc buffer was only 64 bytes long.
Resolved	IPY00007528	32416	18	Springware Fax	Multiple page fax receiving terminates with EFX_DCNFAXRX = 129 (Unexpected DCN while waiting for EOM/EOP/MPS) when receiving from a Canon G3 High Speed Laser 3170MS machine.
Resolved	IPY00037483	--	148	Springware Firmware	Firmware assert during load test causes boards to stop responding to driver.
Resolved	IPY00033185	--	108	Springware Firmware	On Dialogic® Springware ISDN 5ESS and 4ESS protocols, loopback calls from user to network fails.
Resolved	IPY00032803	--	108	Springware Firmware	Ported PTR 35154 fix from NI2 protocol to DMS protocol.
Resolved	IPY00032794	--	108	Springware Firmware	Board rejects incoming calls when a call is disconnected, but is not released yet in the first unblocked channel.
Resolved	IPY00032266	36735	95	Springware Firmware	The Dialogic® D/41JCT Board fails to detect dial tone on outbound calls.
Resolved	IPY00030911	33413	70	Springware Firmware	Call progress analysis incorrectly reports faxtone as PAMD on some occasions.
Resolved	IPY00028575	35232	62	Springware Firmware	With Dialogic® D/240JCT-LS Board and NTT ISDN protocol, channel can be stuck with valid Q.931 call flow.
Resolved	IPY00028544	35104	62	Springware Firmware	After load testing, **cc_Restart(ASYNC)** does not return CCEV_RESTART, and subsequent calls to **cc_Restart( )** return CCEV_RESTARTFAIL.
Resolved	IPY00028536	36587	94	Springware Firmware	ISDN outbound calls fail when using the NI2 protocol and making back to back calls on a Dialogic® Springware Board.
Resolved	IPY00028524	35566	65	Springware Firmware	When running ISDN, if glare scenarios occur where the application initiates a **cc_AnswerCall( )** slightly after or around the same time that a CCEV_DISCONNECTED event comes in, an assert can result on the Dialogic® Springware Board.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00028518	35832	84	Springware Firmware	GCEV_DROPCALL event is not returned after calling **gc_DropCall( )** if GCEV_DISCONNECT is sent by the firmware due to time-out.
Resolved	IPY00028458	35154	58	Springware Firmware	Dialogic® Springware NI2 firmware sets and sends the interface ID bit and Octet 3.1 in its setup message when configured for NFAS circuits.
Resolved	IPY00028448	36319	94	Springware Firmware	Disconnect glare causes the next call to be rejected with cause code 44, requested channel not available.
Resolved	IPY00028415	35011	65	Springware Firmware	When using DPNSS firmware, disconnection is not completed properly. A dropcall complete event is not received after a remote disconnect event, and the application is left hung in a “disconnecting” state.
Resolved	IPY00028313	34814	65	Springware Firmware	When using an R2 protocol and the user attempts to make a call with greater than 10 DNIS digits, the R2MF response buffer contains garbled data.
Resolved	IPY00010668	34476	56	Springware Firmware	DE_RINGS event is not received properly for double interrupted ring in ROLM 9005 with Dialogic® D42JCT-U Board.
Resolved	IPY00010663	34719	56	Springware Firmware	When DCM is used to set the country code to South Africa (ZA), no audio (or sometimes half-duplex audio) is present.
Resolved	IPY00010611	34999	56	Springware Firmware	Dialogic® Springware NI2 firmware sends out the “Interface Identifier” octet (3.1) all the time for the PROCEEDING message, causing the switch to reject the call.
Resolved	IPY00010475	34241	39	Springware Firmware	Using DPNSS firmware, when a call is made to a PBX extension that is on Divert, the application does not receive a Diversion IE and therefore cannot make a call to the “diverted to” extension.
Resolved	IPY00009981	34345	39	Springware Firmware	Events seem to block on certain channels when using CSP firmware.
Resolved	IPY00009068	34788	56	Springware Firmware	Dialogic® D82 Board ports cannot be made VMS port from PBXDRVR.
Resolved	IPY00009008	32192	18	Springware Firmware	Disconnect glare causes the next call to be refused with cause code 44 - requested channel not available. Disconnect glare is caused when the user side hangs up; but before the switch recognizes this disconnect, it sends its own disconnect. The next incoming call should be accepted, but in some cases it is being rejected, with the ISDN firmware responding that the channel is not available.
Resolved	IPY00008640	32704	18	Springware Firmware	There are caller ID issues (with single and multiple hop forwarded calls) when using the Nortel Norstar PBX.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00008592	32678	18	Springware Firmware	Two bytes of corrupt data are observed in the D-channel trace whenever **cc_SetInfoElem( )** is called to set the value for the CALLING_LINE_ID number in a SETUP message.
Resolved	IPY00008537	32161	18	Springware Firmware	CTR4 firmware responds with STATUS message when Calling Party Subaddress contains 0xF9 (or similar value). On some networks, the status message may cause the network to tear down the call.
Resolved	IPY00007984	32104	18	Springware Firmware	When running into PDK R2 glare conditions, false R2MF time-outs occur in the spanplus firmware, resulting in calls not connecting properly.
Resolved	IPY00007421	25633	18	Springware Firmware	Incorrect response to Release message when using Dialogic® D/600JCT-2E1 Board.
Resolved	IPY00007308	31896	22	Springware Firmware	On a Dialogic® D/82JCTU Board using CSP firmware, multiple channels dialing at the same time cause the board to become non-responsive and report device busy errors. The system stops responding with an assert.
Resolved	IPY00007241	31840	18	Springware Firmware	When there is a call collision between an outbound and inbound call, **gc_GetSigInfo(...U_IES...)** sometimes fails with error 0x303=Information not available. The functions **gc_GetANI( )** and **gc_GetDNIS( )** return the correct information.
Resolved	IPY00007235	29328	18	Springware Firmware	When using **gc_SetInfoElem( )** to set the calling party number in a SETUP message, the first two digits are missing.
Resolved	IPY00041345	--	178	Springware ISDN Firmware	Firmware assert occurs due to zero length User-User IE message, and Dialogic® board stops responding to the switch.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description

Resolved	IPY00036949	--	139	Springware ISDN Firmware	With Service Update 108 and later, outbound ISDN calls are rejected by the switch due to missing IE (octet 3.1, Extension Bit/Interface Identifier). In Service Update 108, Dialogic® Springware firmware was changed to stop sending octet 3.1 if a board was not configured in an NFAS group. However, if a switch requires octet 3.1 to be present in the setup message for non-NFAS configurations, the call is now rejected.
					If the switch requires octet 3.1 to be present for non-NFAS configurations, perform the following:
					Step 1. Modify the *nfas.cfg* file so that each span is in its own NFAS group.
					`#NFAS group 1`
					`# Board ID  Interface ID  D-Channel board ID   NFAS group ID`
					`       1             0         1                     1`
					`       2             1         2                     2`
					`       3             2         3                     3`
					Step 2. Do **not** modify parameter 0016 in the PRM file for each span as you normally would within an NFAS environment. Instead, leave the D-channel as enabled:
					`;---`
					`;--- ENABLE/DISABLE the D channel (Parameter type 16H)`
					`;--- Used only when the protocol type (Parameter number 13H) is PRI ISDN`
					`;--- for NFAS configuration.`
					`;--- Possible values for the data are as follows:`
					`;--- 00H = Undefined.`
					`;--- 01H = Enable the D channel.`
					`;--- 02H = Disable the D channel.`
					`0016 01`

Resolved	IPY00007547	29780	--	Springware JCT	Dialogic® D/41JCT and VFX/41JCT Boards do not set correct RING status bit after first boot up.
Resolved	IPY00036665	--	160	Springware Network	When using DPNSS firmware, disconnection is not completed properly. A dropcall complete event is not received after a remote disconnect event, and spurious interrupt firmware crashes occur.
Resolved	IPY00039490	--	174	Springware PBX	The **d42_setparm( )** for the parameter 0x1A does not work on the Dialogic® D/42JCT-U Board.
Resolved	IPY00038206	--	155	Springware PBX	Using **d42_chnstatus( )** causes a memory leak.
Resolved	IPY00030570	35921	84	Springware PBX	Outbound calls made from the PBX are dialing extra digits at random times.
Resolved	IPY00028459	36329	90	Springware PBX	Display is parsed incorrectly while attempting to view ACD statistics when calling **d42_display( )** or **d42_displayex( )** when using Nortel_Meridian_1.fwl.
Resolved	IPY00010787	36134	90	Springware PBX	When a Dialogic® D/82JCT-U Board is connected to a Siemens Hicom, it consistently loses and re-gains carrier on multiple ports.
Resolved	IPY00009297	34095	84	Springware PBX	**d42_displayex( )** doesn't return the correct softkey displays for Mitel SX-200 PBX.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00007425	32712	62	Springware PBX	If a second call comes in after a first caller disconnects and before the ring off event is generated, the call will not be answered.
Resolved	IPY00028211	33249	56	Springware Protocols	BRI/PCI firmware stops normal operation and error -1, **ATDV_LASTERR( )** 3 is returned from **dt_listen( )**.
Resolved	IPY00008416	29635	--	Springware Protocols	Application calling **dx_dial( )** to perform call progress analysis (CPA). Incorrect CPA result of CR_ERROR = 2 (CR_TMOUTON) can be seen under these conditions: • Valid SIT tone can be heard on the other end of the call, but CPA results return CR_ERROR = 2 (CR_TMOUTON). • Similarly, if a single tone beep is heard on the line which is within range of any of the 3 SIT frequencies set in DX_CAP structure, **dx_dial( )** again returns CR_ERROR = 2 (CR_TMOUTON).
Resolved	IPY00028547	35670	100	Springware PSTN	PDK protocol delivers DETECTED/OFFERED event to the channel even if **gc_ReleaseCall( )** was never called to clean up the previous call on this channel. Once a new call attempts to be transmitted/received on this channel, an error occurs.
Resolved	IPY00006846	36711	100	Springware PSTN	A crash occurred due to corruption in PDKRT library internal database caused by application.
Resolved	IPY00006712	36790	100	Springware PSTN	For Dialogic® Springware Boards, no GCEV_MEDIADETECTED event is received when the first sound heard after a connect is a SIT tone (frequency 914 Hz).
Resolved	IPY00040096	--	174	Springware Voice	Failure to increase media play speed by more than 25% when using **dx_adjsv( )** to set the play speed; the documentation specifies a maximum change of 50%.
Resolved	IPY00040052	--	171	Springware Voice	Perfect Call call progress analysis on Dialogic® Springware Boards sometimes falsely detects dial tone and proceeds with dialing while there is no signal matching for the dial tone criteria.
Resolved	IPY00037746	--	148	Springware Voice	An exception occurs when calling **ATDX_CPERROR( )** with RTF logging enabled. When RTF logging is disabled, the exceptions stopped.
Resolved	IPY00030588	27302	--	Springware Voice	The firmware asserts when the application calls **dx_stopch( )** to stop **dx_rec( )**. The firmware assert is an unexpected interrupt.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00028588	35746	94	Springware Voice	An analog Dialogic® Springware Board kept returning NODIALTONE as result of call progress analysis (CPA) when performing outbound calls. This problem occurred after several outbound calls, which were made previously, experienced line busy. When NODIALTONE was returned, the voice recording transmitted on that line was captured, and it showed the dial tone received correctly.
Resolved	IPY00028455	36248	84	Springware Voice	If six tones of the same frequency are defined, only the first through fifth are detected. This problem occurred only with a specific tone.
Resolved	IPY00028383	35321	62	Springware Voice	Busy tones are detected as “no ringback” in call progress analysis when using **dx_dial( )** method in a Dialogic® Global Call application.
Resolved	IPY00028318	35012	56	Springware Voice	The first **ec_reciottdata( )** done on each channel after the service starts generates a TEC_STREAM event with a termination type of LCOFF.
Resolved	IPY00028288	36063	84	Springware Voice	When using global tone detection (GTD), only four tones are detected. If you define more than this, only four will work.
Resolved	IPY00028271	35671	65	Springware Voice	Analog device will not respond to **dx_sethook( )** after dialing an earth recall “&”. This error occurred under normal working conditions where the earth lines are grounded and an incoming call is received.
Resolved	IPY00028229	35270	65	Springware Voice	Call progress analysis comes back with false cadence connects.
Resolved	IPY00010248	33750	84	Springware Voice	When performing call progress analysis, the results come out differently per channel for the same set of audio data.
Resolved	IPY00009562	32733	18	Springware Voice	Using ISDN NT1 with a Dialogic® D/480SC-2T1 Board, when the inbound application is to receive a call, the calling (Telco) side receives an error message 'protocol error' and the call is dropped.
Resolved	IPY00009442	33994	27	Springware Voice	Dialogic® D/82JCT firmware assert in pbxdrvr.c.
Resolved	IPY00009374	33099	39	Springware Voice	If an outbound call (or transfer) is initiated with Perfect Call Progress, a return value of no-ringback is received if the remote answers the call between the first and second ring and does not say anything/or a silence.
Resolved	IPY00009242	31777	18	Springware Voice	Caller ID is not available on Dialogic® D/120JCT Boards if noise is present on the line.
Resolved	IPY00008546	31747	25	Springware Voice	If **dx_play( )** async is called simultaneously with **ec_getparm(DXCH_EC_TAP_LENGTH)** from another thread on the same voice resource, **dx_play( )** returns TDX_ERROR: Command not supported.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00008215	28550	18	Springware Voice	With the Dialogic® DTI/240SC Board, when the inbound application is to receive a call, the calling (Telco) side receives an error message 'protocol error' and the call is dropped.
Resolved	IPY00008056	33690	22	Springware Voice	Turning Silence Compressed Record on via the *voice.prm* file causes distortion on Dialogic® D600JCT2E1 Boards, rendering recorded files unusable. This problem occurs when recording in 8kHz linear format.
Resolved	IPY00007937	32318	22	Springware Voice	An intermittent problem has been seen when **dx_rec( )** is called synchronously and fails to return after calling **dx_stopch( )** on that channel from another thread. Even after exceeding the MAXTIME timeout value for the **dx_rec( )**, it still leaves the channel in a hung state.
Resolved	IPY00037918	--	165	SS7	The RSI link goes down intermittently.
Resolved	IPY00037767	--	148	SS7	The GCSS7 library does not generate the GCEV_MOREINFO event if it receives a SAM message with only STOP digit (0xf) after the application has already issued **gc_CallAck( )**.
Resolved	IPY00037632	--	148	SS7	If there is a delay in the SS7 server picking up messages from the IPC queue, an ERROR_IO_PENDING occurs and the SS7 library terminates the IPC. This causes all the circuits to get blocked, as there is no more connection with the SS7 service. This is causing the IVRs to get a sudden circuit block from the switch in all of its SS7 circuits.
Resolved	IPY00034404	--	131	SS7	In GCSS7, initial alarm conditions are not propagated up to application.
Resolved	IPY00033499	--	115	SS7	Opening of dti devices via GCSS7 library fails.
Resolved	IPY00039334	--	178	Standard Runtime Library (SRL)	An application crash occurred; the stack trace shows SRL library at the top of the stack.
Resolved	IPY00039155	--	165	Standard Runtime Library (SRL)	An application crash occurs with SRL at the top of the stack; the SRL was not initializing all variables of a structure for a given thread, which can cause an access violation.
Resolved	IPY00038708	--	160	Standard Runtime Library (SRL)	An access violation occurs when application calls **sr_waitevtEx( )** for the same device on multiple threads.
Resolved	IPY00038119	--	154	Standard Runtime Library (SRL)	When using a Dialogic® D/120JCT-LS Board, calling the **ATDV_ERRMSGP( )** function caused a LIB crash. The crash occurred when the application called **ATDV_ERRMSGP( )** at the end of fax reception when **fx_rcvfax( )** returns with -1.
Resolved	IPY00014187	29372	--	Standard Runtime Library (SRL)	Synchronous function can become blocked when **sr_waitevt( )** is called from another thread.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00012758	27572	--	UDD	Dialogic® Diagnostics Software (UDD) does not work correctly on Dialogic® D/600JCT-2E1 Boards.
Resolved	IPY00007931	23718	115	UDD	Dialogic® D/240JCT and D/480JCT Boards fail Dialogic® Diagnostics Software (UDD) firmware download.
Resolved	IPY00007492	29714	--	VFX/41JCT	Dialogic® VFX/41JCT Board failed with **ATFX_ESTAT( )** error code 102 - Got DCN while waiting for DIS.
Resolved	IPY00039032	--	162	Voice	Dialogic® DM3 Voice resources don't go to idle state after **dx_stopch( )** function.
Resolved	IPY00038991	--	160	Voice	Previously existing user-defined tones are still being detected after deletion (i.e., call **dx_deltones( )**) on the same channel in which a new set of different user-defined tones have been created.
Resolved	IPY00037777	--	154	Voice	With **sr_enbhdlr( )** being used to enable handler for all events on dxxxdev, after running **dx_stopch( )** to stop **dx_playiottdata( )**, the callback function didn't run. Also, there is no TDX_PLAY event in the log.
Resolved	IPY00037432	--	148	Voice	The **dx_clrdigbuf( )** function overwrites area of thread's stack space, causing the application to crash.
Resolved	IPY00036865	--	142	Voice	If a user attempts to do a play forever (specifying io_length = -1) with UIO plays on Dialogic® DM3 Boards, there is still a hard upper limit on the number of bytes that can be played, which is approximately equal to 2.147 GB (~2 to the 31 bytes).
Resolved	IPY00036345	--	142	Voice	If a user calls **dx_play( )** asynchronously and then calls **dx_stopch( )** synchronously (possibly from another thread) on the same voice device to stop the play, the application sees different behaviors based on whether the voice device is a Dialogic® DM3 or Dialogic® Springware device. DM3 and Springware devices should behave the same way with regards to the eventing mechanism. **Note:** The fix for defect IPY00036345 changed the eventing mechanism behavior for Springware to match that of DM3. Behavior is now such that calling **dx_stopch( )** synchronously no longer consumes TDX_PLAY events. Springware applications will now receive TDX_PLAY events when calling **dx_stopch( )** synchronously.
Resolved	IPY00034378	--	125	Voice	**dx_playiottdata( )** function does not return TDX_PLAY event when directly followed by a **dx_pause( )** and then **dx_stopch( )**.
Resolved	IPY00034365	--	139	Voice	While the Dialogic® Springware voice module in the RTF config file is enabled, running the gc_basic_call_model application to make an outbound call causes a GCEV_FATALERROR.

Table 9. Issues Sorted By Type, Dialogic® System Release 6.0 PCI for Windows® (Continued)

Issue Type	Defect No.	PTR No.	SU No.	Dialogic® Product(s) or Component(s)	Description
Resolved	IPY00033029	--	108	Voice	When playing (dx_playiottdata) from multiple offsets of the same file, only the first portion is played.
Resolved	IPY00032363	--	108	Voice	Random segmentation faults happen due to reading uninitialized memory.
Resolved	IPY00037818	--	154	Voice API	The **dx_setevtmsk( )** function fails to disable the TDX_CST events for DE_DIGITS when setting the DM_DIGOFF flag. **Note:** A documentation update has been added in the Documentation Updates section for the Dialogic® Voice API Library Reference. Please refer to it for important information relevant to this defect resolution.
Resolved	IPY00022258	27159	--	Voice Documentation	Voice documentation does not specify time units for the Voice Board Parameters (Table 4). This was fixed in the *Dialogic® Voice API Library Reference*.

Documentation Updates

This chapter contains information on updates and corrections to the documents included in Dialogic® System Release 6.0 PCI for Windows®. Documentation updates are divided into the following categories:

3.1 System Release Documentation

This section contains updates to the following documents:

- Dialogic® System Release 6.0 PCI for Windows® Release Guide

3.1.1 Dialogic® System Release 6.0 PCI for Windows® Release Guide

Updates to the **System Requirements** chapter

Since the original release of Dialogic® System Release 6.0 PCI for Windows®, additional operating systems are now supported with the Service Update. See Section 1.65, "New Operating System Support", on page 239 of this Release Update.

The **System Requirements** chapter should include the following notes (PTR# 32933):

Note: Dialogic® drivers do not support Physical Address Extensions (PAE). Users using Windows® 2003 with Service Pack 1 will have to disable PAE (which is enabled by default in Service Pack 1).

Note: Dialogic® drivers do not support more than 4 GB of RAM.

The following paragraph should be added to the **Basic Hardware Requirements** section:

This system release supports Intel Hyper-Threading Technology (HT Technology). Multi-Threaded Applications running on HT Technology enabled platforms will interoperate safely with this system release.

The following note should be added to the **Basic Software Requirements** section (PTR# 36031):

Note: Terminal Services Application Server Mode and Active Directory Application Server Mode are not supported on any operating systems.

In the **Basic Software Requirements** section, in the note about using U.S. English versions of the operating system, the information about manual file copy operations if you are using a language other than U.S. English should be deleted (PTR# 36671). The note should simply say:

Note: This system release is designed only for U.S. English versions of the Windows® Operating System.

Updates to the **Features by Product** chapter, **Dialogic DI/0408-LS-A-R2 Features** section

The following item belongs under the "Other Supported Features" heading:

- fixed routing support

The **Dialogic DI/0408-LS-A-R2 Features** section erroneously mentions support for the DI/0408-LS-A Board. This board is not supported in Dialogic® System Release 6.0 PCI for Windows®. Refer to the **Supported Hardware** chapter in the Release Guide for a complete list of supported hardware in this release. (PTR# 33615)

Updates to the **Features by Product** chapter, **Dialogic DI/SIxx-R2 Series Features** section

The following item belongs under the "Other Supported Features" heading:

- fixed routing support

Updates to the **Features by Product** chapter, **Dialogic DMV160LP Analog Loop Start Board Features** section

The following item belongs under the "Other Supported Features" heading:

- flexible routing (exportable voice resources) support

The following feature under the "Other Supported Features" heading is not supported and should be deleted (PTR# 36105):

- Hook-flash through the Global Call API

Updates to the **Features by Product** chapter, **DM/IP Series Features** section

Continuous Speech Processing (CSP) support is listed as a feature. However, CSP is not supported on all Dialogic® DM/IP boards. CSP is supported only on the Dialogic® DM/IP241-1T1-PCI-100BT and DM/IP301-1E1-PCI-100BT boards with the following firmware:

- Dialogic® DM/IP241-1T1-PCI-100BT Board:
  ipvs_cas_311.pcd
  ipvs_evr_cas_311.pcd
  ipvs_evr_isdn_4ess_311.pcd
  ipvs_evr_isdn_5ess_311.pcd
  ipvs_evr_isdn_dms_311.pcd
  ipvs_evr_isdn_ni2_311.pcd
  ipvs_evr_isdn_ntt_311.pcd
  ipvs_evr_isdn_qsigt1_311.pcd
- Dialogic® DM/IP301-1E1-PCI-100BT Board:
  ipvs_evr_isdn_net5_311.pcd
  ipvs_evr_isdn_qsige1_311.pcd
  ipvs_evr_r2mf_311.pcd

Updates to the **OA&M Software** chapter

The SCbus is supported by Dialogic® Springware and Dialogic® DM3 Boards in Dialogic® System Release 6.0 PCI for Windows®. If a DM3 board, however, is configured as the Clock Master and the SCbus is selected as the TDM Bus Type, the

DM3 board cannot use any of the FrontEnd values in the **Derive Primary Clock From** parameter.

In the **Administrative Software** section, the following restriction and limitation on the Board Management API is documented:

Note: A restriction and limitation of Board Management is that it is supported only on JCT single span and dual span boards under the following protocols: T1 4ESS, T1 5ESS, T1 DMS100, and T1 NI2. (E1 protocols are not supported.)

This information is correct except that it also applies to Dialogic® DM3 boards. And with the Service Update, additional protocols are supported for DM3 boards. Replace the restriction and limitation with the following correct one:

Note: The Board Management API is supported on JCT single span and dual span boards using the following protocols: T1 North American ISDN (4ESS, 5ESS, DMS100, DMS250, and NI2). (E1 protocols are not supported.)

The Board Management API is supported on digital DM3 boards (the board must have a network interface for the API to be supported) using the following protocols: T1 North American ISDN (4ESS, 5ESS, DMS100, DMS250, and NI2). With the Service Update, additional protocols are supported: E1/T1 CAS (PDK protocols), additional T1 ISDN (NTT and QSIG-T1), E1 ISDN (NET5 and QSIG-E1), DPNSS, and DASS2.

Updates to the **Development Software** chapter, **New Conferencing Library API** section and **Audio Conferencing API Library** section

The following note, which appears in both the **New Conferencing Library API** section and **Audio Conferencing API Library** section, should be **deleted** because it is no longer correct:

Note: Although the Audio Conferencing (DCB) API continues to be supported, it is recommended that all new conferencing applications be developed using the new Conferencing (CNF) API.

Updates to the **Development Software** chapter, **Standard Runtime Library API** section

The New Features list is incomplete. The correct list of new features is as follows:

- Support for an alternative variant of the extended asynchronous programming model
  A set of functions called the device grouping API has been added to support a more efficient alternative to the **sr_waitevtEx( )** variant of the extended asynchronous model.
- Device mapper functions
  A set of new device mapper functions are available to return information about the structure of the system.
- Support for user context in asynchronous programming model
  The **sr_GetUserContext( )** function has been added to the SRL. This function provides support for user context in asynchronous mode. User context is a mechanism that allows you to match a termination event with a function call by returning the user-supplied pointer originally passed to the function. In this release, user context is only supported in the new Conferencing (CNF) API library.

Updates to the **Development Software** chapter, **Voice API Library** section

The New Features list is incomplete. The correct list of new features is as follows:

- Cached prompt management
  Prompts can be stored in on-board memory rather than in the host computer to reduce latency. The **dx_getcachesize( )** and **dx_cacheprompt( )** functions and the TDX_CACHEPROMPT event have been added to the Voice library to support cached prompt management.

- IMA ADPCM (32.443 Kbps) coder
  The IMA ADPCM coder (VOX and WAVE file formats) is supported. IMA is an acronym for the Interactive Multimedia Association, which defined and published the ADPCM algorithm.
- Enhancement to Multi Frequency (MF) signaling
  MF tone detection is now supported. Previously, only MF tone generation was supported.
- Increased granularity for DX_MAXSIL and DX_MAXNOSIL termination conditions (DV_TPT structure)
  The range of valid values for DX_MAXSIL and DX_MAXNOSIL is now 10 ms to 250 seconds (1 to 25000 in 10 ms units). There are no further restrictions within this range. Previous range of time was 1 second to 30 seconds, with step values.
- Streaming to board
  This feature enables streaming to a network interface in real time which is essential in applications such as text-to-speech and IP gateways. Several new functions have been added to the Voice library.
- Enhancements to call progress analysis
  Enhancements include the ability to modify call progress analysis tone definitions on Dialogic® DM3 boards. The **dx_createtone( )**, **dx_deletetone( )**, and **dx_querytone( )** functions have been added to the Voice library.
- Automatic gain control (AGC) configurable on a per-channel basis through new **dx_setparm( )** parameters
  The new parameters for AGC have the prefix DXCH_AGC_. Previously AGC was configurable on a board basis.
- Record notification beep tone generation (used in call logging applications)
  The **dx_SetRecordNotifyBeepTone( )** function has been added to the Voice library.
- Playback pause and resume
  This feature allows a playback to be paused and then resumed at the exact point it was stopped without loss of data. The **dx_pause( )** and **dx_resume( )** functions have been added to the Voice library.

Updates to the **Supported Hardware** chapter

The following items belong under the “Fax Boards” heading in the **Media Processing - Single Media Boards** section (PTR# 33046):

- CPi/200B
- CPi/400B

In addition, for information about new boards supported with the Service Update, see Section 1.75, “New Boards Supported”, on page 259 of this Release Update.

Update to the **Separately Orderable Products** chapter

This chapter, which refers to the Global Call Protocols Package, is no longer applicable. With the Service Update, the Global Call Protocols Package can now be installed as part of Dialogic® System Release 6.0 PCI for Windows®.

3.2 Installation and Configuration Documentation

This section contains updates to the following documents (click the title to jump to the corresponding section):

- Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide
- Dialogic® Springware Architecture Products on Windows® Configuration Guide
- Dialogic® GDK 5.0 Installation and Configuration Guide for Windows®
- Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide
- Dialogic® System Release 6.0 PCI for Windows® Software Installation Guide

3.2.1 Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide

New information for Mixed ISDN and Clear Channel on Additional Dialogic® DM3 Boards

The Configuration Guide does not currently include information about these new configuration files. See Section 1.38, "Support for Mixed ISDN and Clear Channel on Additional Dialogic® DM3 Boards", on page 109 in this Release Update.

Update for modifying PVD and PAMD qualification template definitions (IPY00006588 = PTR# 36210)

The Configuration Guide does not include information about the PVD and PAMD qualification templates that are defined in the CONFIG file. The relevant parameters are in the [SigDet] section of the CONFIG file.

In addition, the default PVD and PAMD qualification template definitions are no longer suitable for accurate PVD and PAMD on Dialogic® DM3 Boards and should be modified in accordance with the instructions in Technical Note 030 available on the Customer Support web site at http://www.dialogic.com/support/helpweb/dxall/tnotes/legacy/2000/tn030.htm.

Note: Technical Note 030 was written specifically for Dialogic® System Release 5.1.1 for Windows® Feature Pack 1, but the principle applies to subsequent system releases also.

The recommended default PVD (ID=128193) and PAMD (ID=106561) qualification template definitions are as follows:

```
!User defined Pvd template.
PvdDesc signalId 128193
PvdDesc signalLabel 0000
PvdDesc minSnr 5
PvdDesc maxSnr 600
PvdDesc maxPk 2
PvdDesc maxRing 5
PvdDesc ringThresh 10000
PvdDesc PvdWin 8
PvdDesc PvdVthresh 10000
PvdDesc PvdRbLow 380
PvdDesc PvdRbHigh 510
CreatePvd
!User defined PAMD template.
PamdDesc signalId 106561
PamdDesc signalLabel 0000
PamdDesc minRing 190
PamdDesc mask 1
PamdDesc maxAnsiz1 159
PamdDesc maxAnsiz2 159
PamdDesc maxAnsiz3 159
PamdDesc loHiss 22
PamdDesc hiHiss 16
PamdDesc bhParm 5
PamdDesc cvThresh1 80
PamdDesc cvThresh2 165
PamdDesc maxCvThresh 390
PamdDesc nMaxBroad 2
PamdDesc nMaxErg 65
PamdDesc maxSilence 30
PamdDesc voiceThresh 25
PamdDesc silenceThresh 10000
PamdDesc rjFbandLow 0
PamdDesc rjFbandHigh 0
CreatePamd
```

The default PVD qualification template ID is 128193 (0x1f4c1), but other valid PVD qualification template IDs that can be defined in the CONFIG file are:

- 128194 (0x1f4c2)
- 128195 (0x1f4c3)
- 128196 (0x1f4c4)
- 128197 (0x1f4c5)

The default PAMD qualification template ID is 106561 (0x1a041), but other valid PAMD qualification template IDs that can be defined in the CONFIG file are:

- 106564 (0x1a044)
- 106565 (0x1a045)
- 106566 (0x1a046)
- 106567 (0x1a047)

Update for analog line adaptation utility (LineAdapt)

Because of a new feature in the Service Update, a configuration utility is now 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 the LineAdapt utility should be added to this document. For information about this utility, see Section 1.43, "Analog Line Adaptation Utility (LineAdapt)", on page 137 of this Release Update.

Update for IP support on Dialogic® DI0408LSAR2 Boards

Because of a new feature in the Service Update, Voice over IP (VoIP) capability is now supported on Dialogic® DI0408LSAR2 Switching Boards. Some additional parameters are now applicable to these boards. For information about this feature, including configuration information, see Section 1.45, "IP Support on Dialogic® DI0408LSAR2 Boards", on page 152 and Section 1.46, "Dialogic® DI0408LSAR2 Board Support for Host Systems with Multiple NICs", on page 159 of this Release Update.

Update for adjusting DTMF characteristics

Because of a new feature in the Service Update, you can now adjust DTMF parameter values, such as amplitudes and on/off durations, in the Tone Templates [tonegen] section of a particular media load CONFIG file. For information about this feature, see Section 1.49, "Adjusting DTMF Characteristics through the CONFIG File", on page 174 of this Release Update.

Update to Media Load 11 on page 22 (PTR# 33555/34771)

Media Load 11 description on page 22 should read:

Media Load 11 - Enhanced Voice Plus Conferencing (Dialogic® DM/IP241-1T1-PCI-100BT and DM/IP301-1E1-PCI-100BT Boards only)

- All Enhanced Voice features (see Media Load 2)
- Conferencing resource

Update to Table 1 on page 23 (PTR# 33555/34771)

Table 1 on page 23 should be modified as follows:

- The last two board entries, Dialogic® DM/IP481-2T1-PCI-100BT and DM/IP601-2E1-PCI-100BT, should be ignored.
- The note at the bottom of the table associated with the ‡ symbol should read: Media load 11 only applies to Dialogic® DM/IP single-span boards.

Parameters not applicable to this release

The following parameters, which are documented in the guide, are not applicable in Dialogic® System Release 6.0 PCI for Windows®:

- Derive NETREF Two From
- NETREF Two Clock Rate
- NETREF Two FRU
- Using NETREF Two

Update to **Section 2.4, Media Loads** for new media loads

Because of features introduced in the Service Update, several new media loads are available:

- for Dialogic® DMV3600BP Boards: ML9B-LC
- for Dialogic® DMV1200BTEP Boards: QSB-U3, QSB-ML10, QSB-ML10-LC, QSB-U2, and 10b.
- for Dialogic® DMV600BTEP Boards: DSB-U2

These media loads should be documented in **Section 2.4, Media Loads**. For information about these media loads, see Section 1.68, "New Media Load for Dialogic® DMV3600BP Boards", on page 242, Section 1.69, "New Media Loads for Dialogic® DMV1200BTEP Boards", on page 244, and Section 1.70, "New Media Load for Dialogic® DMV600BTEP Boards", on page 247 of this Release Update.

Update to **Section 2.4, Media Loads** for new Dialogic® DI Board feature

Because of a new feature in the Service Update, **Section 2.4.1.1** about features supported on Dialogic® DI Boards should be updated to indicate that conference bridging is supported so a higher number of maximum parties per conference is possible. For further information about this feature, see Section 1.54, "Conference Bridging on Dialogic® DI Boards", on page 221 of this Release Update.

Update to **Section 2.4.1.2, Dialogic® DM/IP, DM/V, DM/V-A, and DM/V-B Boards**

In **Table 2, Channel Densities by Board and Media Load (Universal)**, the table footnote about echo cancellation should be changed as follows:

- Default configuration is EEC (enhanced EC, 32 ms) for CSP supported ML, unless otherwise indicated or set in the component named [0x2c] in the respective CONFIG file. You can only change it to a lower EC tail length, by changing the CSP parameter 0x2c03 accordingly in the respective CONFIG file. Conferencing EC, however, will always be 16 ms, regardless of the EC parameter setting.

Update to **Section 2.5, CT Bus Clock Fallback** (PTR# 35769)

Reference master fallback is **not** supported and should be deleted from the Section 2.5 introduction and from Figure 5, Clock Fallback. The entire **Section 2.5.2, Reference Master Fallback**, should be deleted.

Update to **Section 3.4, [NFAS] Section**

The third note about NFAS D channel backup (DCBU) supported only on ISDN NI-2 protocol is incorrect. DCBU is supported on 4ESS, 5ESS, and NI-2.

Update to **Chapter 4, Configuration Procedures**

Because of an enhancement in the Service Update, it is no longer necessary to use the fcdgen utility to generate an updated FCD file. All references to fcdgen throughout the Configuration Guide can be ignored. For information on the enhanced way to generate an updated FCD file, see Section 1.35.2, "Automatic FCD File Generation", on page 107 of this Release Update.

Update to **Section 4.3, Starting the Configuration Manager (DCM)**

After the Note just before step 2, add the following Note:

Note: To use remote DCM across firewalls, enable the port used by the DCOM Server, *DCMObj.exe*, in the firewall configuration. *DCMObj.exe* is located in the /Program Files/Dialogic/bin directory. To find out the port used by DCMObj.exe, first use the Windows® Task Manager to find out the PID of *DCMObj.exe*. Once you know the PID, you can use a port usage utility to find out the port used by *DCMObj.exe*. Windows® XP users can run netstat -o to find the port.

Update to **Section 4.4, Selecting a Configuration File Set**

Because of a feature in the Service Update, it is now possible to mix ISDN and CAS protocols on the same Dialogic® DMV600BTEP or DMV1200BTEP Media Board using the Trunk Configuration property sheet of DCM. Table 5, Protocol Groups, should be replaced by the following table:

Group 1	Group 2
4ESS (T1)	DPNSS
5ESS (T1)	DASS2
NTT (T1)	
NI2 (T1)	
DMS (T1)	
QSIGT1 (T1)	
QSIGE1 (E1)	
NET5 (E1)	
T1CC (T1)	
CAS (T1)	
E1CC (E1)	
R2MF (E1)	

For further information about this feature, see Section 1.57, “Mixing ISDN and CAS on Dialogic® DM/V-B Boards”, on page 225 of this Release Update.

Update for new **PDK Configuration Property Sheet**

Because of a new feature in the Service Update, a new PDK Configuration property sheet in DCM is now used to assign country dependent parameter (CDP) file variants to trunks that use CAS or R2MF protocols. Information about this property sheet should be added to **Chapter 4, Configuration Procedures,** following **Section 4.4, Selecting a Configuration File Set**. The new property sheet should also be documented in **Chapter 5, DCM Parameter Reference**. For further information about the new property sheet, see Section 1.35.1, “PDK Configuration Property Sheet”, on page 105 of this Release Update.

Update to **Section 4.5, Setting the TDM Bus Clock Source** (PTR# 30175)

The following note is added to **Section 4.5, Setting the TDM Bus Clock Source**:

Note: When configuring a board that has front-end capability as resource only, the system will not detect this and might select this board as a clock master. In this event, the user must manually configure another board in the system as the clock master.

Updates to **Section 5.5, Physical Property Sheet**

The following description of the DCM parameter **PhysicalSlotNumber** should be added:

PhysicalSlotNumber (PCI Boards)

Description: The **PhysicalSlotNumber** parameter specifies a PCI board’s rotary-switch setting. The rotary-switch setting for Dialogic® DM3 PCI boards can be the same for all boards in the system if the value is set to 0.

Values: 0 to 15

Guidelines: Use the **PhysicalSlotNumber** parameter default value.

The description of the DCM parameter **PciID** should be replaced by the following:

PciID

Description: The **PciID** parameter is a positive integer or hexadecimal value in which the lower 5 bits specify a board's rotary-switch setting (PCI boards) or the physical slot number location of the board (CompactPCI boards). The rotary-switch setting for PCI boards can be the same for all boards in the system if the value is set to 0.

Values: A positive integer or hexadecimal value

Guidelines: The **PciID** parameter is set by the system software and should not be changed by the user.

Update to **Section 5.6, TDM Bus Configuration Property Sheet**

The description of the DCM parameter **Derive Primary Clock From (User Defined)** that is contained in **Section 5.6, TDM Bus Configuration Property Sheet**, is replaced by the following:

Derive Primary Clock From (User Defined)

Description: The **Derive Primary Clock From (User Defined)** parameter specifies the clock source that the **Primary Master FRU** uses to drive the Primary Line.

Values:

- Default [default]: The value of this parameter is to be determined by the system software. Its current value is indicated by the Resolved Equivalent.
- FrontEnd_1: Not applicable to Dialogic® DM3 Boards.
- FrontEnd_2: Not applicable to DM3 Boards.
- FrontEnd_3: Not applicable to DM3 Boards.
- FrontEnd_4: Not applicable to DM3 Boards.
- InternalOscillator: The Primary Master derives clocking from its own internal circuitry.
- NETREF_1: The Primary Master derives clocking from NETREF_1 (CT Bus only)
- NETREF_2: This selection is not supported for this release.

Update to **Chapter 6, CONFIG File Parameter Reference**

Because of a feature in the Service Update, there are new transmit and receive parameters that let you change the volume level of the FSK modem signals sent and received by the board. For information about this feature and the new parameters, see Section 1.26, "New FSK Transmit and Receive Signal Level Parameters", on page 90 of this Release Update.

Update to **Section 6.3, [0x2c] Parameters**

Because of a feature in the Service Update, a new parameter, **0x2c22**, can be manually added to the CONFIG file to disable or lower white noise. This parameter should be documented in **Section 6.3, [0x2c] Parameters.** For more information about this feature and the new parameter, see Section 1.24, "Ability to Lower or Disable White Noise", on page 89 of this Release Update.

Update to **Section 6.4, [encoder] Parameters**

Because of enhancements to the **dx_reciottdata( )** function introduced in the Service Update, a new parameter, **0x416**, can be manually added to the CONFIG file to change the default value for the amount of allowable silence when using RM_ISCR mode. This parameter should be documented in **Section 6.4, [encoder] Parameters**. For information about this feature and the new parameter, see Section 1.72, "dx_reciottdata( ) Enhancements", on page 254 of this Release Update.

Update to **Section 6.7, [0x3b] Parameters**

Information about parameters **0x3b03** and **0x3b04** should be added to this section as follows:

Note: This information is intended for experienced users of the Dialogic® DM3 conferencing feature. Changing the default parameter settings is **not** recommended, as this could introduce negative effects on the audio quality and conferencing experience of the participants.

CSUMS_ActTalkerPartiesMinNum

Number: 0x3b03

Description: Specifies the number of talkers in a conference before Active Talker mode is enabled.

Note: Conference Active Talker mode, though related, should not be confused with the Active Talker detection feature.

Values: 0 [default] to 0xff (255).

Guidelines: Refer to the guidelines for the **CSUMS_SmartScalingPartiesMinNum** parameter below.

CSUMS_SmartScalingPartiesMinNum

Number: 0x3b04

Description: Specifies the number of talkers in a conference before scaling mode is enabled.

Values: 0 [default] to 0xff (255).

Guidelines: Audio conferencing provides a mechanism for audio summation of two or more parties in a conference. There are three possible summing modes, which are controlled by CSUMS parameters **0x3b03** and **0x3b04** in the configuration file.

By default, both active talker and scaling are enabled. When parameters **0x3b03** and **0x3b04** are both set to their default values of 0, the default summing mode is Active Talker Summation mode, which sums the three loudest parties. This is advantageous in **large conferences**. Since only the three loudest parties are summed, background noise is reduced. However, there may be times with **small conferences** when a different summation mode is preferable, for example, with soft speakers or when the energy is too low (as with analog lines). The other summation modes are:

- Smart scaling mode - the summation of all parties, but scaling is only done on the ones who are talking.
- No scaling - pure summation, can be used if you want full voice energy in the conference.

The settings for parameters **0x3b03** and **0x3b04** determine the summing mode as shown in the following table.

Parameter 0x3b03, CSUMS_ActTalkerPartiesMinNum	Parameter 0x3b04, CSUMS_SmartScalingPartiesMinNum	Summing Mode
0 (default	0 (default)	Active Talker Detection (default)
> Conf_MaxTotalParties	0 (default)	Smart Scaling
> Conf_MaxTotalParties	> Conf_MaxTotalParties	No Scaling
Conf_MaxTotalParties is the setting for parameter 0x3926 in the configuration file, e.g., SetParm=0x3926,120 !Conf_MaxTotalParties		

To disable the Active Talker algorithm, set the parameter **0x3b03** to a value larger than the maximum number of conferences per DSP; setting it to **Conf_MaxTotalParties**, or per board total number of parties, will suffice, to a maximum of 255.

Even without Active Talker, scaling is also enabled by default. If not required, set the parameter **0x3b04** to a number larger than the maximum number of parties per DSP, and again using **Conf_MaxTotalParties** will suffice, to a maximum of 255.

Update to **Section 6.9, [lineAdmin.x] Parameters (Digital Voice)**

In the guidelines for the **SignalingType** parameter, the note about NFAS D channel backup (DCBU) supported only on ISDN NI-2 protocol is incorrect. DCBU is supported on 4ESS, 5ESS, and NI-2.

Update to **Section 6.9, [lineAdmin.x] Parameters (Digital Voice)**

Because of a feature in the Service Update, a new parameter, **0x1626**, can be manually added to the CONFIG file for trunk preconditioning. This parameter should be documented in **Section 6.9, [lineAdmin.x] Parameters (Digital Voice)**. For information about this feature and the new parameter, see Section 1.73, “Trunk Preconditioning”, on page 256 of this Release Update.

Update to **Section 6.9, [lineAdmin.x] Parameters (Digital Voice)**

The following new parameter is added in **Section 6.9, [lineAdmin.x] Parameters (Digital Voice)**:

InitialBitPattern (Initial CAS Signaling Bit Pattern)

Number: 0x1625

Description: The **InitialBitPattern** parameter defines the values of the CAS ABCD signaling bits that are transmitted for all channels on the specified line at the time the firmware is downloaded and initialized.

Values: 0x0 to 0xf, where the hexadecimal value represents the binary ABCD bit values. For example, 0xd defines the ABCD bit pattern as 1101.

Guidelines: For a T1 line, the default is 0x0. For an E1 line, the default is 0xd.

Update for **CRC Checking Parameter** (PTR# 31812/32282)

The following information about the **CRC Checking Parameter** should be included in the Configuration Guide:

A T1 front end can run two different framing algorithms when configured as extended superframe (ESF): a default algorithm and an alternate CRC-6 checking algorithm. The CRC-6 checking algorithm allows the circuit to confirm the CRC-6 bits in the received multiframe, as a guard against mimic framing patterns, before forcing a new frame alignment.

The CRC Checking parameter allows you to enable the CRC-6 checking algorithm. This parameter only applies to T1 trunks whose Line Type parameter (0x1601) is set to 1 (dsx1_ESF). For all other Line Types, this parameter is invalid.

To include this parameter and enable CRC checking, you must edit the applicable CONFIG file by adding the following line at the end of each [lineAdmin] section of the CONFIG file:

```
SetParm=0x1624,1! CRC checking OFF=0 (default), CRC checking ON=1
```

After editing the CONFIG file, you will need to generate a new FCD file. Refer to **Section 4.9, Modifying the FCD File Parameters**.

Update to **Section 6.12, [NFAS.x] Parameters**

In the description of the **NFAS_Standby_IntID** parameter, the note about NFAS D channel backup (DCBU) supported only on ISDN NI-2 protocol is incorrect. DCBU is supported on 4ESS, 5ESS, and NI-2.

Update to **Section 6.18, [CCS] Parameters**

Because of a feature in the Service Update, a new parameter, **0x26**, can be manually added to the CONFIG file to enable bearer channel time slots to use a sequentially-ordered logical channel numbering scheme, from 1 to 30, for the QSIG protocol. This parameter should be documented in **Section 6.18, [CCS] Parameters**. For information about this feature and the new parameter, see Section 1.44, “New QSIG Channel Mapping Parameter for E1 Boards”, on page 150 of this Release Update.

Update to **Section 6.24, [TSC] defineBSet Parameters** (IPY00033335)

The information about the **DChanDesc** parameter should be replaced with the following:

DChanDesc (D Channel Identifier)

Description: The **DChanDesc** parameter is an ISDN parameter that identifies which trunk the D-channel resides for this B-set. This parameter is ignored for T1 CAS, clear channel, and Global Call protocols.

Values: 1 to 16

Guidelines: For example, on a board with four T1 ISDN lines, **DChanDesc** is set as follows:

```
defineBSet=10,1,1,24, 0,1,1,1,20,1, 1,1,3,24,0
defineBSet=20,2,1,24, 0,1,1,2,20,1, 1,1,3,24,0
defineBSet=30,3,1,24, 0,1,1,3,20,1, 1,1,3,24,0
defineBSet=40,4,1,24, 0,1,1,4,20,1, 1,1,3,24,0
```

3.2.2 Dialogic® Springware Architecture Products on Windows® Configuration Guide

Parameters not applicable to this release

The following parameters, which are documented in the guide, are not applicable in Dialogic® System Release 6.0 PCI for Windows®:

- Derive NETREF Two From
- NETREF Two Clock Rate

- NETREF Two FRU
- Using NETREF Two

Update for analog line adaptation utility (LineAdapt)

Because of a new feature in the Service Update, a configuration utility is now 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 the LineAdapt utility should be added to this document. For information about this utility, see Section 1.43, "Analog Line Adaptation Utility (LineAdapt)", on page 137 of this Release Update.

New stand-alone configuration supported

Because of a new feature in the Service Update, a stand-alone configuration that will eliminate a single point of failure with respect to clocking is now supported on selected Dialogic® JCT Boards. For information about this feature, including new parameters in DCM, see Section 1.50.2, "Stand-Alone Configuration", on page 180 of this Release Update.

New parameter in *ntt.prm* file

Because of a new feature in the Service Update, a new channel block timer parameter has been added to the *ntt.prm* file for the ISDN NTT protocols. For information about this feature, see Section 1.56, "New Channel Block Timer for NTT Protocol", on page 223 of this Release Update.

Update to **Section 2.3.2, Signal Delay** (PTR# 31601)

In **Section 2.3.2, Signal Delay**, the following note is added after the first paragraph in the section:

Note: Due to Host CPU/PCI Bus loading limitations, the minimum firmware buffer size for systems with more than 20 channels is 256 bytes.

Update to **Section 3.3, Starting the Configuration Manager (DCM)**

In **Section 3.3, Starting the Configuration Manager (DCM)**, after the Note just before step 2, add the following Note:

Note: To use remote DCM across firewalls, enable the port used by the DCOM Server, *DCMObj.exe*, in the firewall configuration. *DCMObj.exe* is located in the /Program Files/Dialogic/bin directory. To find out the port used by *DCMObj.exe*, first use the Windows® Task Manager to find out the PID of *DCMObj.exe*. Once you know the PID, you can use a port usage utility to find out the port used by *DCMObj.exe*. Windows® XP users can run netstat -o to find the port.

Update to **Section 3.5, Setting the TDM Bus Clock Source** (PTR# 30175)

The following note is added to **Section 3.5, Setting the TDM Bus Clock Source**:

Note: When configuring a board that has front-end capability for use as a resource-only board, the system will not detect this and might select this board as a clock master. In this event, the user must manually configure another board in the system as the clock master.

Update to **Section 4.4, Misc Property Sheet**, for **EC_Resource** Parameter (IPY00041018)

The following guideline for using the **EC_Resource** parameter is **incorrect**:

- For boards that support continuous speech processing (CSP), set this parameter to OFF (disabled) and, instead, use the **CSPExtraTimeSlot** parameter to enable echo cancellation.

The guideline should be **changed** to:

- For Dialogic® Springware Boards that support continuous speech processing (CSP), set this parameter to ON (enabled) and also set the **CSPExtraTimeSlot** parameter to ON.

Update to **Section 4.4, Misc Property Sheet**, for **CSPExtraTimeSlot** Parameter (IPY00041018)

The following guideline for using the **CSPExtraTimeSlot** parameter is **incorrect**:

- If you enable CSP for a board, do not enable the board's **EC_Resource** parameter.

The guideline should be **changed** to:

- If you enable CSP for a Dialogic® Springware Board, you must also enable the board's **EC_Resource** parameter.

Update to **Section 4.5, TDM Bus Configuration Property Sheet**

The information about the DCM parameter **Derive Primary Clock From (User Defined)** that is contained in **Section 4.5, TDM Bus Configuration Property Sheet**, is replaced by the following:

Derive Primary Clock From (User Defined)

Description: The **Derive Primary Clock From (User Defined)** parameter specifies the clock source that the Primary Master FRU uses to drive the Primary Line.

Values:

- Default [default]: The value of this parameter is to be determined by the system software. Its current value is indicated by the Resolved Equivalent.
- FrontEnd_1: The Primary Master derives clocking from its own front end network interface. This value only applies when the TDM Bus Type (Resolved) is set to SCbus and the Primary Master FRU (Resolved) is a Dialogic® Springware Board.
- FrontEnd_2: The Primary Master derives clocking from its own second network interface. This value only applies when the TDM Bus Type (Resolved) is set to SCbus and the Primary Master FRU (Resolved) is a Springware Board.
- FrontEnd_3: Not applicable to Springware Boards.
- FrontEnd_4: Not applicable to Springware Boards.
- InternalOscillator: The Primary Master derives clocking from its own internal circuitry.
- NETREF_1: The Primary Master derives clocking from NETREF_1 (CT Bus only).
- NETREF_2: This selection is not supported for this release.

Update to **Section 4.6, Country Property Sheet**

The information about the DCM parameter **Digital Signaling** that is included in **Section 4.6, Country Property Sheet**, is replaced by the following:

Digital Signaling

Description: Allows you to designate E1 time slot 16 for signaling or to choose clear-channel signaling.

Values:

- TS16_SIG [default]: Designates time slot 16 to be used for signaling.
- TS16_CLEAR: Selects clear-channel signaling.

Guidelines: To use this parameter, you must also set a value for the **Country** parameter. Consult the Country Parameter Selection Table in the DCM online help to verify that the value you choose for this parameter can be used for the country selected.

3.2.3 Dialogic® GDK 5.0 Installation and Configuration Guide for Windows®

In addition to the updates described in this section, refer to the separate document *Updates for GDK Version 6.0 and Dialogic® System Release 6.0 PCI for Windows®* for other updates applicable to the *Dialogic® GDK 5.0 Installation and Configuration Guide for Windows®*. Also, for additional information about configuring Dialogic® CPI/2400PCIU-T1 Boards, refer to the tech note at this link (PTR# 33698, 33699):
http://www.dialogic.com/support/helpweb/dxall/tnotes/legacy/2000/tn091.htm

Do not use the procedure in **Section 3.3.4** of the *Dialogic® GDK Version 5.0 Installation and Configuration Guide for Windows®* for configuring the **NETREF One FRU** parameter. Instead, follow the procedure titled **"Setting the TDM Bus Clock Source"** in **Section 4.5** of the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide* (PTR# 24782).

3.2.4 Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide

New **CDP_In_ANIBeforeDNIS** parameter

Because of a new feature in the Service Update, a new parameter, **CDP_In_ANIBeforeDNIS**, has been added to the CDP files for all countries/protocols that use the pdk_r2_io protocol module. This parameter specifies the order of DNIS, ANI, and Category digits. For further information about the new parameter, see Section 1.55, "New Parameter for Order of DNIS and ANI", on page 222 of this Release Update.

Update to **Section 2.4.2, Downloading the Protocol and CDP File on a Windows® System**

Because of a new feature in the Service Update, it is no longer necessary to set up the pdk.cfg file manually and run pdkmanagerregsetup to download the protocol and CDP file. Instead, a new PDK Configuration property sheet in DCM is used to assign country dependent parameter (CDP) file variants to trunks that use CAS or R2MF protocols, and the pdk.cfg file is generated automatically. The need to run pdkmanagerregsetup has been eliminated. For further information about this feature, see Section 1.35.1, "PDK Configuration Property Sheet", on page 105 of this Release Update.

The following note should be added after the first paragraph of this section (PTR# 36373):

Note: If the automatically generated pdk.cfg file is deleted and not present in the %INTEL_DIALOGIC_CFG% directory, then all subsequent attempts to start the Dialogic® services will fail with no discernible error.

Update to **Chapter 43, North American Analog Bidirectional Protocol Parameter Configuration**

Because of a new feature in the Service Update, there are additional parameters in the *pdk_na_an_io.cdp* file to support analog call transfer. For further information about this feature, see Section 1.3, "Analog Call Transfer Support on Dialogic® Springware Boards", on page 38 of this Release Update.

3.2.5 Dialogic® System Release 6.0 PCI for Windows® Software Installation Guide

There are currently no updates to this document.

3.3 OA&M Documentation

This section contains updates to the following documents (click the title to jump to the corresponding section):

- Dialogic® Board Management API Library Reference
- Dialogic® Configuration Manager (DCM) Online Help
- Dialogic® System Software Diagnostics Guide
- Dialogic® System Software for PCI Products on Windows® Administration Guide
- Dialogic® Native Configuration Manager API Library Reference
- Dialogic® SNMP Agent Software for Windows® Administration Guide

3.3.1 Dialogic® Board Management API Library Reference

Update to **brd_Open( )** function

Because of a new feature in the Service Update for firmware assert notification on Dialogic® JCT Boards, the **brd_Open( )** function has a new **mode** parameter, **BRD_FW_ASSERT_ENABLE**. There is also a new event, DMEV_FW_ASSERT. For information about this feature, see Section 1.50.3, "Firmware Assert Notification", on page 181 of this Release Update.

Update to **brd_SendAliveEnable( )** function

The description of the **brd_SendAliveEnable( )** function currently says:

The network interface is taken out of service by providing the network with a protocol-specific out-of-service condition.

Because of a new feature in the Service Update, the network interface is taken out-of-service by sending an Alarm Indication Signal (AIS) toward the network rather than a protocol-specific out-of-service condition. For information about this feature, see Section 1.74, "Extended Board Management API Support for Dialogic® DM3 Boards", on page 258 of this Release Update.

3.3.2 Dialogic® Configuration Manager (DCM) Online Help

Parameters that are not applicable

The following parameters, which are documented in the DCM online help, are not applicable in Dialogic® System Release 6.0 PCI for Windows®:

- Derive NETREF Two From
- NETREF Two Clock Rate
- NETREF Two FRU
- Provide NETREF Two
- Provide NETREF Two From
- Using NETREF Two
- Frequency Resolution

New parameters for Dialogic® DI0408LSAR2 Boards

Because of a new feature in the Service Update for IP support on Dialogic® DI0408LSAR2 Boards, two parameters have been added to DCM: **DI_TOS** and **HostIpMediaNetworkAddress**. For information about these parameters, see Section 1.45, "IP Support on Dialogic® DI0408LSAR2 Boards", on page 152 and Section 1.46, "Dialogic® DI0408LSAR2 Board Support for Host Systems with Multiple NICs", on page 159 of this Release Update.

New parameters for stand-alone configuration

Because of a new feature in the Service Update for stand-alone configuration (applicable to selected Dialogic® JCT Boards), two parameters have been added to DCM: **NFASPrimary** and **Using Cable Mode**. For information about these parameters, see Section 1.50.2, "Stand-Alone Configuration", on page 180 of this Release Update.

Update to **CSPExtraTimeSlot** help topic (IPY00041018)

The following note for using the **CSPExtraTimeSlot** parameter is **incorrect**:

- If you enable CSP for a board, do not enable that board's **EC_Resource** parameter.

The note should be **changed** to:

- If you enable CSP for a Dialogic® Springware Board, you must also enable that board's **EC_Resource** parameter.

Update to **Derive Primary Clock From (User Defined)** help topic

The help topic for **Derive Primary Clock From (User Defined)** should read as follows:

Description: This parameter specifies the clock source that the Primary Master FRU uses to drive the Primary Line.

Settings	Value	Explanation
	Default	The value of this parameter is to be determined by the system software. Its current value is indicated by the Resolved Equivalent.
	FrontEnd_1	The Primary Master derives clocking from its own front end network interface. This value only applies when the TDM Bus Type (Resolved) is set to SCbus and the Primary Master FRU (Resolved) is a Dialogic® Springware Board.
	FrontEnd_2	The Primary Master derives clocking from its own second network interface. This value only applies when the TDM Bus Type (Resolved) is set to SCbus and the Primary Master FRU (Resolved) is a Springware Board.
	FrontEnd_3	Not supported
	FrontEnd_4	Not supported
	InternalOscillator	The Primary Master derives clocking from its own circuitry.
	NETREF_1	The Primary Master derives clocking from NETREF_1 (CT Bus only).
	NETREF_2	Not supported

Update to **Digital Signaling** parameter help topic

The information in the help topic for the **Digital Signaling** parameter should read as follows:

Description: Allows you to designate E1 time slot 16 for signaling or to choose clear-channel signaling.

Values:

- TS16_SIG [default] Designates E1 time slot 16 to be used for signaling.
- TS16_CLEAR Selects clear-channel signaling.

Note: To use this parameter, you must also set a value for the Country parameter. Consult the Country Parameter Selection Table in the DCM online help to verify that the value you choose for this parameter can be used for the country selected.

Update to **DisconnectTone** parameter help topic

In the help topic for the **DisconnectTone** parameter, the **Applicability** field in the Rules section should read:

All Dialogic® Springware Voice Boards and Dialogic® DMV160LP Boards.

In the help topic for the **DisconnectTone** parameter, the **Description** field should read:

Enables or disables support of Disconnect Tone Supervision. Disconnect Tone Supervision allows voice processing boards to sense a disconnect has occurred at the PBX by listening for the PBX disconnect tone.

In the help topic for the **DisconnectTone** parameter, the following note is added to the **Settings** field:

Note: For Dialogic® DMV160LP Boards, this parameter must also be enabled in the CONFIG file associated with the board. For information about enabling Disconnect Tone Supervision using the **Tone_SigID4** parameter in the CONFIG file, see the [CHP] Analog Voice Variant Definitions section of the CONFIG File Parameter Reference chapter in the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Update to **EC_Resource** help topic (IPY00041018)

The following note for using the **EC_Resource** parameter is **incorrect**:

- For boards that support Continuous Speech Processing (CSP), set this parameter to OFF (disabled) and, instead, use the **CSPExtraTimeSlot** parameter to enable echo cancellation.

The note should be **changed** to:

- For Dialogic® Springware Boards that support Continuous Speech Processing (CSP), set this parameter to ON (enabled) and also set the **CSPExtraTimeSlot** parameter to ON.

Update to **Group One Clock Rate, ..., Group Four Clock Rate** parameter help topics

The help topics for the four parameters **Group One Clock Rate, ..., Group Four Clock Rate** should each include the following note:

Note: This parameter does not apply when the Bus Type parameter is set to SCbus.

Update to **FCDFilename** and **PCDFilename** parameter help topics

The following note is added to the help topics for the **FCDFilename** and **PCDFilename** parameters:

Note: Dialogic® HDSI Boards use country-specific PCD and FCD files. Depending on the PCD/FCD files selected for an HDSI Board, the PCM encoding method will be set to either A-law or mu-law, based on the default value for that country. If this value is not the same as the TDM Bus Media Type parameter setting, the HDSI Board will fail to download.

To change the PCM encoding method for the HDSI Board from the default value, you will need to edit the Encoding parameter in the associated Config file, re-generate the FCD file, and then restart the system. For additional information about modifying FCD file parameters, see the *Dialogic® DM3 Architecture PCI Products on Windows® Configuration Guide*.

Update to **PciID** parameter help topic (Physical property sheet)

The information in the help topic for the **PciID** parameter should read as follows:

Description: A positive integer or hexadecimal value in which the lower 5 bits specify a board's rotary-switch setting (PCI boards) or the physical slot number location of the board (CompactPCI boards). The rotary-switch setting for PCI boards can be the same for all PCI boards in the system if it is set to 0.

Note: The PciID parameter is set by the Dialogic® System Software and should not be changed by the user.

Update to **PhysicalSlotNumber** parameter help topic (Physical property sheet)

The information in the help topic for the **PhysicalSlotNumber** parameter should read as follows:

Description: For a PCI board, specifies the board's rotary-switch setting. The rotary-switch setting for Dialogic® DM3 PCI boards can be the same for all boards in the system if the value is set to 0.

For a CompactPCI board, specifies the number of the physical slot in which the board is installed. A value of 1 indicates the first slot in the chassis. (The chassis slot numbers are usually marked on the front of the chassis.)

Settings: For a PCI board, 0 to 15.

For a CompactPCI board, a positive integer or hexadecimal value.

Note: This parameter is read-only and cannot be modified through the DCM.

Update to **SCbus Clock Rate (User Defined)** parameter help topic

The help topic for the **SCbus Clock Rate (User Defined)** parameter should read as follows:

Description: If the bus is running in SCbus mode (the TDM Bus Type parameter is set to SCbus), this parameter determines the clock rate for the SCbus.

Settings	Value	Explanation
	Default	The value of this parameter is to be determined by the system software. Its current value is indicated by the Resolved.
	2MHz	Not currently supported.
	4MHz	The SCbus operates at 4 MHz.
	8MHz	Not currently supported.

Update for UK country parameter file (PTR# 25019)

The UK country parameter file for the Dialogic® D/120JCT-LS Board (*uk_120j.prm*) is missing from the release. If the board is configured for use in the United Kingdom (DCM Country parameter is set to United Kingdom), the system will start, but the system event log will report the following error:

```
Error downloading file. The uk_120j.prm file cannot be found.
```

To correct this problem, locate the file *eu_120j.prm* in the *Program Files\Dialogic\data* directory and rename this file to *uk_120J.prm*.

3.3.3 Dialogic® System Software Diagnostics Guide

Update for Remote Diagnostics Package

A remote diagnostics package is now available that allows you to run Dialogic® diagnostics utilities remotely from a central site. For further information, see Section 1.10, "Remote Diagnostics Package", on page 67 of this Release Update.

Updates to **Chapter 20, ISDN Trace Reference**

Because of a new feature in the Service Update, the ISDNtrace tool has been enhanced to include support for DPNSS tracing. For more information about this feature, see Section 1.29, "Enhanced ISDN Trace Functionality for DPNSS Tracing", on page 96 of this Release Update.

Because of a new feature in the Service Update, the ISDNtrace tool has been enhanced to include new command line options to set the output log file size and to create multiple backup log files to be archived. For more information about this feature, see Section 1.16, "File Management Enhancements for ISDNtrace Tool", on page 74.

Update to **Chapter 21, Telecom Subsystem Summary Tool Reference**

Because of an enhancement in the Service Update, the *its_sysinfo.htm* file now includes a Windows® Package Info section at the beginning of the file. For further information about this feature, see Section 1.61, "Telecom Subsystem Summary Tool (its_sysinfo)", on page 234 of this Release Update.

Update to **Chapter 24, PDK Trace Reference**

Because of a new feature in the Service Update, PDK Trace supports CAS/R2MF/Tone tracing and readable log files. A new option for enhanced tracing for

CAS, R2MF, tone-on and tone-off has been added. For more information about this feature, see Section 1.23, “PDK Trace Supports CAS/R2MF/Tone Tracing”, on page 86.

Update to **Chapter 26, PSTN Diagnostics Tool Reference**

An enhanced version of the PSTN Diagnostics Tool (pstndiag) is provided in the Service Update. The previous version of the tool is still supported. For information about the new version, see Section 1.21.1, “PSTN Diagnostics (pstndiag)”, on page 81 of this Release Update.

Update to **Chapter 28, Runtime Trace Facility (RTF) Reference** (IPY00037518)

The following information about using binary log files should be added to **Section 28.3.2, Logfile Tag**:

For installations with high channel densities, or which have enabled all or most RTF trace levels, the volume of logging may result in an increased CPU utilization by the RtfServer executable as a result of the increased volume of log messages.

As shipped, the RTF log files are generated in ASCII text mode. There is a configuration parameter in the RTF configuration file (*RtfConfigWin.xml* for Windows®, *RtfConfigLinux.xml* for Linux) that allows log files to be generated in either “text” or “binary” format. Testing on high channel density systems with most or all of the RTF trace levels enabled has shown that the generation of binary format RTF log files has less of an impact on CPU usage than does the generation of text format RTF log files.

If the volume of logging results in high CPU usage, then using binary format will reduce the usage.

Enabling Binary Format RTF Log Files

The XML file contains the following line, which allows changes to log file parameters to be made:

```
<Logfile path="$(INTEL_DIALOGIC_DIR)\log" size="300" maxbackups="10"
preserve_size="300" preserve_maxbackups="10"
duplicate_to_debug_console="0" log_format="text" />
```

The “log_format” value controls the type of log files that are written. Valid values for this parameter are “text” and “binary”. Once a change has been made to the XML file, it must be reloaded using the rtftool reload command.

Converting Binary Format RTF Log Files to Text Format

In order for binary log files to be examined, they must be converted into text format. This can be done by using the rtftool export command.

```
rtftool export [-d source_dir | -s source_file]
[-f [dest_file] | -m dest_dir]
```

By default, the name of the text format files generated by this command will be *EXPORT-<RTF binary log file name>*. For example, if the binary format file is named *rtflog-LOCAL-20070306-15h09m26.506s.txt*, then the default name of the generated text format file will be *EXPORT-rtflog-LOCAL-20070306-15h09m26.506s.txt.* This behavior can be overridden using the -f command line option.

The rtftool utility is a stand-alone program, and it is not necessary to have the Dialogic® System Release installed on the system in order to convert RTF log files from binary to text format.

Note: When generating large binary files with RTF, do not split the single large binary file and then use the individual split files with the rtftool utility. Rtftool will not work with chopped binary files.

Update to **Chapter 29, RTFManager Reference** (IPY00037518)

Section 29.5, General Tab, says that binary log format is not supported by the current release. This is not correct; binary log format is supported. For information about binary log files, see the update to **Chapter 28, Runtime Trace Facility (RTF) Reference** above.

Update to **Chapter 30, Status Monitor Reference**

An enhanced version of the Status Monitor tool (statusmon) is provided in the Service Update. The previous version of the tool is still supported. For information about the new version, see Section 1.21.2, "Status Monitor (statusmon)", on page 82 of this Release Update.

3.3.4 Dialogic® System Software for PCI Products on Windows® Administration Guide

Single board start/stop for selected Dialogic® JCT Boards

Because of a new feature in the Service Update, the ability to stop and start a single Dialogic® JCT Board (after the system has initially started) is now supported. Guidelines for performing a single board stop/start should be added to the Administration Guide. For information about this feature, see Section 1.50, "Single Board Start/Stop for Selected Dialogic®JCT Boards", on page 178 of this Release Update

Update to **Section 2.1.1.2, Start Server Only Mode**

Change **Section 2.1.1.2, Start Server Only Mode**, to read:

Selecting the Start Server Only mode from the System/Device autostart submenu causes the Dialogic® Service to start automatically when the system reboots. The boards will be automatically detected, but not started. In this mode, you will need to start the boards manually through the DCM GUI or the NCM API.

This mode allows Windows® NT Service applications to start and stop the boards without any dependency on the Dialogic® Service.

Updates to **Section 3.3, Replacing a Board in the System**

The paragraph preceding Step 1 of the procedure is replaced by the following:

The following procedure describes the basic steps for removing a PCI board and replacing it with a board identical in model and type in the same slot in the system.

Step 7 of the procedure is replaced by the following:

7. Depending on the DCM System/Device autostart option selected, the replacement board will be detected when the system is rebooted and either be started, or remain in the stopped state, allowing you to manually configure and start the board. See Figure 4 for a display of the Device menu.

Note: For the current configuration of the replaced board to be downloaded to the new board, the replacement board must be installed in the same slot as the board that was removed. Otherwise, the new board will be configured with the default values of the replaced board.

- If the **Detect Only** option has been selected from the **System/Device autostart** submenu, the Dialogic® System and the boards will have to be manually started using the DCM GUI (or NCM API). The replacement board will be detected by the system and displayed in the DCM main window, but will not be started automatically. You will have to manually start the new board using the DCM GUI (or NCM API).
- If the **Start Server Only** option has been selected, the Dialogic® System will start automatically when the system is restarted and the replacement board will be detected automatically. You will, however, need to start the replacement board manually using the DCM GUI (or NCM API).
- If the **Start System** option has been selected, the Dialogic® System will be automatically started and the replacement board will be detected by the system, displayed in the DCM main window, and automatically started using the existing system configuration for that board.

3.3.5 Dialogic® Native Configuration Manager API Library Reference

Updates to **NCM_ApplyTrunkConfiguration( )**

On the **NCM_ApplyTrunkConfiguration( )** function reference page, the first two inputs for **NCM_ApplyTrunkConfiguration( )** are not pointers. Therefore the words “pointer to a” should be deleted from the descriptions of the inputs **NCMFamily* pncmFamily** and **NCMDevice* pncmDeviceUnique**. (PTR# 36260)

On the **NCM_ApplyTrunkConfiguration( )** function reference page, the following two parameters should be added:

pMediaLoad	pre-defined sets of supported features. A media load consists of a configuration file set and associated firmware. Universal media loads support voice, fax, and conferencing resources simultaneously. See the “Media Loads Supported” table below.
pErrorMsg	API provides as needed. This is a pointer to BYTE that the API fills with an error message on return. BYTE is defined as an unsigned character.

Media Loads Supported

Boards	Media Loads	Protocols
DMV1200BTEP	UL1	Group 1: CAS, 4ESS, 5ESS, DMS, NI2, NTT, QSIGT1, QSIGE1, NET5, R2MF, T1CC, E1CC
		Group 2: DPNSS, DASS2
	UL2	Group 1: CAS, 4ESS, 5ESS, DMS, NI2, NTT, QSIGT1, QSIGE1, NET5, R2MF, T1CC, E1CC
		Group 2: DPNSS, DASS2

Boards	Media Loads	Protocols
	UL3	Group 1: CAS, 4ESS, 5ESS, DMS, NI2, NTT, QSIGT1, QSIGE1, NET5, R2MF, T1CC, E1CC
		Group 2: DPNSS, DASS2
	ML10B	Group 1: CAS, 4ESS, 5ESS, DMS, NI2, NTT, QSIGT1, QSIGE1, NET5, R2MF, T1CC, E1CC
		Group 2: DPNSS, DASS2
	ML10	Group 1: CAS, 4ESS, 5ESS, DMS, NI2, NTT, QSIGT1, QSIGE1, NET5, R2MF, T1CC, E1CC
		Group 2: DPNSS, DASS2
DMV600BTEP	UL1	Group 1: CAS, 4ESS, 5ESS, DMS, NI2, NTT, QSIGT1, QSIGE1, NET5, R2MF, T1CC, E1CC
		Group 2: DPNSS, DASS2
	UL2	Group 1: CAS, 4ESS, 5ESS, DMS, NI2, NTT, QSIGT1, QSIGE1, NET5, R2MF, T1CC, E1CC
		Group 2: DPNSS, DASS2
DMT160TEC	Tone	Group 1: 4ESS, 5ESS, DMS, NI2, NTT, QSIGT1, QSIGE1, NET5, ISDNT1CC, ISDNE1CC
		Group 2: CAS, T1CC
		Group 3: R2MF, E1CC
DMN160TEC	Network Only	Group 1: 4ESS, 5ESS, DMS, NI2, NTT, QSIGT1, QSIGE1, NET5, ISDNT1CC, ISDNE1CC
	ML10	Group 1: CAS, 4ESS, 5ESS, DMS, NI2, NTT, QSIGT1, QSIGE1, NET5, R2MF, T1CC, E1CC
		Group 2: DPNSS, DASS2
	UL2	Group 1: CAS, 4ESS, 5ESS, DMS, NI2, NTT, QSIGT1, QSIGE1, NET5, R2MF, T1CC, E1CC
		Group 2: DPNSS, DASS2
	ML10B	Group 1: CAS, 4ESS, 5ESS, DMS, NI2, NTT, QSIGT1, QSIGE1, NET5, R2MF, T1CC, E1CC
		Group 2: DPNSS, DASS2
DMV600BTEC	UL1	Group 1: CAS, 4ESS, 5ESS, DMS, NI2, NTT, QSIGT1, QSIGE1, NET5, R2MF, T1CC, E1CC
		Group 2: DPNSS, DASS2

On the **NCM_ApplyTrunkConfiguration( )** function reference page, the sample code should be updated to show the two new parameters:

```
#include <stdio.h>
#include "ncmapi.h"
void main()
{
```

```
NCMRetCode ncmRetCode;
char buffer[300] = {0};

         NCMFamily family;
         family.name = "DM3";
         family.next = NULL;

         NCMDevice UniqueName;
         UniqueName.name = "DMV1200BTEP #1 in slot 2/10";
         UniqueName.next = NULL;

         NCMTrunkConfig ncmTruckConfig[4] = {0};

         NCMFeatureType ncmFeatureType = {0};

         ncmTruckConfig[0].TrunkName  = "Trunk1";
         ncmTruckConfig[0].TrunkValue = "4ESS(T1, Group 1)";
         ncmTruckConfig[0].next      = &(ncmTruckConfig[1]);

         ncmTruckConfig[1].TrunkName  = "Trunk2";
         ncmTruckConfig[1].TrunkValue = "4ESS(T1, Group 1)";
         ncmTruckConfig[1].next      = &(ncmTruckConfig[2]);

         ncmTruckConfig[2].TrunkName  = "Trunk3";
         ncmTruckConfig[2].TrunkValue = "5ESS(T1, Group 1)";
         ncmTruckConfig[2].next      = &(ncmTruckConfig[3]);

ncmTruckConfig[3].TrunkName  = "Trunk4";
         ncmTruckConfig[3].TrunkValue = "4ESS(T1, Group 1)";
         ncmTruckConfig[3].next = NULL;

         strncpy(ncmFeatureType.MediaLoad, "ML10", MEDIA_LOAD_LENGTH);

ncmRetCode = NCM_ApplyTrunkConfiguration(family,UniqueName,
ncmTruckConfig, &ncmFeatureType, reinterpret_cast<unsigned
char*>(buffer));
         if (ncmRetCode != NCM_SUCCESS)
         {
           printf("Error calling NCM_ApplyTrunkConfiguration(). It
returned: %d \n", ncmRetCode;
           printf( " Error Msg:  %s \n", buffer);
         }
              else
              {
                    printf("SUccessful calling
NCM_ApplyTrunkConfiguration\n");
              }
              printf("press any key to exit\n");
              getchar();
}
...
```

3.3.6 Dialogic® SNMP Agent Software for Windows® Administration Guide

Update to **Chapter 2, Stopping and Starting the System**

The following new section is added at the end of **Chapter 2, Stopping and Starting the System**:

Starting the Boardserver

When you use SNMP, you must start the Boardserver in one of the following ways:

- From the Windows® Control Panel, go to **Administrative Tools > Services**, select the Boardserver, and click **Start**.
- Use the Windows® NT Service Control Manager to set the startup mode from Manual to Automatic.
- Use the following command at a command line prompt:
  ```
  net start Boardserver
  ```

3.4 Programming Library Documentation

This section contains updates to the following documents (click the title to jump to the corresponding section):

- Dialogic® Audio Conferencing API Library Reference
- Dialogic® Audio Conferencing API Programming Guide
- Dialogic® Continuous Speech Processing API Library Reference
- Dialogic® Continuous Speech Processing API Programming Guide
- Dialogic® D/42 Series Software API Reference
- Dialogic® D/42 Series User's Guide
- Dialogic® Digital Network Interface Software Reference
- Dialogic® Fax Software Reference
- Dialogic® GDK Programming Reference Manual
- Dialogic® Global Call API Library Reference
- Dialogic® Global Call API Programming Guide
- Dialogic® Global Call Analog Technology Guide
- Dialogic® Global Call E1/T1 CAS/R2 Technology Guide
- Dialogic® Global Call IP Technology Guide
- Dialogic® Global Call ISDN Technology Guide
- Dialogic® Global Call SS7 Technology Guide
- Dialogic® IP Media Library API Programming Guide
- Dialogic® IP Media Library API Library Reference

- Dialogic® ISDN Software Reference
- Dialogic® Modular Station Interface API Library Reference
- Dialogic® Modular Station Interface API Programming Guide
- Dialogic® PBX Integration Board User's Guide
- Dialogic® PBX Integration Software 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

3.4.1 Dialogic® Audio Conferencing API Library Reference

There are currently no updates to this document.

3.4.2 Dialogic® Audio Conferencing API Programming Guide

There are currently no updates to this document.

3.4.3 Dialogic® Continuous Speech Processing API Library Reference

Update for single echo canceller convergence

Because of a new feature in the Service Update, information about single echo canceller convergence should be added to the **ec_stream( )** function description, and information about the ECCH_CONVERGE parameter should be added to the **ec_setparm( )** function description. For information about this feature, see Section 1.63, "Single Echo Canceller Convergence", on page 235 of this Release Update.

3.4.4 Dialogic® Continuous Speech Processing API Programming Guide

Update to **Section 4.1.2, Reserving Extra Time Slots for Streaming to TDM Bus** (IPY00041018)

The paragraph about using the **CSPExtraTimeSlot** parameter to configure the extra time slot should also include the **EC_Resource** parameter, as follows:

On Dialogic® Springware Boards in Linux, you configure this time slot at initialization time in *dialogic.cfg*. On Dialogic® Springware Boards in Windows®, you configure this time slot at initialization time in the Dialogic® Configuration Manager (DCM). Both the **CSPExtraTimeSlot** and **EC_Resource** parameters must be enabled. See the appropriate Configuration Guide for more information about these parameters.

Update to **Chapter 7, Echo Cancellation Convergence Notification**

Because of a new feature in the Service Update, information about single echo canceller convergence should be added to **Chapter 7, Echo Canceller Convergence Notification**. For information about this feature, see Section 1.63, “Single Echo Canceller Convergence”, on page 235 of this Release Update.

3.4.5 Dialogic® D/42 Series Software API Reference

This document has been added to the online bookshelf to support the Dialogic® D/42-NE2 PCI PBX Integration Board, which is now supported in the Dialogic® System Release 6.0 PCI for Windows® Service Update. Other boards referred to in this document are **not** supported in the System Release 6.0 PCI Windows Service Update.

3.4.6 Dialogic® D/42 Series User’s Guide

This document has been added to the online bookshelf to support the Dialogic® D/42-NE2 PCI PBX Integration Board, which is now supported in the Dialogic® System Release 6.0 PCI for Windows® Service Update. Other boards referred to in this document are **not** supported in the System Release 6.0 PCI Windows Service Update.

3.4.7 Dialogic® Digital Network Interface Software Reference

The *Dialogic® Digital Network Interface Software Reference* does not mention the *dtixxx.h* file that includes many defines including:

- NTT_CAS_TEMPLATE_MATCH
- NTT_CAS_TEMPLATE_SEND_END

In **Chapter 5, Function Reference,** beginning on page 31, the following **dt_getstatistics( )** function reference page has been omitted.

dt_getstatistics( )

Name:	int dt_getstatistics(a_hSrlDevice, a_statisticsList, a_mode)	
Inputs:	int a_hSrlDevice	• logical board device handle (for example, dtiB1)
	TSdtStatisticsList* a_statisticsList	• pointer to statistics
	unsigned short a_mode	• synchronous/asynchronous
Returns:	0 for success -1 for failure	
Includes:	srllib.h dtilib.h	
Category:	Statistics Functions	
Mode:	synchronous/asynchronous	

■ Description

The **dt_getstatistics( )** function returns the statistics queried. The application must specify the type of statistics to be queried in the m_StatisticsType field in the TsdtStatisticsList structure. The m_nStatisticsCount field specifies the number of statistics returned. The statistics are available as an array of TSdtLayer1Statistics structures. In asynchronous mode (EV_ASYNC) the list of statistics is part of the event data.

Parameter	Description
a_hSrlDevice	SRL handle for logical board device
a_statisticsList	pointer to TSdtStatisticList structure
a_mode	EV_SYNC or EV_ASYNC

The **dt_getstatistics( )** function uses the following data structures either directly or indirectly:

- dtStatisticsType, which is defined as follows:

```
typedef enum
{
    dtStatisticsType_Invalid = 0, /* No statistics to be collected */
    dtStatisticsType_Layer1,      /* All Layer 1 Statistics */
    dtStatisticsType_Max
}dtStatisticsType;
```

- dtStatisticsMode, which is defined as follows:

```
typedef enum
{
    dtStatisticsMode_Invalid = 0, /* No statistics Mode */
    dtStatisticsMode_Clear,       /* Clear statistics counters */
    dtStatisticsMode_Preserve,    /* Preserve statistics counters */
    dtStatisticsMode_Max
}dtStatisticsMode;
```

- dtLayer1StatisticsId, which is defined as follows:

```
typedef enum
{
    dtLayer1StatisticsId_Invalid = 0,
    dtLayer1StatisticsId_LCV = 1,     /* Line Coding Violations(LCV)           */
    dtLayer1StatisticsId_PCV,         /* Path Coding Violations(PCV)           */
    dtLayer1StatisticsId_ES,          /* Errored Seconds(ES)                   */
    dtLayer1StatisticsId_SES,         /* Severely Errored Seconds(SES)         */
    dtLayer1StatisticsId_UAS,         /* Unavailable Seconds(UAS)              */
    dtLayer1StatisticsId_BES,         /* Bursty Errored Seconds(BES)           */
    dtLayer1StatisticsId_LOFC,        /* Loss of Frame Count(LOFC)             */
    dtLayer1StatisticsId_CSS,         /* Controlled Slip Seconds(CSS)          */
    dtLayer1StatisticsId_SEFS,        /* Severly Errored Framing Seconds(SEFS) */
    dtLayer1StatisticsId_LES,         /* Line Errored Seconds(LES)             */
    dtLayer1StatisticsId_Max
}dtLayer1StatisticsId;
```

- TSdtLayer1Statistics, which is defined as follows:

```
typedef struct SdtLayer1Statistics
{
    dtLayer1StatisticsId    m_Layer1StatisticsId;
    unsigned int            m_nIntervalTotal;
    unsigned int            m_nCurrentIntervalTimer;
    unsigned int            m_nCurrentValue;
    unsigned int            m_nPreviousValue;
}TSdtLayer1Statistics;
```

- TSdtStatisticsList, which is defined as follows:

```
typedef struct SdtStatisticsList
{
   unsigned int m_nVersion; /* Version of this structure */
   dtStatisticsType  m_StatisticsType;   /* Statistics Type  */
   dtStatisticsMode  m_StatisticsMode;   /* Statistics Mode  */
   unsigned int m_nStatisticsCount;      /* Statistics Count */
   union
   {
        TSdtLayer1Statistics  m_Layer1Statistics[dtLayer1StatisticsId_Max];
   }m_Stats;
} TSdtStatisticsList;
```

Cautions

None.

Errors

Possible errors for this function include:

EDT_INVTS
    Invalid DTI device handle

EDT_PARAMERR
    Invalid parameter

EDT_TMOERR
    Synchronous function timed out waiting for reply

■ Example

```
/* OS Header Files */
#ifdef WIN32
#include <windows.h>
#include <process.h>    /* _beginthread, _endthread */
#include <conio.h>
#else
#include <unistd.h>
#endif
#include <stdio.h>
#include <iostream.h>
#include <iomanip.h>
#include <errno.h>
#include <stddef.h>
#include <stdlib.h>
#include <string.h>
#include <signal.h>
#include <sys/timeb.h>
#include <time.h>

/* Dialogic Header Files */
#include <gcip.h>
#include <gclib.h>
#include <gcisdn.h>
#include <dtilib.h>
#include <srllib.h>

bool repeat = true;     /* Global repeat flag and video variable */
bool EventReceived = true;
LINEDEV a_LineDev=0;
LINEDEV a_BoardDev=0;
void sig_hdlr(int temp);
void OpenBoard(void);

long EventHandler (unsigned long temp)
{
    unsigned int Loop=0;
    int dev=sr_getevtdev();
    long event=sr_getevttype();
    TSdtStatisticsList* myStatisticsList=(TSdtStatisticsList*) sr_getevtdatap();;
    printf("DevH = %d Event = 0x%X\n",dev,event);
    if(event==DTEV_GETSTATISTICS)
    {
        printf("TSdtStatisticsList - Version(%d) StatisticsType(0x%X) Count(%d) Mode(%d)\n",
            myStatisticsList->m_nVersion,myStatisticsList->m_StatisticsType,
            myStatisticsList->m_nStatisticsCount,myStatisticsList->m_StatisticsMode);

        for(Loop=0;Loop<myStatisticsList->m_nStatisticsCount;Loop++)
        {
            printf("TSLayer1Statistics(%d) - Version(%d) StatisticsId(%d) IntervalTotal(%d) 
CurrentIntervalTimer(%d) CurrentValue(%d) PreviousValue(%d)\n",
                Loop,
                myStatisticsList->m_Stats.m_Layer1Statistics[Loop].m_nVersion,
                myStatisticsList->m_Stats.m_Layer1Statistics[Loop].m_Layer1StatisticsId,
                myStatisticsList->m_Stats.m_Layer1Statistics[Loop].m_nIntervalTotal,
                myStatisticsList->m_Stats.m_Layer1Statistics[Loop].m_nCurrentIntervalTimer,
                myStatisticsList->m_Stats.m_Layer1Statistics[Loop].m_nCurrentValue,
                myStatisticsList->m_Stats.m_Layer1Statistics[Loop].m_nPreviousValue);
        }
        EventReceived=true;
    }
    return 0;
}
```

```
int main(void)
{
        /* Start GlobalCall */
    signal(SIGINT,(void (*)(int))sig_hdlr);
    signal(SIGTERM,(void (*)(int))sig_hdlr);
    if (gc_Start(NULL) != GC_SUCCESS) {
        printf("gc_Start(startp = NULL) Failed\n");
        exit(0);
    }
    OpenBoard();
    gc_Close(a_BoardDev);
    gc_Stop();
    return 0;
}

void sig_hdlr(int temp)
{
    cout << "Inside sig_hdlr -> Resetting repeat flag!!" << endl;
    repeat=false;
}

void OpenBoard(void)
{
    TSdtStatisticsList myStatisticsList;
    int t_NetworkBaordDev=0;
    char a_DeviceName[120];
    strcpy(a_DeviceName,":N_dtiB1:P_ISDN");
    /* Open the board device */
    if (gc_OpenEx(&a_BoardDev,a_DeviceName, EV_SYNC, 0) != GC_SUCCESS)
    {
        printf("gc_OpenEx() failed for :%s\n",a_DeviceName);
        exit(0);
    }
    else
        printf("gc_OpenEx() successful for %s- Device Handle = %d\n",
                a_DeviceName,a_BoardDev);
    Sleep(7000);
    if (sr_enbhdlr(EV_ANYDEV, EV_ANYEVT, &EventHandler) < 0)
            cout << "Error enabling the event handler" << endl;

//Query All Layer1 metrics and clear the counters after the query
    memset(&myStatisticsList,0,sizeof(TSdtStatisticsList));
    myStatisticsList.m_nVersion=TSdtStatisticsList_VERSION_0;
    myStatisticsList.m_StatisticsType=dtStatisticsType_Layer1;
    myStatisticsList.m_StatisticsMode=dtStatisticsMode_Preserve;

    while(repeat)
    {
        if(EventReceived)
        {
            Sleep(5000);
            gc_GetNetworkH(a_BoardDev,&t_NetworkBaordDev);
            EventReceived=false;
            printf("Trying dt_getstatistics\n");
            if(dt_getstatistics(t_NetworkBaordDev,&myStatisticsList,EV_ASYNC) != 0)
            {
                printf("dt_getstatistics failed on %s Error = %s\n",
                    ATDV_NAMEP(t_NetworkBaordDev),ATDV_ERRMSGP(t_NetworkBaordDev));
            repeat = false;
            }
        }
        else Sleep( 1000 );
    }
}
```

See Also

None.

In **Appendix B - Message Blocks**, the following Command Message Blocks were omitted:

DTCAS_CLEAR_ALL_TEMPLATE

This command clears all templates for a particular channel. The **devh** handle must be a valid DTI channel device handle. The reply message code, DTCAS_CLEAR_ALL_TEMPLATE_COMPLETE, is received in response to this command.

The typedef for the DTCAS_CLEAR_ALL_TEMPLATE structure is as follows:

```
typedef struct t_clear_all_template_msg

{
    unsigned char  msg_code;
    unsigned char  rfu;
    unsigned short template_id;
} DTCAS_CLEAR_ALL_TEMPLATE_MSG;
```

Parameter	Description
msg_code	identifies the message type and must be set to DTCAS_CLEAR_ALL_TEMPLATE
rfu	reserved; must be set to 0 for future compatibility
template_id	specifies the template identifier

DTCAS_GET_TEMPLATE

This command gets the template for a particular channel. The **devh** handle must be a valid DTI channel device handle. The reply message code, DTCAS_GET_TEMPLATE_COMPLETE, is received in response to this command.

The typedef for the DTCAS_GET_TEMPLATE structure is as follows:

```
typedef struct t_get_template_msg
{
    unsigned char  msg_code;
    unsigned char  rfu;
    unsigned short template_id;
} DTCAS_GET_TEMPLATE_MSG;
```

Parameter	Description
msg_code	identifies the message type and must be set to DTCAS_GET_TEMPLATE
rfu	reserved; must be set to 0 for future compatibility
template_id	specifies the template identifier

DTCAS_GET_NEXT_TEMPLATE

This command gets the template for a particular channel. The **devh** handle must be a valid DTI channel device handle. The reply message code, DTCAS_GET_NEXT_TEMPLATE_COMPLETE, is received in response to this command.

The typedef for the DTCAS_GET_NEXT_TEMPLATE structure is as follows:

```
typedef struct t_get_next_template_msg
{
    unsigned char  msg_code;
    unsigned char  rfu;
    unsigned short template_id;
} DTCAS_GET_NEXT_TEMPLATE_MSG;
```

Parameter	Description
msg_code	identifies the message type and must be set to DTCAS_GET_NEXT_TEMPLATE
rfu	reserved; must be set to 0 for future compatibility
template_id	specifies the template identifier

In **Appendix B - Message Blocks**, the following Reply Message Blocks were omitted:

DTCAS_CLEAR_ALL_TEMPLATE_COMPLETE

This reply message is sent in response to a DTCAS_CLEAR_ALL_TEMPLATE command. The result code within the reply message block indicates the success or failure of the command. The buffer referenced by the **replymsgp** argument will contain a valid DTCAS_REPLY_MSG message block if **dt_castmgmt( )** completes successfully.

The typedef for the DTCAS_REPLY_MSG structure is as follows:

```
typedef struct t_create_reply_msg {
    unsigned char msg_code;
    unsigned char rfu;
    unsigned short template_id;
    unsigned short result;
} DTCAS_REPLY_MSG;
```

Parameter	Description
msg_code	identifies the message type; must be set to DTCAS_CLEAR_ALL_TEMPLATE_COMPLETE
rfu	reserved; must be set to 0 for future compatibility
template_id	specifies the template identifier
result	indicates the success or failure of the command. This field is set to 0 on success, or one of the following error values if the command fails: • DTCAS_ERR_TEMPLATE_NOT_DEFINED – The template was not found in the template table • DTCAS_ERR_TEMPLATE_TABLE_EMPTY – The template table is empty; no templates are defined

DTCAS_GET_TEMPLATE_COMPLETE

This reply message is sent in response to a DTCAS_GET_TEMPLATE command. The result code within the reply message block indicates the success or failure of the command. The buffer referenced by the **replymsgp** argument will contain a valid DTCAS_REPLY_MSG message block if **dt_castmgmt( )** completes successfully.

The typedef for the DTCAS_REPLY_MSG structure is as follows:

```
typedef struct t_create_reply_msg {
    unsigned char msg_code;
    unsigned char rfu;
    unsigned short template_id;
    unsigned short result;
} DTCAS_REPLY_MSG;
```

Parameter	Description
msg_code	identifies the message type; must be set to DTCAS_GET_TEMPLATE_COMPLETE
rfu	reserved; must be set to 0 for future compatibility
template_id	specifies the template identifier
result	indicates the success or failure of the command. This field set to 0 on success, or one of the following error values if the command fails: • DTCAS_ERR_TEMPLATE_NOT_DEFINED – The template was not found in the template table • DTCAS_ERR_TEMPLATE_TABLE_EMPTY – The template table is empty; no templates are defined

DTCAS_GET_NEXT_TEMPLATE_COMPLETE

This reply message is sent in response to a DTCAS_GET_NEXT_TEMPLATE command. The result code within the reply message block indicates the success or failure of the command. The buffer referenced by the **replymsgp** argument will contain a valid DTCAS_REPLY_MSG message block if **dt_castmgmt( )** completes successfully. The typedef for the DTCAS_REPLY_MSG structure is as follows:

```
typedef struct t_create_reply_msg {
    unsigned char msg_code;
    unsigned char rfu;
    unsigned short template_id;
    unsigned short result;
} DTCAS_REPLY_MSG;
```

Parameter	Description
msg_code	identifies the message type; must be set to DTCAS_GET_NEXT_TEMPLATE_COMPLETE
rfu	reserved; must be set to 0 for future compatibility

template_id	specifies the template identifier
result	indicates the success or failure of the command. This field set to 0 on success, or one of the following error values if the command fails: • DTCAS_ERR_TEMPLATE_NOT_DEFINED – The template was not found in the template table • DTCAS_ERR_TEMPLATE_TABLE_EMPTY – The template table is empty; no templates are defined • DTCAS_ERR_END_TMPL_TABLE- The next template was not found; no other templates are defined

3.4.8 Dialogic® Fax Software Reference

Updates to **Chapter 3, Fax API for DM3**

The following should be added as a note in **Section 3.3, Programming Considerations** (PTR# 36674):

- **Note:** All programmers for Dialogic® DM3 fax devices need to be aware that unrouting cannot be accomplished while the fax device is busy. If faxing is in an unknown state, first call **fx_stopch( )** and when that terminates, then route or unroute via **fx_listen( )** or **fx_unlisten**( ) respectively.

The following bullet should be added to **Section 3.3, Programming Considerations** for Dialogic® DM3 Boards (IPY00006570 = PTR# 35992):

- When sending raw and ASCII files, the width of the image is limited to 1728 pixels per line. The io_width field in the DF_IOTT data structure only supports the DF_WID1728 value.

Update to **Section 5.5.6, Sending TIFF/F Files** (IPY00006556 = PTR# 35326)

In the “Handling Multi-Page TIFF/F Files” subsection on page 56, the following note should be added:

- **Note:** On Dialogic® DM3 Boards, DFC_EOM is not supported.

Update to **Section 5.5.9, Setting Phase D Continuation Values** (IPY00006556 = PTR# 35326)

In the DFC_EOM row in Table 10 on page 59, the following note should be added:

- **Note:** On Dialogic® DM3 Boards, DFC_EOM is not supported.

Update to **Section 5.6.3, Defining a Fax Page Header** (IPY00006520 = PTR# 36259)

The first paragraph in this section is incomplete and should be revised as follows.

Fax page header parameters can be set to print a special line of text in a compressed font at the top of every transmitted fax page. This is referred to as the user-definable page header option in Table 1, “Key Fax Features and Specifications.”

There are two possible formats for the fax page header, which is controlled by the FC_HDRATTRIB and other FC_HDRname parameters in **fx_setparm( )**. The default format specified in FC_HDRATTRIB is DF_HDRFMT1.

To create a custom fax page header, set the FC_HDRATTRIB parameter to DF_HDRFMT2 and set the FC_HDRUSER2 parameter to the string to be displayed. The string in FC_HDRUSER2 may contain %R and %P to display the remote ID and page number.

For more information, see the parameter descriptions in the **fx_setparm( )** function reference section.

Update to **Section 5.7.5, Select Resolution for Fax Transmission** (IPY00040796)
This section refers to the **fx_sendfax( ) sndflag** argument resolution settings of DF_TXRESLO and DF_TXRESHI. There are two new values for the **sndflag** argument, DF_TXRES_300_300 and DF_TXRES_200_400. Refer to the update for **Chapter 10, Fax Library Function Reference**, for **ATFX_RESLN( )**, **ATFX_WIDTH( )**, **fx_rcvfax( ),** and **fx_sendfax( )** functions below.

Update to **Section 6.2, Setting Parameters for Receive Fax**
Because of a new feature in the Service Update, a new fax parameter, **FC_MDM_RX_LVL**, has been added to allow setting of the fax modem receiver sensitivity from -43 dBm to -47 dBm. This parameter is supported on Dialogic® Springware Fax Boards only. For information about this new parameter, see Section 1.33, "New Fax Parameter for Modem Receive Level", on page 99.

Update to **Section 6.2.2, Storing Incoming Fax Data** (IPY00031917 = PTR# 27337)
The following note should be added under **Storing in Multiple TIFF/F Files**:

- **Note:** If **fx_sendfax**( ) is called to send a multiple-page TIFF/F with io_phdcont=DFC_EOM, once the first page of the fax is received, a TDX_PHASED event is issued but no TFX_FAXRECV event is returned. TFX_FAXRECV is returned when all fax pages are transmitted.

Update to **Section 6.3.8, Resolution for Storing Incoming Fax Data** (IPY00040796)
This section refers to the **fx_rcvfax( ) rcvflag** argument resolution settings of DF_RXRESLO and DF_RXRESHI. There are two new values for the **rcvflag** argument, DF_RXRES_300_300 and DF_RXRES_200_400. Refer to the update for **Chapter 10, Fax Library Function Reference**, for **ATFX_RESLN( )**, **ATFX_WIDTH( )**, **fx_rcvfax( ),** and **fx_sendfax( )** functions below.

Update to **Section 6.3.8, Resolution for Storing Incoming Fax Data** (PTR# 33036)
The following should be added as a note in **Section 6.3.8, Resolution for Storing Incoming Fax Data**:

- **Note:** On Dialogic® DM3 Boards, you cannot change the resolution of an incoming fax. The resolution specified for fax transmission is used for fax reception.

Update to **Section 8.3.2, DF_ASCIIDATA Field Descriptions** (IPY00037855)
In **Table 12, DF_ASCIIDATA Fields,** the description of the font field states that the default fax font provides 10 characters per inch spacing. This statement should be changed as follows:

The default fax font character spacing may differ on various Windows® operating system releases depending on the OEM font supplied on the particular system.

Updates to **Section 8.6.2, DF_IOTT Field Descriptions** (IPY00040796)
In **Table 14, DF_IOTT Fields,** the description of the io_resln field should have two more values:

- DF_RES_300_300 - 300 (horizontal) x 300 (vertical) resolution (DM3 Boards only)
- DF_RES_200_400 - 200 (horizontal) x 400 (vertical) resolution (DM3 Boards only)

The description of the io_width field should have one more value:

- DF_WID2592 - 2592 pixels per line (only for vertical resolution DF_RES_300_300, DM3 Boards only)

Update to **Chapter 10, Fax Library Function Reference**, for **ATFX_RESLN( )**, **ATFX_WIDTH( )**, **fx_rcvfax( ),** and **fx_sendfax( )** Functions (IPY00040796)

There are two additional return values for the **ATFX_RESLN( )** function, DF_RES_300_300 and DF_RES_200_400. These values are applicable to Dialogic® DM3 Boards only. Additionally, the function will now return 0 when the resolution is not supported. In the description of the **ATFX_RESLN( )** function, the list of valid values should be:

DF_RESHI	High vertical resolution (fine) - 196 lines or pels per inch
DF_RESLO	Low vertical resolution (coarse) - 98 lines or pels per inch
DF_RES_300_300	300 (horizontal) x 300 (vertical) resolution (DM3 Boards only)
DF_RES_200_400	200 (horizontal) x 400 (vertical) resolution (DM3 Boards only)
0	Resolution is not supported

The new resolution values can be specified in the **fx_rcvfax( ) rcvflag** parameter as DF_RXRES_300_300 and DF_RXRES_200_400. The new resolution values can be specified in the **fx_sendfax( ) sndflag** parameter as DF_TXRES_300_300 and DF_TXRES_200_400.

The **ATFX_WIDTH( )** function also has an additional return value:

2592	(pixels per line) Only for vertical resolution DF_RES_300_300.

See the related documentation update for the DF_IOTT structure above.

3.4.9 Dialogic® GDK Programming Reference Manual

Update to **Chapter 4, Queue Record Programming** (IPY00040964)

In the **Queue Record Fields** section, the following information should be added to the description of the **fn_received** field:

When an inbound fax transmission terminates prematurely, resulting in an invalid tiff file, GDK deletes the tiff file and the **fn_received** field in the queue record will be blank.

3.4.10 Dialogic® Global Call API Library Reference

Update to **Section 1.15, Global Call Function Support by Technology** section

Because of a new feature in the Service Update, this section should be updated to indicate that the GCAMS functions (with the exception of **gc_TransmitAlarms( )** and **gc_StopTransmitAlarms( )**) are supported for SS7 technology. Also, the individual GCAMS function reference pages should be updated to indicate support for SS7 technology.

Clarification of linedev parameter for several functions (PTR# 32501)

For the **gc_GetAlarmConfiguration( )**, **gc_GetAlarmSourceObjectList( )**, **gc_GetAlarmSourceObjectNetworkID( )**, **gc_SetAlarmConfiguration( )**, **gc_SetAlarmNotifyAll( )**, and **gc_TransmitAlarms( )** functions, the description of the linedev parameter should specify that it is the Global Call line device handle.

Update to **gc_DropCall( )** (PTR# 34237)

On the **gc_DropCall( )** function reference page, the following caution should be added:

- With CAS protocols, the GCEV_DROPCALL event may be delayed when **gc_DropCall**( ) is called. GCEV_DROPCALL is sent to the application only when the channel becomes Idle. This is expected behavior of a CAS protocol. In the Offered state (ringing), there is no way for the receiving side to tell the calling side to stop ringing. It will not happen until the CO times out (ring no answer) and drops their bits to IDLE. That is why the GCEV_DROPCALL event may seem to be "delayed" in this situation; it is affected by the time-out time governed by CO.

Updates to **gc_GetCallInfo( )**

Because of a new feature in the Service Update, the **gc_GetCallInfo( )** CATEGORY_DIGIT parameter is now supported on Dialogic® DM3 Boards. The **gc_GetCallInfo( )** function reference page (in particular, **Table 6, gc_GetCallInfo( ) info_id Parameter ID Definitions**) should be updated to indicate this.

Because of a new feature in the Service Update, a new billing type has been added to CALLINFOTYPE called "CHARGE WITH CLEARING FROM INBOUND." The **gc_GetCallInfo( )** function reference page (in particular, **Table 6, gc_GetCallInfo( ) info_id Parameter ID Definitions**) should be updated to indicate this. In addition, **gc_GetCallInfo(CALLINFOTYPE)** is now supported on Dialogic® DM3 Boards. For information about this feature and the new billing type, see Section 1.27, "Support for Reporting Billing Type", on page 92 of this Release Update.

Update to **gc_GetLinedevState( )** (PTR# 32616)

In the **gc_GetLinedevState( )** function example, there is an error in two of the printf statements.

Instead of:

```
printf("D Channel Status: %s\n", statebuf);
printf("B Channel Status: %s\n", statebuf);
```

the %s arguments should be changed to **%d**. The correct statements are:

```
printf("D Channel Status: %d\n", statebuf);
printf("B Channel Status: %d\n", statebuf);
```

Update to **gc_InitXfer( )** (IPY00038401)

In the code example, the **gc_InitXfer( )** parameters are shown in the wrong order. The line:

```
if (gc_InitXfer(pline->crn, NULL, &gc_pRetParmBlk, EV_ASYNC) == -1)
```

should be changed to:

```
if (gc_InitXfer(pline->crn, &gc_pRetParmBlk, NULL, EV_ASYNC) == -1)
```

Updates to **gc_MakeCall( )**

On the **gc_MakeCall( )** function reference page, the following caution should be added (PTR# 33852):

- In synchronous mode, calls to **gc_MakeCall**( ) must be serialized. Multiple gc_MakeCalls cannot be made on the same channel from multiple threads.

On the **gc_MakeCall( )** function reference page, on page 226, there is a change to the following paragraph (PTR# 35965):

In the asynchronous mode, if the function is successfully initiated but connection is not achieved (no GCEV_CONNECTED event returned), then the application must issue **gc_DropCall**( ) and

gc_ReleaseCallEx( ) functions to terminate the call completely. If GCEV_TASKFAIL is received, just use **gc_ReleaseCallEx**( ) to terminate the call; there is no need to use **gc_DropCall**( ).

The last sentence of this paragraph is incorrect and should be deleted. Information about error handling in asynchronous mode, including what to do when GCEV_TASKFAIL is received, appears later on page 226.

Update to **gc_util_delete_parm_blk( )**, **gc_util_find_parm( )**, and **gc_util_next_parm( )** (PTR# 32544)

On the **gc_util_delete_parm_blk( )**, **gc_util_find_parm( )**, and **gc_util_next_parm( )** function reference pages, the Errors section is not correct. The **gc_util_delete_parm_blk( )** function returns nothing. The **gc_util_find_parm( )** and **gc_util_next_parm( )** functions return NULL if there is an error.

Update to **gc_SndMsg( )**

Because of a new feature in the Service Update, the **gc_SndMsg( )** function has a new message type, SndMsg_RawEEM. For information about this message type, see Section 1.34, “Ability to Send and Receive DPNSS End to End Messages”, on page 100 of this Release Update.

Updates for analog call transfer support on Dialogic® Springware Boards

Because of a new feature in the Service Update, the **gc_BlindTransfer( )**, **gc_SetupTransfer( )**, **gc_CompleteTransfer( )**, and **gc_SwapHold( )** functions are now supported for Dialogic® Springware Analog technology. **Table 1, Global Call Function Support by Technology**, and the individual function reference pages should be updated to indicate this.

Updates for call transfer support on Dialogic® DMV160LP Board

Because of a new feature in the Service Update, the **gc_SetupTransfer( )**, **gc_CompleteTransfer( )**, and **gc_SwapHold( )** functions are now supported for Dialogic® DM3 Analog technology (DMV160LP Board only). **Table 1, Global Call Function Support by Technology**, and the individual function reference pages should be updated to indicate this.

New alarms for Dialogic® DM3 Boards

Because of a new feature in the Service Update, several new T1/E1 alarms are supported. For information about these alarms, see Section 1.60, “Enhanced GCAMS on Dialogic® DM3 Boards”, on page 233 of this Release Update.

Update to the **Data Structures** chapter

Because of a new feature in the Service Update, dynamically retrieving and modifying selected protocol parameters when using Dialogic® DM3 Boards, information about some new and modified data structures should be added. For information about the new feature, see Section 1.42, “Dynamically Retrieving and Modifying Selected Protocol Parameters When Using Dialogic® DM3 Boards”, on page 113 of this Release Update. Information about the data structures is in Section 1.42.2, “Extended and New Data Structures”, on page 133.

Update to GCLIB_MAKECALL_BLK

On the GCLIB_MAKECALL_BLK data structure reference page, in the Description section, GC**MAKE**CALLBLK_DEFAULT should be changed to GC**MK**CALLBLK_DEFAULT.

Update to GCEV_EXTENSION

Because of a new feature in the Service Update, a new event has been added, EXTENSIONEVT_RAWEEM. For further information about the new feature, see Section 1.33, “New Fax Parameter for Modem Receive Level”, on page 99 of this Release Update.

3.4.11 Dialogic® Global Call API Programming Guide

Figure 27 and Figure 28 have incorrect titles and appear in the wrong order and position (PTR# 32481):

The existing Figure 28 should have the caption “Call State Model for Supervised and Unsupervised Transfers” and should appear in Section 3.6.4.2, “Supervised Transfers” immediately before the existing Figure 27. The paragraph immediately before the figure should read:

The call state transitions that occur during a supervised transfer are shown in Figure 27 (which also shows the call state transitions for an unsupervised transfer).

The existing Figure 27 should have the caption “Call Termination by the Network or Application During a Transfer” and now becomes the new Figure 28. The paragraph immediately before the figure should read:

If the network or application terminates a call during a transfer, call state transitions are as shown in Figure 28.

Finally, in Section 3.6.4.3, “Unsupervised Transfers” the last paragraph should read:

Figure 27 illustrates the call state transitions that occur in an unsupervised transfer, which basically includes only:

- The transition of Call 1 from the Connected to the Idle state (invoked by the **gc_BlindTransfer( )** function)
- The transition of Call 1 from the Idle to the Null state (invoked by the **gc_ReleaseCallEx( )** function)

Update to **Section 7.2.3, Configuring Call Progress Analysis on a Per Call Basis**

Because of a new feature in the Service Update, new custom special information tones (SITs) are allowed and are reported to the application via the GCEV_DISCONNECTED event once any one of them is detected via Dialogic® Global Call Software. For further information about this feature, see Section 1.8, “Enhanced Special Information Tones on Dialogic® DM3 Boards Using Voice and Global Call APIs”, on page 55 of this Release Update.

Update to **Section 7.2.4, Setting Call Analysis Attributes on a Per Call Basis** (IPY00006588 = PTR# 36210)

This section (page 121) provides incorrect information regarding the CCPARM_CA_PVD_QTEMP parameter. The correct information is as follows:

CCPARM_CA_PVD_QTEMP

PVD Qualification Template. Specifies which PVD template to use. Possible values are:
- PAMD_QUAL1TMP – Predefined qualification template. This is the default value.
- -1 – No qualification template

The CCPARM_CA_PVD_QTEMP parameter can also be set to a qualification template ID that is defined in the CONFIG file.

Setting CCPARM_CA_PVD_QTEMP to a value of PAMD_QUAL2TMP is **not** supported.

CCPARM_CA_PAMD_QTEMP

PAMD Qualification Template. Specifies which PAMD template to use. Possible values are:

- PAMD_QUAL1TMP – Predefined qualification template. This is the default value.
- -1 – No qualification template

The CCPARM_CA_PAMD_QTEMP parameter can also be set to a qualification template ID that is defined in the CONFIG file.

Setting CCPARM_CA_PAMD_QTEMP to a value of PAMD_QUAL2TMP is **not** supported.

Note: The default qualification templates are no longer suitable for accurate PVD and PAMD on Dialogic® DM3 Boards and should be modified in accordance with the instructions in Technical Note 030 available on the Customer Support web site at http://www.dialogic.com/support/helpweb/dxall/tnotes/legacy/2000/tn030.htm. The technical note is written specifically for Dialogic® System Release 5.1.1 for Windows® Feature Pack 1, but the principle applies to subsequent releases also.

Update to **Chapter 12, Debugging**

In addition to the various Dialogic® Global Call Technology Guides that are referenced here for technology-specific debugging information, this chapter should also mention the Runtime Trace Facility (RTF) logging tool. For information about the RTF tool, see Section 1.41, “Enhancements to Runtime Trace Facility (RTF) Logging”, on page 113 of this Release Update.

3.4.12 Dialogic® Global Call Analog Technology Guide

Update for **gc_GetANI( )** (PTR# 33202)

Section 3.7, gc_GetANI( ), should include the following note:

Note: The **gc_GetANI**( ) function is deprecated in this software release. The suggested equivalent is **gc_GetCallInfo**( ).

Updates for **gc_MakeCall( )** (PTR# 32379)

When using the PDK analog protocol, **gc_MakeCall( )** returns the GCEV_TASKFAIL event when no dial tone is detected. This should be added to Table 9, Analog Call Conditions and Results, in the “Event/Return Value” column for the “No dial tone detected” condition. This should also be added to **Section 2.5, Call Progress and Call Analysis**, in the paragraph about **gc_MakeCall( )** as follows:

The **gc_MakeCall**( ) function defines the maximum time (in seconds) within which a call must be answered. Within that interval, busy and ringback tones can be detected. Dialogic® Global Call Software will disconnect an outbound call and report a GCEV_CALLSTATUS, GCEV_DISCONNECTED, **or GCEV_TASKFAIL** event to the application if the call is not answered within the default time-outs defined by the protocol or the **gc_MakeCall**( ) function.

Update for loop current reversal detection on Dialogic® DMV160LP Board

Because of a new feature in the Service Update, loop current reversal detection is now supported on the Dialogic® DMV160LP Board. The *Dialogic® Global Call Analog Technology Guide* does not currently include any information about how to implement this. For information about this feature, see Section 1.48, “Loop Current Reversal Detection on Dialogic® DMV160LP Boards”, on page 170 of this Release Update.

Update for analog call transfer support on Dialogic® Springware Boards

Because of a new feature in the Service Update, blind and supervised analog call transfers are now supported for Dialogic® Springware Boards. The *Dialogic® Global*

Call Analog Technology Guide does not currently include any information about call transfer. For information about this feature, see Section 1.3, "Analog Call Transfer Support on Dialogic® Springware Boards", on page 38 of this Release Update.

Update for call transfer support on Dialogic® DMV160LP Board

Because of a new feature in the Service Update, call transfer is now supported for Dialogic® DM3 Analog technology (Dialogic® DMV160LP Board only). The *Dialogic® Global Call Analog Technology Guide* does not currently include any information about call transfer. For information about this feature, see Section 1.71, "Call Transfer Support on the Dialogic® DMV160LP Board", on page 249 of this Release Update.

3.4.13 Dialogic® Global Call E1/T1 CAS/R2 Technology Guide

Runtime control of single or double hookflash on consultation drop for FXS/LS protocol

Because of a new feature in the Service Update, information about how to send either a single or double hookflash when dropping a supervised transfer consultation call for FXS/LS protocol should be added to the *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*. For information about this feature, see Section 1.2, "Runtime Control of Single or Double Hookflash on Consultation Drop for FXS/LS Protocol", on page 34 of this Release Update.

Dynamically retrieving and modifying selected protocol parameters when using Dialogic® DM3 Boards

Because of a new feature in the Service Update, information about how to retrieve and modify selected protocol parameters when using Dialogic® DM3 Boards should be added to the *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide*. For information about this feature, see Section 1.42, "Dynamically Retrieving and Modifying Selected Protocol Parameters When Using Dialogic® DM3 Boards", on page 113 of this Release Update.

Update to **Section 3.2.1, Call Analysis with DM3 Boards** (IPY00032691)

After the sentence beginning with "After the normal **gc_MakeCall( )** processing finishes..." (top of page 20), add the following sentence:

The order in which GCEV_CONNECTED and GCEV_MEDIADETECTED events are received may vary; refer to the specific protocol in the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for more details.

Update to **Section 4.2, gc_AnswerCall( )**

Because of a new feature in the Service Update, a new define has been added to **gc_AnswerCall( )** called GC_DBL_ANSWER. For information about this feature, see Section 1.28, "Runtime Control of Double Answer for R2MF", on page 94 of this Release Update.

Update to **Section 4.9.2, gc_Extension( ) with Springware Boards**

This section describes functionality that is not supported on Dialogic® Springware Boards. The section should be ignored.

Updates to **Section 4.10, gc_GetCallInfo( )**

Because of a new feature in the Service Update, the **gc_GetCallInfo( )** CATEGORY_DIGIT parameter is now supported on Dialogic® DM3 Boards. **Section 4.10, gc_GetCallInfo( )**, should be updated to indicate this.

Because of a new feature in the Service Update, a new billing type has been added to CALLINFOTYPE called "CHARGE WITH CLEARING FROM INBOUND." In addition, **gc_GetCallInfo(CALLINFOTYPE)** is now supported on Dialogic® DM3 Boards. For information about this feature and the new billing type, see Section 1.27, "Support for Reporting Billing Type", on page 92 of this Release Update.

Update to **Section 4.13.1, Use of the timeout Parameter** (PTR# 29448)

The information about PDK protocols should be changed as follows (the change is in the third bullet):

For PDK protocols, the time-out value used is determined by:

- The **timeout** parameter in the **gc_MakeCall( )** function.
- The **PSL_DefaultMakeCallTimeout** parameter specified in the .cdp file if the **timeout** parameter in the **gc_MakeCall( )** function is 0 and call analysis is not specified.
- The **PSL_CallProgressMaxDialingTime** parameter specified in the .cdp file if the **timeout** parameter in the **gc_MakeCall( )** function is 0, call analysis is specified, and **PSL_DefaultMakeCallTimeout** is less than **PSL_CallProgressMaxDialingTime**.

Note: PDK protocols do not use the outbound number of ringback tones to define the time-out.

Update to **Section 4.13.3, PDK_MAKECALL_BLK** (PTR# 35050)

In Table 8, PDK_MAKECALL_BLK Field Descriptions, the description of the **flags** field should be changed as follows:

Contains a bitmask that controls call analysis and media type detection on a per call basis. The possible values that can be ORed are:

- NO_CALL_PROGRESS - To disable call analysis.
- MEDIA_TYPE_DETECT - To enable media type detection

Examples:

```
/* To enable Media Detection and disable CPA*/
if (disableCPA && enableMediaDetection)
{
    m_pdkMakecallBlk.flags |= (NO_CALL_PROGRESS|MEDIA_TYPE_DETECT);
    m_gcMakecallBlk.cclib = &m_pdkMakecallBlk;
}

/* To disable CPA */
if (disableCPA)
{
    m_pdkMakecallBlk.flags |= NO_CALL_PROGRESS;
    m_gcMakecallBlk.cclib = &m_pdkMakecallBlk;
}

/* To enable Media Detection */
if (enableMediaDetection)
{
    m_pdkMakecallBlk.flags |= MEDIA_TYPE_DETECT;
    m_gcMakecallBlk.cclib = &m_pdkMakecallBlk;
}
```

Update to **Section 4.18, gc_SetChanState( )** (PTR# 36726)

The following note should be added to this section:

Note: When a channel is set to out-of-service state, not all protocols send the blocking pattern by default. For such protocols, a parameter in the .cdp file has to be set to the appropriate value so that the blocking pattern is sent when the channel is put out-of-service. Refer to the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for more information.

Event cause values (PTR# 34490)

The *Dialogic® Global Call E1/T1 CAS/R2 Technology Guide* should include a list of the E1/T1 CAS/R2-specific event cause values that are retrieved by **gc_ResultValue( )** and **gc_ResultInfo( )**. The following information should be added:

Note: This information is applicable to Dialogic® DM3 Boards only.

Call Control Library Cause Values When Using Dialogic® DM3 Boards

Cause Value (Decimal)	Cause Value (Hex)	Description
128	0x80	Requested information available. No more expected.
129	0x81	Requested information available. More expected.
130	0x82	Some of the requested information available. Timeout.
131	0x83	Some of the requested information available. No more expected.
132	0x84	Requested information not available. Timeout.
133	0x85	Requested information not available. No more expected.
134	0x86	Information has been sent successfully.

Note: The cause values in this table are ORed with the value 0x300, which identifies them as call control library cause values.

Firmware-Related Cause Values When Using Dialogic® DM3 Boards

Cause Value (Decimal)	Cause Value (Hex)	Description
01	0x01	Busy
02	0x02	Call Completion
03	0x03	Canceled
04	0x04	Network congestion
05	0x05	Destination busy
06	0x06	Bad destination address
07	0x07	Destination out of order
08	0x08	Destination unreachable
09	0x09	Forward
10	0x0A	Incompatible
11	0x0B	Incoming call
12	0x0C	New call
13	0x0D	No answer from user

Note: The cause values in this table are ORed with the value 0xC0, which identifies them as firmware-related cause values.

Cause Value (Decimal)	Cause Value (Hex)	Description
14	0x0E	Normal clearing
15	0x0F	Network alarm
16	0x10	Pickup
17	0x11	Protocol error
18	0x12	Redirection
19	0x13	Remote termination
20	0x14	Call rejected
21	0x15	Special Information Tone (SIT)
22	0x16	SIT Custom Irregular
23	0x17	SIT No Circuit
24	0x18	SIT Reorder
25	0x19	Transfer
26	0x1A	Unavailable
27	0x1B	Unknown cause
28	0x1C	Unallocated number
29	0x1D	No route
30	0x1E	Number changed
31	0x1F	Destination out of order
32	0x20	Invalid format
33	0x21	Channel unavailable
34	0x22	Channel unacceptable
35	0x23	Channel not implemented
36	0x24	No channel
37	0x25	No response
38	0x26	Facility not subscribed
39	0x27	Facility not implemented
40	0x28	Service not implemented
41	0x29	Barred inbound
42	0x2A	Barred outbound
43	0x2B	Destination incompatible
44	0x2C	Bearer capability unavailable

Note: The cause values in this table are ORed with the value 0xC0, which identifies them as firmware-related cause values.

3.4.14 Dialogic® Global Call IP Technology Guide

Update for IP support on Dialogic® DI0408LSAR2 Boards

Because of a new feature in the Service Update, Voice over IP (VoIP) capability is now supported on Dialogic® DI0408LSAR2 Switching Boards. The *Dialogic® Global Call IP Technology Guide* does not currently include any information about DI0408LSAR2 Boards. For information about this feature, including configuration information, see Section 1.45, "IP Support on Dialogic® DI0408LSAR2 Boards", on page 152 and

Section 1.46, “Dialogic® DI0408LSAR2 Board Support for Host Systems with Multiple NICs”, on page 159 of this Release Update.

Update for early media

Because of a new feature in the Service Update, information about early media should be added to **Chapter 3, IP Call Scenarios**. For information about this feature, see Section 1.52, “Early Media”, on page 213 of this Release Update.

Update for SIP call transfer

Because of a new feature in the Service Update, information about SIP call transfer should be added to **Chapter 4, IP-Specific Operations**. For information about this feature, see Section 1.51, “SIP Call Transfer”, on page 182 of this Release Update.

Update for SIP message header fields (PTR# 35268)

The ability to set and retrieve SIP message header fields is not supported in Dialogic® System Release 6.0 PCI for Windows®. The information in **Section 4.5, Setting and Retrieving SIP Message Information Fields**, and all of its subsections (pages 59-62) should be ignored.

The version of the IP_VIRTBOARD data structure that is implemented in Dialogic® System Release 6.0 PCI for Windows® (structure version 0x100) does not include the sip_msginfo_mask field. The line of the typedef on page 184 that defines this field and the description of the field on page 185 should both be ignored, and applications should not attempt to set the value of this undefined field.

Update for adjusting the Windows® TimedWait state

The following information on adjusting the Windows® TimedWait state should be added to the guide:

Running ONLY call control on 10 or more timeslots may cause the error:

IPEC_Q931Cause34NoCircuitChannelAvailable

Each IP call uses a Windows® socket that binds the call to a unique TCP address/port. The Dialogic® Global Call stack uses these ports starting at port address 20000. When an IP call is completed, the socket associated with that call closes and then enters into a TimedWait state, during which the socket.s associated address/port is not available for use until the time expires. The default time for this TimedWait state is 240 seconds.

If an application is stopped and then immediately restarted before the TimedWait state expires, as may be the case during application development and debugging, calls may fail. Reducing the duration of the TimedWait state can alleviate this problem.

Another problem that may result from the TimedWait state duration is when a server experiences a high call rate. Even though the maximum number of TCP connections that can be opened simultaneously is large, in a high call rate scenario the potential exists for hundreds of TCP sockets to be in the TimedWait state causing the system to reach the maximum number of TCP connections. Again, reducing the duration of the TimedWait state can alleviate this problem.

Changing the TimedWait state to a value less than the 240 second default requires a change to the Windows® registry:

System Key:

HKEY_LOCAL_MACHINE\System\CurrentControlSet\Services\Tcpip\Parameters
(PTR# 32165)

Parameter Name:
TcpTimedWaitDelay

Value Type:
REG_DWORD (DWORD Value)

Valid Range:
30 to 300 seconds

Also, see the following Microsoft information at these links:

- Windows® 2000 *http://support.microsoft.com/default.aspx?scid=kb;en-us;120642*
- Windows® XP *http://support.microsoft.com/default.aspx?scid=kb;en-us;314053*

Update to **Section 7.2.12, gc_MakeCall( ) Variances for IP** (IPY00029956 = PTR# 36646)

The following paragraph is added to **Section 7.2.12, gc_MakeCall( ) Variances for IP**, just before Section 7.2.12.1 (page 123):

When using SIP, if the remote side does not send a final response to an outgoing INVITE (sent by the call control library) within 64 seconds, the **gc_MakeCall( )** function times out and the library generates a GCEV_DISCONNECTED event to the application. If the application attempts to drop the call before the 64 second timeout is reached, the library's behavior depends on whether a provisional response was received. When no provisional response was received before the application cancels the call, the library cleans up the call immediately. But if a provisional response was received before the application attempts to cancel the call, the library sends a CANCEL to the remote endpoint and generates a GCEV_DROPCALL to the application after it receives the 200OK response to the CANCEL and a 487RequestTerminated response for the original INVITE, or when an additional 32-second timeout expires.

Update to the **gc_OpenEx( ) Variances for IP** section (PTR# 32087)

The following paragraph should be added to **Section 7.2.13, gc_OpenEx( ) Variances for IP** (pages 136-137). This paragraph should also be considered to be a variance for **gc_Close( )**, which does not have a “Variances for IP” section in this edition of the document.

- Applications should avoid closing and re-opening devices multiple times. Board devices and channel devices should be opened during initialization and should remain open for the duration of the application.

Update for **INIT_IPCCLIB_START_DATA( )** and **INIT_IP_VIRTBOARD( )**

The initialization functions **INIT_IPCCLIB_START_DATA( )** and **INIT_IP_VIRTBOARD( )**, which are documented in the *Dialogic® Global Call IP Technology Guide*, are not supported in Dialogic® System Release 6.0 PCI for Windows®. Ignore the discussions of the **INIT_IPCCLIB_START_DATA( )** and **INIT_IP_VIRTBOARD( )** functions in the following sections:

- Section 4.1, Call Control Configuration
- Section 4.5.1, Enabling Access to SIP Message Information Fields
- Section 7.2.20, gc_Start( ) Variances for IP
- Section 7.5.1, INIT_IPCCLIB_START_DATA( )
- Section 7.5.2, INIT_IP_VIRTBOARD( )
- IP_VIRTBOARD data structure reference page

- IPCCLIB_START_DATA data structure reference page

In the absence of the **INIT_IPCCLIB_START_DATA( )** and **INIT_IP_VIRTBOARD( )** functions, the user must manually initialize the IP_VIRTBOARD and IPCCLIB_START_DATA data structures before calling **gc_Start( )**. Refer to the reference pages for these structures for more details.

Update for **IP_H221NONSTANDARD** data structure

On the reference page for the IP_H221NONSTANDARD data structure (page 182), the descriptions of the three data fields are updated as follows:

country_code

The country code. Range: 0 to 255; any value x>255 is treated as x%256.

extension

The extension number. Range: 0 to 255; any value x>255 is treated as x%256.

manufacturer_code

The manufacturer code. Range: 0 to 65535; any value x>65535 is treated as x%65636.

3.4.15 Dialogic® Global Call ISDN Technology Guide

Updates for Two B Call Transfer Support (IPY00006590 = PTR# 36501)

Section 1.2, ISDN Features and Benefits, provides incorrect information about Two B Call Transfer (TBCT) and the level of support provided by Dialogic® Global Call Software. The following is the correct information:

TBCT is a National ISDN-2 (NI2) supplementary service described in the Telcordia GR 2865 standard. The feature enables a user to request the switch to connect together two independent calls on the user's interface. The user who made the request is released from the calls and the other two users are directly connected. This feature is supported on 5ESS and DMS switches provisioned to implement NI2; see Section 3.1, "General ISDN Call Scenarios", for details.

Section 3.1.23, Network Facility Request - Two B Channel Transfer (Synchronous Mode) provides the following incorrect statement about TBCT support: "(this feature is supported for the 5ESS and 4ESS protocols only)." The correct statement is: "(this feature is supported for 5ESS and DMS switches implementing the NI2 protocol only)."

Dynamically retrieving and modifying selected protocol parameters when using Dialogic® DM3 Boards

Because of a new feature in the Service Update, information about retrieving a protocol ID should be added in the **ISDN-Specific Operations** chapter. Also, additional subsections should be added under **Using Dynamic Trunk Configuration** for setting the line type and coding for a trunk, and specifying the protocol for a trunk. For information about this feature, see Section 1.42, "Dynamically Retrieving and Modifying Selected Protocol Parameters When Using Dialogic® DM3 Boards", on page 113 of this Release Update. In particular, see Section 1.42.1.2, "Retrieving a Protocol ID", on page 115 and Section 1.42.1.6, "Dynamically Configuring a Trunk", on page 129.

Support for QSIG NCAS on Dialogic® DM3 Boards

Because of a new feature in the Service Update, information regarding support for making and receiving NCAS calls using the QSIG protocol on Dialogic® DM3 Boards should be added in the **ISDN-Specific Operations** chapter. For information about this feature, see Section 1.47, “Support for QSIG NCAS Calls on Dialogic® DM3 Boards”, on page 161 of this Release Update.

Update to **Chapter 2, Global Call Architecture for ISDN** (IPY00006540 = PTR# 34211)
The following information should be appended to the end of Chapter 2:

GCEV_EXTENSION Events

There are ISDN-specific Global Call events, which will eventually be mapped to GCEV_EXTENSION. But to maintain backward compatibility, the Global Call application has the option to choose ISDN-specific events or GCEV_EXTENSION. The default is ISDN-specific events. For more information, refer to Section 4.2, “Operations Performed Using RTCM”.

Note: When using Dialogic® DM3 Boards, the GCEV_EXTENSION event is not supported. DM3 Boards use ISDN-specific events only.

If the application needs to use the new generic call model or extension features, **gc_Start( )** should be called as shown below:

```
CCLIB_START_STRUCT cclib_struct;
GC_START_STRUCT gc_start_struct;
GC_PARM_BLK *parmblk = NULL;

gc_util_insert_parm_val( &parmblk,
                         GCIS_SET_GENERIC,
                         GCIS_PARM_EXTENSIONEVENT,
                         sizeof( char ), 1);

gc_util_insert_parm_val( &parmblk,
                         GCIS_SET_GENERIC,
                         GCIS_PARM_GENERICCALLMODEL,
                         sizeof( char ), 1);
gc_start_struct.num_cclibs = 1;
gc_start_struct.cclib_list = &cclib_struct;
gc_start_struct.cclib_list[0].cclib_name =  "GC_ISDN_LIB";
gc_start_struct.cclib_list[0].cclib_data = parmblk;

if ( gc_Start( &gc_start_struct ) != GC_SUCCESS ) {
    exit(1);
}
gc_util_delete_parm_blk(parmblk);
```

The field extevtdatap of the METAEVENT structure points to EXTENSIONEVT_BLK.

```
typedef struct {
    unsigned char    ext_id;
    GC_PARM_BLK      parmblk;
} EXTENSIONEVTBLK;
```

The following table defines the different possible extension IDs in the GCEV_EXTENSION event.

Event	Description
GCIS_EXEV_NOTIFYGLOBAL when using Dialogic® Springware Boards **Note:** When using Dialogic® DM3 Boards, this event is not supported.	A DROP request has been received; the request was made by sending the SndMsg_Drop message type via the **gc_Extension(GCIS_EXID_SNDMSG)** function. This event has two different meanings that depend upon the type of call: • Two-party call - the event is a request to disconnect the call. The application should respond by issuing a **gc_DropCall( )**. • Conference call - the event is a request to remove the last party that was added to the conference. The application needs to respond to this request with either a SndMsg_DropAck or SndMsg_DropRej message to indicate the acceptance or rejection of the request. If the request is accepted, the party is dropped from the conference. This event only pertains to a Custom BRI 5ESS switch type.
GCIS_EXEV_CONGESTION when using Springware Boards When using DM3 Boards, the equivalent event is GCEV_CONGESTION	A CONGESTION message has been received by the application, indicating that the remote end is not ready to accept incoming user information. Use the **gc_GetCallInfo( )** function to retrieve additional information about the event or look into the extension event data.
GCIS_EXEV_DIVERTED when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	NAM with divert information has been received by the application. An outgoing call has been successfully diverted to another station.
GCIS_EXEV_DROPACK when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	The network has honored a DROP request for a conference call; the request was made by sending the SndMsg_Drop message type via the **gc_Extension(GCIS_EXID_SNDMSG)** function. The event is sent on the corresponding line device. This event pertains only to a Custom BRI 5ESS switch type.
GCIS_EXEV_DROPREJ when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	The network has not honored a DROP request for a conference call. The event is sent on the corresponding line device.This event pertains only to a Custom BRI 5ESS switch type.
GCIS_EXEV_FACILITY when using Springware Boards When using DM3 Boards, the equivalent event is GCEV_FACILITY	A FACILITY REQUEST message has been received by the application.
GCIS_EXEV_FACILITY_ACK when using Springware Boards. **Note:** When using DM3 Boards, this event is not supported.	A FACILITY_ACKNOWLEDGEMENT message has been received by the application.
GCIS_EXEV_FACILITY_REJ when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	A FACILITY_REJECT message has been received by the application.
GCIS_EXEV_FACILITYGLOBAL when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	An ISDN_FACILITY message containing a Global CRN value was received. This event is sent on the board level device, as the event is associated with all calls on the device. Upon receipt of this event, the application may issue a **gc_Extension(GCIS_EXID_GETNONCALLMSG)** function to retrieve the data into its local structure or look into the extension event data.

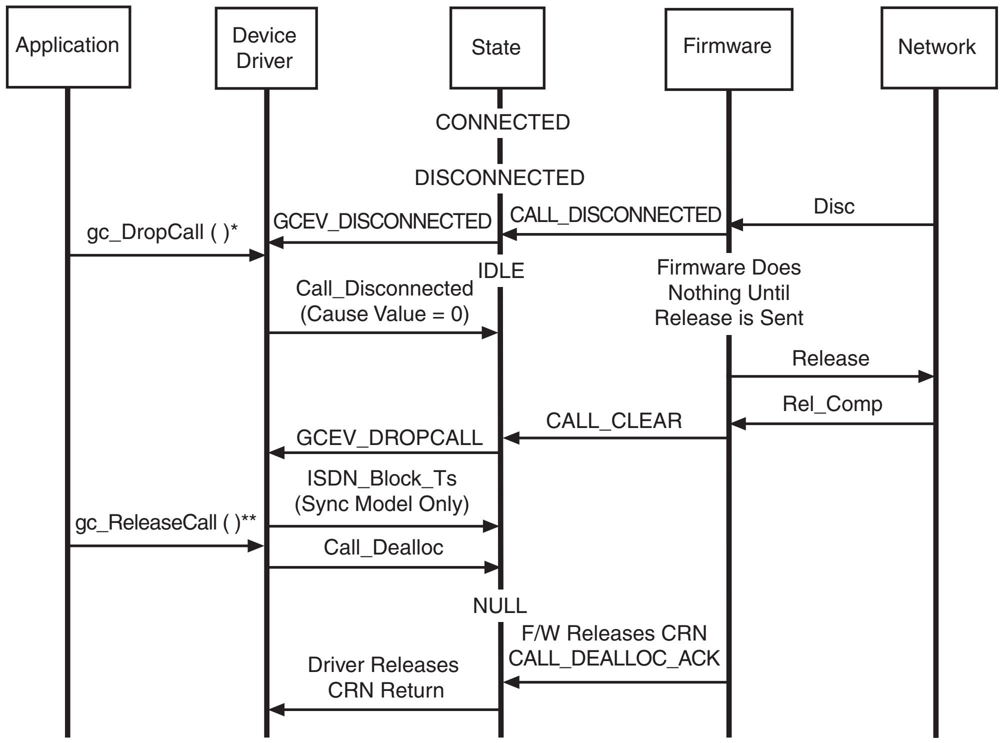
Event	Description
GCIS_EXEV_FACILITYNULL when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	An ISDN_FACILITY message was received containing a Dummy (NULL) CRN. Upon receipt of this event, the application may issue a **gc_Extension(GCIS_EXID_GETNONCALLMSG)** function to retrieve the data into its local structure or look into the extension event data.
GCIS_EXEV_INFOGLOBAL when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	An ISDN_INFORMATION message containing a Global CRN value was received. This event is sent on the board level device, as the event is associated with all calls on the device. Upon receipt of this event, the application may issue a **gc_Extension(GCIS_EXID_GETNONCALLMSG)** function to retrieve the data into its local structure or look into the extension event data.
GCIS_EXEV_INFONULL when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	An ISDN_INFORMATION message was received containing a NULL CRN. Upon receipt of this event, the application may issue a **gc_Extension(GCIS_EXID_GETNONCALLMSG)** function to retrieve the data into its local structure or look into the extension event data.
GCIS_EXEV_L2BFFRFULL when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	Reserved for future use.
GCIS_EXEV_L2FRAME when using Springware Boards When using DM3 Boards, the equivalent event is GCEV_L2FRAME	A data link layer frame has been received by the application. The application should use the **gc_Extension(GCIS_EXID_GETFRAME)** function to retrieve the received frame. It is the application's responsibility to analyze the contents of the frame or look into the extension event data.
GCIS_EXEV_L2NOBFFR when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	There are no buffers available to save the incoming frame.
GCIS_EXEV_NOFACILITYBUF when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	Facility buffer is not ready.
GCIS_EXEV_NOTIFY when using Springware Boards When using DM3 Boards, the equivalent event is GCEV_NOTIFY	A NOTIFY message has been received by the application. Use the **gc_GetCallInfo( )** function to retrieve additional information about the event or look into the extension event data.
GCIS_EXEV_NOTIFYGLOBAL when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	An ISDN_NOTIFY message containing a Global CRN value was received. This event is sent on the board level device, as the event is associated with all calls on the device. Upon receipt of this event, the application may issue a **gc_Extension(GCIS_EXID_GETNONCALLMSG)** function to retrieve the data into its local structure or look into the extension event data.
GCIS_EXEV_NOTIFYNULL when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	An ISDN_NOTIFY message was received containing a Dummy (NULL) CRN. Upon receipt of this event, the application may issue a **gc_Extension(GCIS_EXID_GETNONCALLMSG)** function to retrieve the data into its local structure or look into the extension event data.

Event	Description
GCIS_EXEV_NOUSRINFOBUF when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	User IE buffer is not ready.
GCIS_EXEV_NSI when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	A Network Specific Indication (NSI) message was received from the network. The application should use **gc_GetCallInfo( )** to retrieve the NSI string(s) or look into the extension event data.
GCIS_EXEV_PLAYTONE when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	User-defined tone successfully played.
GCIS_EXEV_PLAYTONEFAIL when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	Request to play user-defined tone failed.
GCIS_EXEV_PROGRESSING when using Springware Boards When using DM3 Boards, the equivalent event is GCEV_PROGRESSING	A PROGRESS message has been received by the application. By default, the firmware will send this event to the application. The application may block this event by clearing the CCMSK_PROGRESS bit. Use the **gc_GetCallInfo( )** function to retrieve additional information about the event or look into the extension event data.
GCIS_EXEV_STATUS when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	A STATUS message has been received from the network.
GCIS_EXEV_STATUS_ENQUIRY when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	A STATUS_ENQ message has been received from the network.
GCIS_EXEV_STOPTONE when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	The tone operation was terminated.
GCIS_EXEV_STOPTONEFAIL when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	The request to terminate the playing of a tone failed.
GCIS_EXEV_TIMER when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	An unsolicited event indicating that a timer has expired.
GCIS_EXEV_TONEREDEFINE when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	The tone(s) in the firmware tone template table were successfully redefined.

Event	Description
GCIS_EXEV_TONEREDEFINEFAIL when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	The request to redefine tone(s) in the firmware tone template table failed.
GCIS_EXEV_TRANSFERACK when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	A TRANSFER ACKNOWLEDGE message was received from the network. The indicated network has accepted a request to transfer a call.
GCIS_EXEV_TRANSFERREJ when using Springware Boards **Note:** When using DM3 Boards, this event is not supported.	A TRANSFER REJECT message was received from the network. The indicated network has rejected a request to transfer a call.
GCIS_EXEV_TRANSIT when using Springware Boards When using DM3 Boards, the equivalent event is GCEV_TRANSIT	After a transfer is established, transit messages are used for relating messages between the originating end and the terminating end.
GCIS_EXEV_USRINFO when using Springware Boards When using DM3 Boards, the equivalent event is GCEV_USRINFO	A USER INFORMATION message has been received by the application, indicating that a user-to-user information (UUI) event is coming. For example, this event is received in response to a **gc_Extension(GCIS_EXID_SNDMSG)** function call, from the far end, in which the msg_type is SndMsg_UsrInformation. Use the **gc_GetCallInfo( )** function to retrieve the UUI or look into the extension event data. Field parmblk of EXTENSIONEVTBLK will hold following parameters: GCIS_SET_IE, GCIS_PARM_UIEDATA (char array, maximum length can go to MAXLEN_IEDATA): Unprocessed IEs in CCITT format. The IEs are returned as raw data and must be parsed and interpreted by the application.

Update to **Section 3.1.18, Simultaneous Disconnect From Any State**

The scenario depicted in Figure 21 is incorrect. The following figure shows the correct scenario:



Notes:
* = Application Should Set a "Drop Call" Flag
** = Application should ignore GCEV _DISCONNECTED if "Drop Call" Flag is Set
***= gc_ReleaseCall ( ) always clears "Drop Call" Flag

Update to **Section 3.1.24, Non-Call Associated Signaling (Synchronous Mode)** (PTR# 32165 and PTR# 35249)

In **Section 3.1.24, Non-Call Associated Signaling (Synchronous Mode)** on page 70, the first two paragraphs provide incorrect information. The following paragraphs provide the correct information:

Non-Call Associated Signaling (NCAS) allows users to communicate by user-to-user signaling without setting up a circuit-switched connection (this signaling does not occupy B channel bandwidth). A temporary signaling connection is established (and cleared) in a manner similar to the control of a circuit-switched connection. The NCAS feature is supported for 4ESS, 5ESS, CTR4, and QSIG protocols.

Since NCAS calls are not associated with any B channel, applications should receive and transmit NCAS calls on the D channel. For T1 interfaces, this is channel 24, that is, dtiB#T24. For E1 interfaces, there is no channel (dtiB#T#) that corresponds to a D channel line device, therefore NCAS calls (identified by the Bearer Capabilities IE content) are automatically associated with the D channel internally on dtiB#T30. Once the NCAS connection is established, the application can transmit user-to-user messages using the CRN associated with the NCAS call. The Dialogic® software and firmware support 16 simultaneous NCAS calls per D channel.

Update to **Section 4.1.7, Retrieve the Network Call Reference Value (CRV)** (PTR# 32418)

In **Section 4.1.7, Retrieve the Network Call Reference Value (CRV)** on page 100, the first note incorrectly references a function called **gc_GetCRV( )**. There is no such function. The correct function name is **gc_GetNetCRV( )**.

Update to **Section 4.4.1, Alarm Handling for DM3 Boards**

In **Section 4.4.1, Alarm Handling for DM3 Boards** beginning on page 137, the following alarms are incorrectly listed as supported alarms:

- DTE1_DCHAN_CFA
- DTE1_DCHAN_CFAOK
- DTT1_DCHAN_CFA
- DTT1_DCHAN_CFAOK

These alarms are not supported and therefore should not be listed.

Update to **Section 8.2.2, gc_AnswerCall( ) Variances for ISDN** (PTR# 35844)

In **Section 8.2.2, gc_AnswerCall( ) Variances for ISDN** beginning on page 178, under the "Springware-specific variances" subheading, the last sentence incorrectly states "The application should restart the timeslot using **gc_ResetLineDev( )** to handle this glare condition." The sentence should read: "The application should restart the timeslot by issuing a **gc_DropCall( )** followed by a **gc_ReleaseCallEx( )** to handle this glare condition."

Update to **Section 8.2.13, gc_GetNetCRV( ) Variances for ISDN** (PTR# 32418)

In **Section 8.2.13, gc_GetNetCRV( ) Variances for ISDN** on page 185, the first note incorrectly references a function called **gc_GetCRV( )**. There is no such function. The correct function name is **gc_GetNetCRV( )**.

Update to **Chapter 11, ISDN-Specific Event Cause Values** (IPY00041046)

The following information should be added to **Table 54, Firmware-Related Cause Values When Using DM3 Boards**:

Cause Value (Decimal)	Cause Value (Hex)	Description
45	0x2D	Normal unspecified
46	0x2E	Req timed out
47	0x2F	Remote retrieve
48	0x30	Remote reconnect
49	0x31	Local timeout
50	0x32	Ack
51	0x33	No ringback
52	0x34	SIT vacant circuit
53	0x35	SIT operator intercept
54	0x36	SIT no circuit interlata
55	0x37	SIT reorder interlata
56	0x38	SIT ineffective other

3.4.16 Dialogic® Global Call SS7 Technology Guide

There are currently no updates to this document.

3.4.17 Dialogic® IP Media Library API Programming Guide

Update to **Section 6.1, Introduction to DTMF Handling** (PTR# 33826)

The fourth paragraph in **Section 6.1, Introduction to DTMF Handling** (page 21) and the note that follows the fourth paragraph should be ignored. The IPM_RFC2833MUTE_AUDIO parameter that the paragraph refers to is not supported; DTMF audio is always muted when in RFC2833 mode. Similarly, Step 5 in the procedure in **Section 6.2.3, Setting RFC 2833 Mode** (page 24) should also be ignored.

Update to **Section 7.4, Using QoS Alarms**

The example code in **Section 7.4, Using QoS Alarms**, is missing the declaration and initialization for the **l_pVoid** variable within the **CheckEvent( )** subroutine on page 35. The code should include the line:

```
void* l_pVoid = sr_getevtdatap();
```

3.4.18 Dialogic® IP Media Library API Library Reference

In the reference information for the **ipm_GetLocalMediaInfo( )** function, the first Caution refers to incorrect defines for eMediaType. The first Caution is replaced with the following paragraph:

- To retrieve RTP or T.38 information, set the eMediaType field to MEDIATYPE_LOCAL_RTP_INFO or MEDIATYPE_LOCAL_UDPTL_T38_INFO and set unCount to 1. See the example for details.

In the reference information for the **ipm_GetLocalMediaInfo( )** function, the following corrections are made in the code example:

The line:

```
  // MediaInfo.MediaData[0].eMediaType = MEDIATYPE_LOCAL_T38_INFO;
```

is replaced with:

```
  // MediaInfo.MediaData[0].eMediaType = MEDIATYPE_LOCAL_UDPTL_T38_INFO;
```

The line:

```
  printf("MediaType = MEDIATYPE_RTP_INFO!!\n");
```

is replaced with:

```
  printf("MediaType = MEDIATYPE_LOCAL_RTP_INFO!!\n");
```

3.4.19 Dialogic® ISDN Software Reference

In the **cc_GetEvtMsk( )** function reference pages, **Table 20**, Bitmask Values, incorrectly indicates the default values for CCMSK_SERVICE_ACK and CCMSK_SETUP_ACK as “Not enabled”. The correct default values are “Enabled.”

In the **cc_GetEvtMsk( )** function reference pages, **Table 20**, Bitmask Values, incorrectly lists CCMSK_TERMINATE as a supported bitmask type. The CCMSK_TERMINATE bitmask type is **not** supported (PTR# 29203).

In the **cc_SetEvtMsk( )** function reference pages, **Table 24**, Bitmask Values, incorrectly indicates the default values for CCMSK_SERVICE_ACK and CCMSK_SETUP_ACK as "Not enabled". The correct default values are "Enabled."

The descriptions of the CCMSK_TMREXPEVENT bitmask in the **cc_GetEvtMsk( )** and **cc_SetEvtMsk( )** functions mention that the CCEV_TIMER event is generated when a Layer 3 timer expires, but there is no description of how to retrieve the Timer ID and Call ID values associated with the CCEV_TIMER event (PTR# 29036). The following text describes how to retrieve these values with the assumption that the CCEV_TIMER event has been enabled:

In the application, define a TIMER_DATA structure as follows:

```
typedef struct _TIMER_DATA {
    unsigned char tbd_1;     // 0
    unsigned long CallId;    // 1 2 3 4
    unsigned short TimerId;  // 5 6
    unsigned short tbd_2;    // 7 8
}TIMER_DATA, *PTIMER_DATA;
```

Then, retrieve the values as follows:

```
(evtdatap = sr_getevtdatap(...)
  case CCEV_TIMER:
    { PTIMER_DATA pData = (PTIMER_DATA)evtdatap;
      m_TimerCallId = pData->CallId;
      m_TimerId     = pData->TimerId;
      Log(MSG_EVENT,"Timer: Call_id = %d, Timer expired ID = (%d) 0x%x",
          m_TimerCallId, m_TimerId);
    }
   break;
   .
   .
   .
```

The following caution should be included in the **cc_MakeCall( )** and **cc_SetCallingNumber( )** function reference pages (PTR# 28720):

- When using **cc_MakeCall**( ) to make an outbound call, if the origination_phone_number field in the MAKECALL_BLK structure is set to NULL or '\0' (null string), the destination_number_plan and the destination_number_type fields in the MAKECALL_BLK structure are ignored. This precludes the option of using the **cc_SetCallingNumber**( ) function to set the origination phone number and specifying a value of NULL or '\0' for the origination_phone_number field in the MAKECALL_BLK structure, when the destination number plan and the destination number type values (as specified in the destination_number_plan and destination_number_type fields in the MAKECALL_BLK structure) must be included in the outgoing message.

In the reference information for the **cc_GetDLinkState( )** function, the description paragraph is replaced with the following:

The **cc_GetDLinkState**( ) function retrieves the logical data link state (operable or inoperable) of the specified board device for PRI or station device for BRI.

In the description of the **state_buf** parameter for the **cc_GetDLinkState( )** function, only two possible data link states are defined: DATA_LINK_UP and DATA_LINK_DOWN. DATA_LINK_DISABLED is not a valid value (PTR# 25745).

In the **cc_MakeCall( )** function reference information, the description of the **numberstr** parameter is replaced by the following (PTR# 22842):

The destination (called party's) telephone number string. The maximum number of digits is dictated by the protocol switch specification. Users need to find out the specification limits for the protocol they wish to use, otherwise the protocol stack will reject the request to make a call.

3.4.20 Dialogic® Modular Station Interface API Library Reference

Update to **ms_SendData( )**

The following information is added to the description of the **ms_SendData( )** function:

Make sure an interval of at least 2 seconds elapses between the reception of an MSEV_RING event with MSMM_RNGOFFHK event data and a call to the **ms_SendData**( ) function to ensure reliable reception of call waiting caller ID. If there is still a problem receiving the call waiting caller ID, it may be due to the configuration of the phone.

Update to MS_CDT data structure (PTR# 35565)

A note should be added to the MS_CDT chan_sel field indicating that MSPN_STATION is supported on Dialogic® Springware Boards only and MSPN_TS is supported on Dialogic® DI, HDSI, and Springware Boards.

Update for new event

Because of a new feature in the Service Update, information about the MSEV_CHANSTATE event should be added to the descriptions of the **ms_setevtmsk( )** and **ms_getevtmsk( )** functions and to the Events chapter. For information about this feature, see Section 1.66, "New Station Interface Alarms", on page 239 of this Release Update.

3.4.21 Dialogic® Modular Station Interface API Programming Guide

Because of a new feature in the Service Update, information about the MSEV_CHANSTATE event should be added to the Event Handling chapter. For information about this feature, see Section 1.66, "New Station Interface Alarms", on page 239 of this Release Update.

3.4.22 Dialogic® PBX Integration Board User's Guide

PBX Integration Support for Nortel BCM

Because of enhancements introduced in the Service Update, the Nortel Business Communications Manager (BCM) is now supported when using the Dialogic® D/82JCT-U Board. For information about this feature, including programming requirements, see Section 1.40, “PBX Integration Support for Nortel BCM”, on page 111 of this Release Update.

Updates for implementation of ROLM Call Waiting LED

Because of enhancements introduced in the Service Update, the **d42_indicators( )** function can now return the LED status of the Call Waiting LED for the ROLMphone 400. **Table 8, ROLMphone 400 Direct Key Dialing Strings for Feature Keys** on pages 44 and 45, and the figure on page 46 should be updated to reflect this. For the correct table and figure and for further information about this feature, see Section 1.58, “Implementation of ROLM Call Waiting LED”, on page 227 of this Release Update.

Update to **Section 4.2.1, Siemens ROLM Programming Requirements** (IPY00006024 = PTR# 29612)

An additional note should be added to the NOTES at the end of this section:

With all switches supporting ROLM phone 400, the asynchronous event TD42_ASYNCCHSTATUS for reporting carrier gain is only received once, when the board starts and the port is connected to the switch. If the port is disconnected and connected again, the application does not receive any other carrier loss and gain events.

Update to **Section 4.2.5**, **Setting the Message Waiting Indicator**, for Siemens ROLM PBX

In **Section 4.2.5**, **Setting the Message Waiting Indicator**, the last paragraph of the section on page 50 states: “The PBX integration board can determine the state of its Message Waiting Indicator using the **d42_indicators( )** function to retrieve the LED Indicators data. Byte 40 contains the Message Waiting indicator status (0x00 is off; 0x01 is on). Refer to the *Dialogic® PBX Integration Board Software Reference* for more information about using the **d42_indicators( )** function.” The paragraph is followed by an Example also on page 50. The functionality described is not supported for Siemens ROLM systems and therefore the entire paragraph and the example should not be included.

Update to **Section 4.5.1, Nortel Norstar Programming Requirements** (IPY00006258 = PTR# 36353)

Under the sub-heading **Nortel Norstar Programming Requirements for MICS and CICS** beginning on page 99, the following information should be included:

Nortel MICS switches typically support connections for up to 16 voicemail ports. If more than 16 voicemail ports are connected simultaneously, the resulting traffic may overload the switch resulting in symptoms such as very slow switch response, lost LED updates, incomplete display updates and lost calls.

In order to use Dialogic® D/42JCT-U or D/82JCT-U Boards with the Nortel MICS switch, “Nortel_Norstar.fwl” is selected in the Dialogic® Configuration Manager (DCM). This firmware boots all of the board ports into voicemail mode. Therefore, it is strongly recommended that the programming of any MICS system be carefully reviewed to ensure that no more than 16 ports are connected in voicemail mode.

Note: It is possible to use D/42JCT-U or D48/JCT-U Boards with the Nortel MICS switch without the connected ports operating in voicemail mode. To do this, load “Nortel_BCM.FWL” in DCM. This firmware does not boot the ports in voicemail mode. Keep in mind that in this configuration, these ports have blank CPID information and they are not able to set or clear Message Waiting indicators.

Update to **Section 4.6.5**, **Setting the Message Waiting Indicator**, for Nortel Meridian 1

In **Section 4.6.5**, **Setting the Message Waiting Indicator**, the last paragraph (on page 125) should be ignored. The ability to determine the state of the Message Waiting Indicator is not supported for Nortel Meridian 1 systems.

3.4.23 Dialogic® PBX Integration Software Reference

Updates for implementation of ROLM Call Waiting LED

Because of enhancements introduced in the Service Update, the **d42_indicators( )** function can now return the LED status of the Call Waiting LED for the ROLMphone 400.

On the **d42_indicators( )** function reference page (page 44), in the Description subsection, the list showing the number of indicators on each PBX type has an incorrect value for Siemens/ROLM. The value shown is 35; the correct value is now 36. In addition, the figure at the top of page 50 should be replaced by the figure in Section 1.58, "Implementation of ROLM Call Waiting LED", on page 227 of this Release Update.

3.4.24 Dialogic® Standard Runtime Library API Library Reference

There are currently no updates to this document.

3.4.25 Dialogic® Standard Runtime Library API Programming Guide

There are currently no updates to this document.

3.4.26 Dialogic® Voice API Library Reference

New function

The **dx_resetch( )** function is now supported in Dialogic® System Release 6.0 PCI for Windows®. The **dx_resetch( )** function recovers a channel that is "stuck" (busy or hung) and in a recoverable state, and brings it to an idle and usable state. For further information, see Section 1.5, "Dialogic® DM3 Media Channel Reset Capability (Stuck Channel Recovery)", on page 45 of this Release Update.

Functions not supported

The **r2_creatfsig( )** and **r2_playbsig( )** functions, which were previously provided for backward compatibility only, are no longer supported. All references to these functions should be deleted. R2MF signaling is typically accomplished through the Dialogic® Global Call API.

Update to **ATDX_CRTNID( )** function

Because of enhancements introduced in the Service Update, the **ATDX_CRTNID( )** function is now supported on Dialogic® DM3 Boards with new tone IDs. For information about this feature, see Section 1.8, "Enhanced Special Information Tones on Dialogic® DM3 Boards Using Voice and Global Call APIs", on page 55 and

Section 1.59, “Enhanced Special Information Tone Frequency Detection on Dialogic® DM3 Boards”, on page 229 of this Release Update.

Update to **dx_createtone( )** function

Because of a new feature in the Service Update, the **dx_createtone( )** function can be used with the new custom special information tones (SITs) described in Section 1.8, “Enhanced Special Information Tones on Dialogic® DM3 Boards Using Voice and Global Call APIs”, on page 55 of this Release Update.

Update to **dx_deletetone( )** function

Because of a new feature in the Service Update, the **dx_deletetone( )** function can be used with the new custom special information tones (SITs) described in Section 1.8, “Enhanced Special Information Tones on Dialogic® DM3 Boards Using Voice and Global Call APIs”, on page 55 of this Release Update.

Update to **dx_dial( )** function (PTR# 36660)

The following information should be added to the **dx_dial( )** function, **dialstrp** parameter description:

The maximum dial string size (number of digits) is 275 for Dialogic® DM3 Boards and 200 for Dialogic® Springware Boards.

Update to **dx_getdig( )** function (IPY00038453)

For Dialogic® DM3 Boards, the return value of **dx_getdig( )** in synchronous mode has been changed to return 0 instead of 1 when there are no digits in the buffer. The NULL character in the digit string ‘dg_value’ is no longer counted as a digit. Similarly, when **dx_getdig( )** returns the number of digits, the terminating NULL is no longer added to the number of digits. (The NULL was previously counted in the numdig return value calculation, but since it is not a digit, the NULL is no longer included.)

For Dialogic® Springware Boards, the terminating NULL **is** included in the number of digits. So for Springware Boards, **dx_getdig( )** still returns 1 when there are no digits in the buffer.

Update to **dx_querytone( )** function

Because of a new feature in the Service Update, the **dx_querytone( )** function can be used with the new custom special information tones (SITs) described in Section 1.8, “Enhanced Special Information Tones on Dialogic® DM3 Boards Using Voice and Global Call APIs”, on page 55 of this Release Update.

Update to **dx_setevtmsk( )** (IPY00038053)

The following information should be added to the description of the **mask** parameter:

User defined tones that are associated an optional digit (**dx_addtone( )**) have digit reporting enabled by default in Dialogic® System Release 6.0 PCI for Windows®. The user defined tones digit reporting can be turned off by using **dx_setevtmsk( )** with DM_DIGOFF mask. To reactivate digit reporting, use **dx_setevtmsk( )** with DM_DIGITS mask.

Updates to **Events** chapter

In the Call Status Transition Events section, the DE_DIGOFF event (Dialogic® DM3 Boards and Dialogic® Springware Boards) is not supported and should be removed from the documentation (IPY00033772).

With the Service Update, a time stamp has been added to the DE_TONEON and DE_TONEOFF events for Dialogic® DM3 Boards (supported on Dialogic®

DM/V2400A Board). For further information, see Section 1.32, “Time Stamp for Tone-On/Off Events”, on page 97 of this Release Update.

New TN_TIMESTAMP data structure

Because of a new feature in the Service Update, a new data structure, TN_TIMESTAMP, has been added to provide a time stamp for tone-on/off events. For further information, see Section 1.32, “Time Stamp for Tone-On/Off Events”, on page 97 of this Release Update.

3.4.27 Dialogic® Voice API Programming Guide

Functions not supported

The **r2_creatfsig( )** and **r2_playbsig( )** functions, which were previously provided for backward compatibility only, are no longer supported. All references to these functions should be deleted. R2MF signaling is typically accomplished through the Dialogic® Global Call API.

Update to **Chapter 6, Application Development Guidelines**

The following note should be added to **Section 6.4.2, Multithreading and Multiprocessing**:

Note: The continuous speech processing architecture allows a voice channel to be shared between processes (or applications) on Dialogic® JCT Boards, on Dialogic® DM3 Boards, and on Dialogic® Host Media Processing (HMP) Software (starting with Dialogic® Host Media Processing Software Release 1.3 for Windows®), providing one process does the play activity and the other process does the record/stream activity. Other CSP scenarios are not supported, such as playing or recording/streaming from both processes.

Update to **Chapter 7, Call Progress Analysis**

Because of enhancements introduced in the Service Update, **Section 7.5.6, SIT Frequency Detection**, is superseded by the information in this Release Update. For information about this feature, see Section 1.8, “Enhanced Special Information Tones on Dialogic® DM3 Boards Using Voice and Global Call APIs”, on page 55 and Section 1.59, “Enhanced Special Information Tone Frequency Detection on Dialogic® DM3 Boards”, on page 229 of this Release Update.

Updates to **Section 8.5, Voice Encoding Methods**

The following row is added to **Table 9, Voice Encoding Methods (DM3 Boards)** (PTR# 31773):

Digitizing Method	Sampling Rate (kHz)	Resolution (bits)	Bit Rate (kbps)	File Format
IMA ADPCM coder	8	4	32	VOX, WAVE

In **Table 10, Voice Encoding Methods (Springware Boards)**, the 16-bit linear PCM coder is not supported on Dialogic® Springware Boards and should be removed from the table. (IPY00006594 = PTR# 36685)

Update to **Section 8.7, Transaction Record** (IPY00006537 = PTR# 35666)

The following paragraph should be added to this section:

For information on running transaction record on a single board, see the technical note posted on the Dialogic® web site at:
http://www.dialogic.com/support/helpweb/dxall/tnotes/legacy/dlsoft/tn253.htm

Update to **Section 10.6, Fixed-Line Short Message Service (SMS)**

This section erroneously states that SMS is not supported on Dialogic® Springware Boards. SMS **is** supported on Springware Boards.

Fixed-line SMS solutions can be created using the standard Telcordia Technologies (formerly Bellcore) ADSI specification or using the ETSI-FSK specification ETSI ES 201 912. On Springware Boards, to set the voice channel to ETSI compatibility, specify the two-way FSK transmit framing parameters in the *voice.prm* file. For more information on these parameters, see the *Dialogic® Springware Architecture Products on Windows® Configuration Guide*.

Update to **Section 10.7.2, Library Support on Springware Boards**

Fixed-line short message service (SMS) is supported on Dialogic® Springware Boards. The information in this section should be updated as follows:

Dialogic® Springware Boards support ADSI one-way, two-way FSK, and fixed-line short message service (SMS).

The following voice library functions and data structures support this functionality on Springware Boards:

dx_RxIottdata( ) function
Receives ADSI data on a specified channel.

dx_TxIottdata( ) function
Transmits ADSI data on a specified channel.

dx_TxRxIottdata( ) function
Starts a transmit-initiated reception of data (two-way ADSI) on a specified channel.

ADSI_XFERSTRUC data structure
Stores information for the transmission and reception of ADSI data. It is used by the **dx_RxIottdata( )**, **dx_TxIottdata( )**, and **dx_TxRxIottdata( )** functions.

DV_TPT data structure
Specifies a termination condition for an I/O function; in this case, **dx_RxIottdata( )**, **dx_TxIottdata( )**, or **dx_TxRxIottdata( )**. DX_MAXDATA termination condition is not supported on Springware Boards.

ATDX_TERMMSK( ) function
Returns the reason for the last I/O function termination. TM_MAXDATA is not supported on Springware Boards.

To determine whether your board supports FSK, use **dx_getfeaturelist( )** to return information about the features supported in the FEATURE_TABLE structure; the ft_play field, FT_ADSI bit, is used to indicate FSK support on Springware Boards.

Update to **Section 13.1.5, Retrieving Tone Events** (PTR# 32681)

The following should be added as the last paragraph of this section:

Cadence tone on events are reported differently on Dialogic® DM3 Boards versus Dialogic® Springware Boards. On DM3 Boards, if a cadence tone occurs continuously, a DE_TONEON event is reported each time a match is detected. On Springware Boards, only the first DE_TONEON event is reported. On both types of boards a DE_TONEOFF event is reported when the tone is no longer present.

Update to **Section 13.1.9, Guidelines for Creating User-Defined Tones** (IPY00006580 = PTR# 34546)

The following guideline should be added to this section:

- On Dialogic® DM3 Boards, building and adding tones of zero frequency values to a tone template can cause firmware failures.

Update to **Section 13.1.10.2, Detecting Leading Edge Debounce Time** (IPY00006581 = PTR# 35616)

The values currently listed in this section apply to Dialogic® DM3 Boards. The following text should be added for Dialogic® Springware Boards:

On Dialogic® Springware Boards, to detect leading edge debounce time, specify the following values for the **dx_bldstcad( )** or **dx_blddtcad( )** function parameters listed below:

- For **ontime**, specify the desired debounce time.
- For **ontdev**, specify 3.
- For **offtime**, specify 0.
- For **offtdev**, specify 0.
- For **repcnt**, specify 1.

Update to **Section 14.3, Enabling Global DPD**

Because of a new feature in the Service Update, it is no longer necessary to order a separate GDPD enablement package to enable Global Dial Pulse Detection on a board. Information about the GDPD enablement package should be removed from this section. See Section 1.7, "Global DPD Enabled on Dialogic® Springware Boards", on page 54of this Release Update for further information.

Update to **Chapter 17, Building Applications** (PTR# 32966)

Run-time linking using the source code in the CLIB subdirectory is no longer supported. Run-time linking can be accomplished using Windows® functions. In the *Dialogic® Voice API Programming Guide*, **Section 17.2.3, Run-time Linking**, should be revised as follows:

Run-time linking resolves the entry points to the Dialogic® DLLs when the application is loaded and executed. This allows the application to contain function calls that are not contained in the DLL that resides on the target system.

To use run-time linking, the application can call the Windows® **LoadLibrary( )** function to load a specific technology DLL and a series of **GetProcAddress( )** function calls to set up the address pointers for the functions.

3.5 Demonstration Software Documentation

This section contains updates to the following documents (click the title to jump to the corresponding section):

- Dialogic® IP Multicast Client (IPML) Demo Guide
- Dialogic® IP Multicast Server (IPML) Demo Guide

3.5.1 Dialogic® IP Multicast Client (IPML) Demo Guide

The following note is added to **Section 2.1, Hardware Requirements** (PTR# 31488):

Note: When using a single span Dialogic® DM/IP Board, the demo supports only one board in the system.

3.5.2 Dialogic® IP Multicast Server (IPML) Demo Guide

The following note is added to **Section 2.1, Hardware Requirements** (PTR# 31488):

Note: When using a single span Dialogic® DM/IP Board, the demo supports only one board in the system.