Dialogic® Global Call E1/T1 CAS/R2

Technology Guide

October 2008

05-2445-003

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Publication Date: October 2008

Document Number: 05-2445-003

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Revision History

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

Document No.	Publication Date	Description of Revisions
05-2445-003	October 2008	Made global changes to reflect Dialogic brand. E1/T1 CAS/R2-Specific Operations chapter: Added Run-Time Control of Single or Double Hookflash on Consultation Drop for FXS/LS Protocol. Added Retrieving Line Signaling Access. E1/T1 CAS/R2-Specific Function Information chapter: Under gc_AnswerCall( ) Variances for E1/T1 CAS/R2, added Run-Time Control of Double Answer. Under gc_GetCallInfo( ) Variances for E1/T1 CAS/R2, added more information about Use of the CALLINFOTYPE info_id Parameter.
05-2445-002	April 2006	Global Call Architecture for E1/T1 CAS/R2 chapter: Clarified which call control libraries are applicable to Dialogic® DM3 Boards and to Dialogic® Springware Boards. E1/T1 CAS/R2-Specific Operations chapter: In Dynamic Trunk Configuration section, added note clarifying the use of **gc_ResetLineDev( )** before **gc_SetConfigData( )** when performing dynamic trunk configuration. Updated Alarm Handling for DM3 Boards section to more accurately specify the alarms that can be transmitted to the remote side and provide a mapping to the 0x1626 parameter in the CONFIG file, which is used for trunk preconditioning. Updated Alarm Handling for Springware Boards section to more accurately specify the alarms that can be transmitted to the remote side. E1/T1 CAS/R2-Specific Function Information chapter: In gc_SetChanState( ) Variances for E1/T1 CAS/R2 section, added note about protocols that do not send the blocking pattern by default (PTR 36726). In gc_SetupTransfer( ) Variances for E1/T1 CAS/R2 section, corrected the name of the **gc_SetupTransfer( )** function; the "u" is lower case, not upper case (PTR 35811).
05-2445-001	July 2005	Initial version of document. Much of the information contained in this document was previously published in the *Global Call E1/T1 CAS/R2 Technology User's Guide for Linux and Windows*, document number 05-0615-011.

About This Publication

The following topics provide information about this publication:

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

Purpose

This guide is for users of the Dialogic® Global Call API who choose to write applications using E1/T1 CAS/R2 technology. This guide provides Global Call E1/T1 CAS/R2 specific information only and should be used in conjunction with the *Dialogic® Global Call API Programming Guide* and the *Dialogic® Global Call API Library Reference*, which describe the generic behavior of the Dialogic® Global Call API.

Applicability

This document version is applicable to Dialogic® Host Media Processing (HMP) Software and to Dialogic® System Release Software for Linux and Windows® operating systems.

Check the Release Guide for your software release to determine whether this document is supported.

Intended Audience

This guide is intended for:

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

This publication assumes that the audience is familiar with the Windows® and Linux operating systems and has experience using the C programming language.

How to Use This Publication

This guide is divided into the following chapters:

- Chapter 1, "E1/T1 CAS/R2 Overview" gives a brief introduction to E1/T1 CAS/R2 concepts for novice users.
- Chapter 2, "Dialogic® Global Call Architecture for E1/T1 CAS/R2" provides an overview of the Dialogic® Global Call Software architecture when using E1/T1 CAS/R2 technology.
- Chapter 3, "E1/T1 CAS/R2 Call Scenarios" discusses call scenarios.
- Chapter 4, "E1/T1 CAS/R2-Specific Operations" describes how one can use the Dialogic® Global Call API to perform E1/T1 CAS/R2 specific operations, such call progress and call analysis, resource association, and others.
- Chapter 5, "E1/T1 CAS/R2 Protocols" describes the protocol conventions used and programming considerations if incorporating individual country protocol(s) into an application. (More detailed information about each protocol appears in the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*.)
- Chapter 6, "Building Dialogic® Global Call E1/T1 CAS/R2 Applications" describes the E1/T1 CAS/R2 specific header files and libraries required if building applications.
- Chapter 7, "Debugging Dialogic® Global Call E1/T1 CAS/R2 Applications" provides information for debugging Dialogic® Global Call applications that use E1/T1 CAS/R2 technology.
- Chapter 8, "E1/T1 CAS/R2-Specific Function Information" describes the additional functionality of specific Dialogic® Global Call API functions used with E1/T1 CAS/R2 technology.
- Chapter 9, "E1/T1 CAS/R2-Specific Data Structures" lists data structures specific to E1/T1 CAS/R2 technology.
- Chapter 10, "E1/T1 CAS/R2-Specific Event Cause Values" lists the supported E1/T1 CAS/R2-specific event cause values, and provides a description of each value.
- Chapter 11, "Supplementary Reference Information" lists references to publications about E1/T1 CAS/R2 technology.

Related Information

See the following for additional information:

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

E1/T1 CAS/R2 Overview 1

This chapter provides overview information about the following topics:

- Making Telephone Calls: Transmission of Digits and Signaling Information 13
- Making Long Distance and Global Telephone Calls 15
- T1 Robbed Bit Signaling Concepts 15
- E1 CAS Signaling Concepts 16
- R2MF Signaling Concepts. 16
- Direct Dialing In (DDI) Service 20

1.1 Making Telephone Calls: Transmission of Digits and Signaling Information

Historically, making a telephone call started with taking your telephone handset out of its cradle. This action caused your telephone to go off-hook. For analog telephones, going off-hook closes a circuit (called the local loop) connected to the local Central Office (CO) and causes a loop current to flow through the local loop circuit created.

The CO reacts by generating dial tone (typically, a combination of 350 Hz and 440 Hz tones), which indicates that you can dial. Traditionally, you would dial your number using pulse dialing (also called rotary dialing). Pulse dialing sends digit information to the CO by momentarily opening and closing (or breaking) the local loop from the calling party to the CO. This local loop is broken once for the digit 1, twice for 2, etc., and 10 times for the digit 0. As each number is dialed, the loop current is switched on and off, resulting in a number of pulses being sent to your local CO.

Alternatively, you may dial a number using tone dialing, wherein sounds represent the digits dialed (0 through 9, # and * are dialing digits). Each digit is assigned a unique pair of frequencies called Dual Tone Multi Frequency (DTMF) digits (see Table 1). Although DTMF signaling is designed for operation on international networks with 15 multifrequency combinations in each direction, in national networks it can be used with a reduced number of signaling frequencies (for example, 10 multifrequency combinations).

In addition to the DTMF digit standard, telcos also use a Multi Frequency (MF) digit standard (see Table 1). MF digits are typically used for CO-to-CO signaling. The MF digit standard is similar to the DTMF digit standard except that different pairs of frequencies are assigned. Some MF digits use approximately the same frequencies as DTMF digits; for example, the digit 4 uses 770 and 1209 Hz for DTMF transmissions or 700 and 1300 Hz for MF transmissions. Because of this frequency overlap, MF digits could be mistaken for DTMF digits if the incorrect tone detection is enabled.

The accuracy of digit detection depends on:

- the digit sent
- the type of detection, MF or DTMF, enabled when the digit is detected. See the *Dialogic® Voice API Library Reference* for details.

Table 1. Signaling Used to Dial (Hz)

Code	Pulse	DTMF	MF	R2MF Forward	R2MF Backward
1	1	697, 1209	700, 900	1380, 1500	1140, 1020
2	2	697, 1336	700, 1100	1380, 1620	1140, 900
3	3	697, 1477	900, 1100	1500, 1620	1020, 900
4	4	770, 1209	700, 1300	1380, 1740	1140, 780
5	5	770, 1336	900, 1300	1500, 1740	1020, 780
6	6	770, 1477	1100, 1300	1620, 1740	900, 780
7	7	852, 1209	700, 1500	1380, 1860	1140, 660
8	8	852, 1336	900, 1500	1500, 1860	1020, 660
9	9	852, 1477	1100, 1500	1620, 1860	900, 660
0	10	941, 1336	1300, 1500	1740, 1860	780, 660
*	-	941, 1209	1100, 1700	1380, 1980	1140, 540
#	-	941, 1477	1500, 1700	1500, 1980	1020, 540

For each call, signaling information (off-hook, number dialed) must be detected by the local CO and then sent to each successive CO until the destination CO is reached. The destination CO attempts to connect to the called party. Concurrently, the destination CO sends back signaling information (such as line busy, network busy signals, etc.) representing the condition or status of the called party's line. This signaling information passes through the network as audio tones or as signaling bits. The number of tones used and the frequency combinations used to convey this signaling information vary from country to country and from telco to telco. In addition, private networks may combine various signaling techniques.

After dialing, you listen to hear the progress and status of the call:

- Ringing tones (ringback) indicate that ring voltage has been applied to the called party's line.
- A busy tone is heard when the called party's telephone is off-hook.
- A fast busy tone may be heard if the telephone network is busy.
- An operator intercept signal is heard if an invalid number is dialed. The operator intercept signal is three rising tones followed by a recording.

Note: No ringing tones are heard when connected to some telcos.

The CO typically indicates the progress of making a call by generating these various tones. When making long distance calls, the telco may make brief drops in loop current to indicate:

- an acknowledgment that the distant CO was reached
- that the calling party's line went off-hook

After a call is connected, a telco service may be requested by a flash-hook. A flash-hook puts the telephone on-hook briefly, long enough for the CO to detect the flash-hook, but not long enough to cause a disconnect. A flash-hook may signal a request for a second dial tone to allow 3-way conferencing or to transfer the call.

At the completion of the call, one or both parties hang up the telephone. Typically, the CO sends a disconnect signal. However, some telcos don't send a disconnect signal; therefore a local CO must use other methods to detect a remote disconnect.

1.2 Making Long Distance and Global Telephone Calls

Long distance calls may involve transmitting dialing and other signaling information from the local CO, through several intermediate COs, to the distant called party's CO and then connecting to the called party. A mixture of signaling systems and protocols may be encountered, especially when making global calls. Local call signaling must be translated into signaling that may pass over analog lines, T1 digital trunks, E1 digital trunks, optical fiber, satellite links, etc. All signaling sent over digital trunks must be converted to bits that can be transmitted or multiplexed with the digitized voice transmissions.

Each telco, country, or region tends to apply different signaling standards that must be observed to ensure that a call gets switched through to the called party. For example, some telcos may encode E&M (Ear and Mouth) signals onto the voice path using a single frequency (SF) tone. When present, this tone indicates an on-hook condition. Otherwise, the line is considered to be off-hook (absence of tone). Typically, when the same manufacturer's product is connected to both ends of a digital trunk, then the signaling technique used is transparent as long as all signaling is handled.

1.3 T1 Robbed Bit Signaling Concepts

A T1 trunk operates at 1.544 Mbps divided into 24 time slots with each time slot operating at 64 kbps [digital signal level 1 (DS-1) rate]. A single 8-bit sample from each of 24 voice channels comprises a D4 frame of 24 time slots on a T1 trunk. Twelve D4 frames make up a D4 superframe.

Signaling information is carried on a T1 trunk by two signaling bits, an A-bit and a B-bit. Each time slot in the sixth frame of a D4 superframe has the least significant bit replaced with A-bit signaling information. Likewise, each time slot in the twelfth frame of the D4 superframe has the least significant bit replaced with B-bit signaling information. This method of replacing the least significant bit with signaling information is called robbed bit signaling. Thus, a T1 robbed bit trunk carries all signaling within the voice time slot (channel) itself.

Dialing, if not done using DTMF or MF tones, is accomplished by alternating the A and B signaling bits between 0 and 1 to mimic rotary dial pulses. Signaling bits represent the state of the M lead on the E&M interface of the calling party. When the called party answers, the M lead returns continuous 1s. When a party hangs up, their signal bits revert to 0s to indicate on-hook. Some telcos invert these signaling bits so that 0 = off-hook and 1 = on-hook.

New telco services may require the use of more than the four signaling states provided by the A and B bits. An extended superframe (ESF) adopted by AT&T provides two additional signaling bits, the C-bit in frame 18 and the D-bit in frame 24.

1.4 E1 CAS Signaling Concepts

An E1 digital trunk operates at 2.048 Mbps divided into 32 time slots with each time slot operating at 64 kbps. These 32 time slots include:

- 30 time slots available for up to 30 voice calls
- one time slot dedicated to carrying frame synchronization information (time slot 0)
- one time slot dedicated to carrying signaling information (time slot 16)

With this method of signaling, each traffic channel has a dedicated signaling channel, that is, channel associated signaling (CAS). The signaling for a particular traffic circuit is permanently associated with that circuit. For E1 CAS, the signaling channel for each traffic channel is located in time slot 16, which is multiplexed between all 30 traffic channels.

E1 CAS service is available in Europe, Africa, Australia, and in parts of Asia and South America. The Conference des Administrations Europeenes des Postes et Telecommunications (CEPT) defines how a PCM carrier system in E1 areas will be used. In addition, the E1 CAS service may carry national and international signaling bits set in time slot 0:

- The international bit occupies the most significant bit (bit position 7) in time slot 0 of each frame.
- The national bits occupy bit positions 0 through 4 of time slot 0 of every second frame.

For each E1 CAS call, signaling information is sent to the local CO and then to each successive CO until the destination CO is reached. The destination CO attempts to connect to the called party. Concurrently, the destination CO sends back signaling information representing the condition or status of the called party’s line. This signaling information passes through the network as audio tones. R2MF signaling is the international standard for conveying call status using these audio tones. However, the number of tones used, the frequency combinations used, and the adherence to the R2 standard can vary from country to country.

Also, whenever a call is switched via networks or protocols that do not support full R2MF signaling, call information may be lost. Although many protocols do not require call analysis because the called party condition is received via R2 tones, when operating in environments where call information may be lost, call progress tones (busy, ringback, SIT tones, etc.) may be useful in determining the condition of a call.

1.5 R2MF Signaling Concepts

R2MF signaling is an international signaling system on E1 that transmits numeric and other information relating to the called and the calling subscribers’ lines. R2MF signals that control the call setup are referred to as interregister signals.

For each call, whether an inbound or an outbound call, the entity making the call is the "calling party" and the entity receiving the call is the "called party." For an inbound call, the calling party is eventually connected to a central office (CO) that connects to the customer premises equipment (CPE) of the called party. For this inbound call, the CO is referred to as the outgoing register and the CPE as the incoming register. Signals sent from the CO are forward signals; signals sent from the CPE are backward signals. The outgoing register (CO) sends forward interregister signals and receives backward interregister signals. The incoming register (CPE) receives forward interregister signals and sends backward interregister signals.

For an outbound call, the calling party's CPE connects to the CO that switches the outbound call to the called party. For an outbound call, the signaling described above is reversed. That is, signals sent from the CPE are forward signals and signals sent from the CO are backward signals.

In addition, address signals can provide the telephone number of the called party's line. For national traffic, the address signals can also provide the telephone number of a calling party's line for automatic number identification (ANI) applications.

R2MF signals used for supervisory signaling on the network are called line signals.

For example, a calling party sends the first dialed digits to the local CO. The local CO uses these digits to determine the next CO in the connection chain. The next CO uses these first dialed digits to determine if they are the destination CO or if the call is to be switched to another CO. Eventually, the call reaches the destination CO. At the destination CO, the call is received and acknowledged. The destination CO eventually gets the last dialed digits, which exactly identify the called party.

The destination CO checks the called party's line to determine if it is clear, idle, busy, etc. The destination CO then generates and sends a B-tone backwards to the calling party to indicate the condition of the line. If the called party's line is free, the destination CO applies ringing to the line and sends ringback tones backwards to the calling party. When the called party answers the call, the calling party is switched through to the called party. If the called party's line is busy, or in some other condition, the destination CO sends this information backwards to the calling party via R2 tones. The local CO sends all information received from the destination CO to the calling party. When calls are made in countries that adhere to the full R2 protocol standard (for example, Belgium), the condition of the called party's line is always returned to the calling party.

When traversing networks, protocols, or countries, R2 tonal information can be lost. For example:

- In Italy, for an ICAPI protocol, the calling party would need to use busy tone 103 and ringback tone 105 to determine the condition of the called party's line.
- When the call is switched over a T1 span, the B-tones (also called Group B signals, see Section 1.5.2, "R2MF Signal Meanings", on page 18) are lost and the condition of the called party's line cannot be detected using R2 tones. In this environment, the application must rely on the call progress tones received to determine the condition of the called party's line.
- In Spain, the network is not a full Socotel backbone; therefore, B-tones defining the condition of the called party's line may or may not be sent backwards to the calling party.

1.5.1 R2MF Multifrequency Combinations

R2MF signaling uses a multifrequency code system based on six fundamental frequencies in the forward direction (1380, 1500, 1620, 1740, 1860, and 1980 Hz) and a different set of six frequencies in the backward direction (1140, 1020, 900, 780, 660, and 540 Hz).

Each signal is composed of two of the six frequencies, providing 15 different tone combinations in each direction. Although R2MF signaling is designed for operation on international networks with 15 multifrequency combinations in each direction, in national networks it can be used with a reduced number of signaling frequencies (for example, 10 multifrequency combinations). See the *Dialogic® Voice API Library Reference* for lists of these signal tone pairs.

1.5.2 R2MF Signal Meanings

The 15 forward signals are classified into Group I forward signals and Group II forward signals. The 15 backward signals are classified into Group A backward signals and Group B backward signals.

In general, Group I forward signals and Group A backward signals are used to control call setup and to transfer address information between the outgoing register (CO) and the incoming register (CPE). The incoming register can signal the outgoing register to change over to Group II and Group B signaling.

Group II forward signals provide the calling party’s category and Group B backward signals provide the condition of the called subscriber’s line. Group B signals, also called B-tones, are typically the last tone in the protocol. For example, typically a B-3 tone indicates that the called party’s line is busy.

Signaling must always begin with a Group I forward signal followed by a Group A backward signal that serves to acknowledge the signal just received; this Group A backward signal may request additional information. Each signal requires a response from the other party. Each response becomes an acknowledgment of the event and an event to which the other party must respond.

Backward signals serve to indicate certain conditions encountered during call setup or to announce switchover to changed signaling, for example, forward signaling switching over to backward signaling. Changeover to Group II and Group B signaling allows information about the state of the called subscriber’s line to be transferred.

The incoming register backward signals can request:

- transmission of address:
  - send next digit
  - send digit previous to last digit
  - send second digit previous to last digit sent
  - send third digit previous to last digit sent
- category of the call (the nature and origin):
  - national or international call
  - operator or subscriber

  - data transmission
  - maintenance or test call
- whether the circuit includes a satellite link
- country code and language for international calls
- information on use of an echo suppressor

The incoming register backward signals can indicate:

- address complete - send category of call
- address complete - put call through
- international, national, or local congestion
- condition of subscriber's line:
  - send SIT to indicate long term unavailability
  - line busy
  - unallocated number
  - line free - charge on answer
  - line free - no charge on answer (only for special destinations)
  - line out of order

The meaning of certain forward multifrequency combinations may also vary depending upon their position in the signaling sequence.

See the *Dialogic® Voice API Library Reference* for more details and definitions of R2MF signals.

1.5.3 R2MF Compelled Signaling

Compelled signaling protocols vary from country to country and are grouped into two main categories, both of which are supported by the Dialogic® Global Call Software:

- R2MF derived from the CCITT (International Telegraph and Telephone Consultative Committee) standard, where the response tones can carry information from the receiver to the sender. This standard provides a consistent handshake, where the sender always initiates with a forward tone, and the receiver always responds with a backward tone.
- MF Socotel, where the response tone is a standard, single frequency acknowledgment tone that cannot carry additional information. In this standard, the handshake of forward and backward tones changes direction when the receiver needs to send information back to the sender.

The Global Call Software provides network device independence by shielding the application from protocol-specific details while giving access to each protocol's full range of features. The compelled signaling feature uses tone generation and detection IDs that are defined at system initialization.

R2MF interregister signaling uses forward and backward compelled signaling. With compelled signaling, each signal is sent until a response (a return) signal is generated. This return signal is sent until responded to by the other party. Each signal stays on until the other party responds, thus compelling a response from the other party. Compelled signaling provides a balance between speed and reliability because it adapts its signaling speed to the working conditions with a minimum loss of reliability.

Compelled signaling must always begin with a Group I forward signal. For an inbound call:

- The CO starts to send the first forward signal.
- As soon as the CPE recognizes this signal, the CPE starts to send a backward signal that serves as an acknowledgment and may also request additional information.
- As soon as the CO recognizes the CPE acknowledging signal, the CO stops sending the forward signal.
- As soon as the CPE recognizes the end of the forward signal, the CPE stops sending the backward signal.
- As soon as the CO recognizes that the CPE stopped sending the backward signal, the CO may start to send the next forward signal.

The above scenario describes the CPE handling of an inbound call. The roles of the CO and the CPE are reversed when the CPE makes an outbound call.

1.6 Direct Dialing In (DDI) Service

Since DTMF, MF, and R2MF tone signals can provide the telephone number of the called subscriber's line, these signals may be used by applications providing Direct Dialing In (DDI) service, also called dialed number identification service (DNIS) and analog DNIS for direct inward dialing (DID).

DDI service allows an outside caller to dial an extension within a company without requiring an operator's assistance to transfer the call. The CO passes the last 2, 3, or 4 digits of the dialed number to the CPE, and the CPE completes the call.

Dialogic® Global Call Architecture for E1/T1 CAS/R2 — 2

This chapter describes the Dialogic® Global Call Software architecture when using E1/T1 CAS/R2 technology.

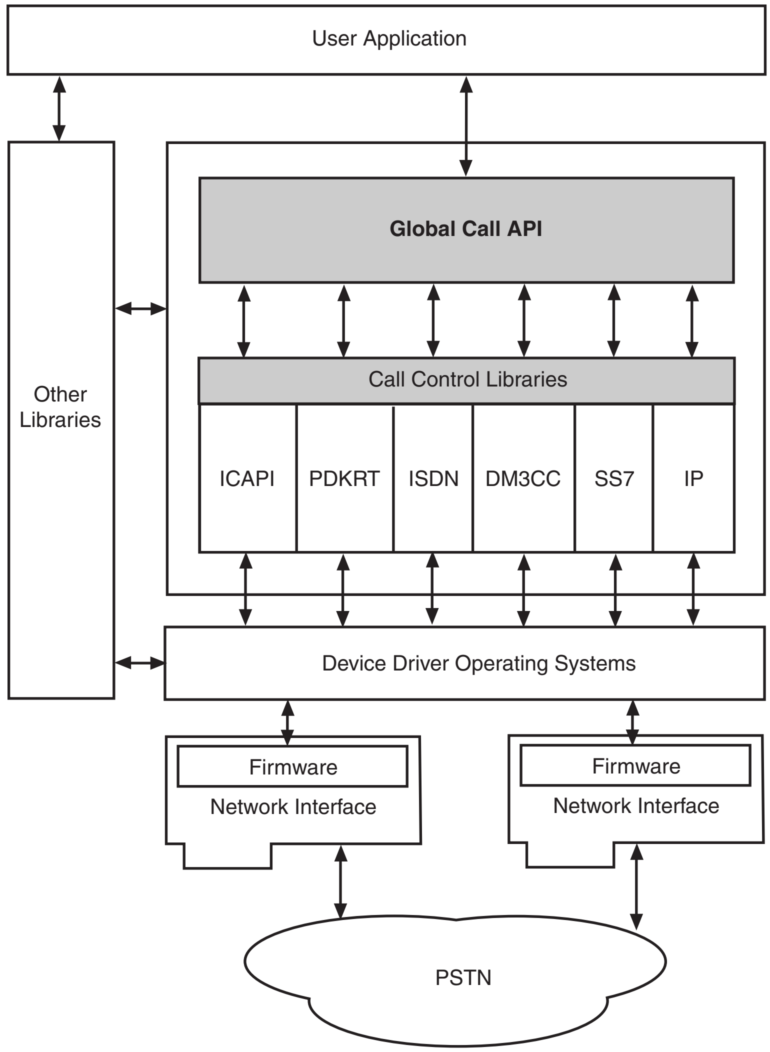
Figure 1 shows the Dialogic® Global Call Software architecture with the two key elements from an E1/T1 CAS/R2 viewpoint highlighted:

- The Dialogic® Global Call API is a library of functions that provide primarily call control, but also operation and maintenance functionality to applications.
- The underlying call control libraries provide the interface between the network and the Global Call API library.
  - GC_DM3CC_LIB is the DM3CC call control library. This library is used for call control using CAS/R2MF (PDK protocols) signaling on Dialogic® DM3 Boards. (It is also used for Integrated Services Digital Network (ISDN) on Dialogic DM3 Boards, which is not discussed in this manual; ISDN is covered in the *Dialogic® Global Call ISDN Technology Guide*.)
  - GC_PDKRT_LIB is the Protocol Development Kit Run Time (PDKRT) call control library. This library is used for call control using CAS/R2MF (PDK protocols) signaling on Dialogic® Springware Boards only.
  - GC_ICAPI_LIB is the Interface Control Application Programming Interface (ICAPI) call control library. This library is used for call control using CAS/R2MF (ICAPI protocols) signaling on Dialogic® Springware Boards only.

  Note: The development of the ICAPI protocols supported by Global Call has been capped. Customers should migrate to equivalent protocols developed using the PDK. New protocol development as well as existing protocol support will be on the PDK. ICAPI protocols are supported only on Dialogic Springware Boards. PDK protocols are supported on both Dialogic DM3 Boards and Dialogic Springware Boards.

See the *Dialogic® Global Call API Programming Guide* for more information about the Global Call architecture.

Figure 1. Dialogic® Global Call Architecture When Using E1/T1 CAS/R2



E1/T1 CAS/R2 Call Scenarios 3

The E1/T1 CAS/R2 technology currently supports all of the call scenarios that are described in the *Dialogic® Global Call API Programming Guide*. Please refer to that publication for information on call scenarios.

E1/T1 CAS/R2-Specific Operations 4

This chapter offers information for programmers who choose to design and code Dialogic® Global Call E1 CAS or T1 robbed bit applications in a Linux or Windows® environment. Topics include the following:

- Call Progress and Call Analysis . . . 25
- CAS Pattern Signal Declarations . . . 37
- Dynamic Trunk Configuration . . . 40
- Resource Association . . . 44
- Resource Allocation and Routing . . . 45
- Alarm Handling . . . 49
- Run-Time Configuration of the PDKRT Call Control Library . . . 56
- Run-Time Configuration of PDK Protocol Parameters . . . 57
- Determining the Protocol Version . . . 59
- Run-Time Control of Single or Double Hookflash on Consultation Drop for FXS/LS Protocol 60
- Retrieving Line Signaling Access . . . 63

4.1 Call Progress and Call Analysis

Call analysis consists of both pre-connect and post-connect information about the progress of the call. Pre-connect call progress determines the status of the call connection - that is, busy, no dial tone, no ringback, etc. Post-connect call analysis, which is also known as media type detection, determines the destination party's media type - that is, answering machine, fax, voice, etc. The term *call progress analysis* (CPA) is used to refer to call progress and call analysis collectively.

Note: For **CAS** (not R2) protocols, separate pre-connect CPA needs to be carried out (initiated) since the CAS signaling itself does not indicate the condition of the line at any point of call establishment. For **R2** protocols, the signaling is more intelligent. When using R2 compelled signaling, the Group B tones provide information on the condition of the line as discussed in Section 1.5.2, "R2MF Signal Meanings", on page 18.

For R2 protocols, if digits are being sent using R2 tones, pre-connect CPA is part of R2 signaling. Pre-connect CPA such as NoAnswer can only be initiated after the R2 signaling has been issued. However, R2 protocols can be configured to dial digits using DTMF/MF, in which case R2 compelled signaling is **not** used. In this case, full pre-connect CPA is supported and must be explicitly enabled.

The following sections discuss:

- Call Analysis with Dialogic® DM3 Boards
- Call Analysis with Dialogic® Springware Boards
- Call Analysis Functionality for PDK Protocols
- Tone Definitions for PDK Protocols
- Call Analysis Functionality for ICAPI Protocols

4.1.1 Call Analysis with Dialogic® DM3 Boards

Note: When using Dialogic® DM3 Boards, Dialogic® Global Call Software provides a consistent method of pre-connect call progress and post-connect call analysis across analog, CAS, and ISDN protocols. Refer to the *Dialogic® Global Call API Programming Guide* for information about this method of CPA.

The information included below is specific to the E1/T1 CAS technology and is provided for backward compatibility only. For new applications, it is recommended to use the cross-technology CPA method described in the *Dialogic® Global Call API Programming Guide*.

There are two methods available for CPA when using Dialogic® DM3 Boards: the Global Call method and the **dx_dial( )** method.

The Global Call media detection method is well suited for performing post-connect call analysis. When activated by setting the **GCPR_MEDIADETECT** parameter to GCPV_ENABLE for a particular channel, post-connect call analysis is performed as part of the **gc_MakeCall( )** function's operation. The **gc_MakeCall( )** function is used to place a call; the signal detector analyzes the incoming signals to perform CPA.

After the normal **gc_MakeCall( )** processing finishes and a GCEV_CONNECTED event is sent, call analysis runs and generates a GCEV_MEDIADETECTED event that tells the application the result of the analysis (for example, FAX, PVD, or PAMD is detected). 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.

The outcome of the analysis determines the events generated and the action that can be taken as follows:

- If the call is successful, **gc_MakeCall( )** finishes and a GCEV_CONNECTED event is sent, call analysis runs, and generates a GCEV_MEDIADETECTED event. The **gc_ResultValue( )** and **gc_GetCallInfo( )** functions can then be used to get more information about the type of media detected, such as voice, answering machine, and fax.
- If the call is not successful—for example, there is no ringback—a GCEV_DISCONNECTED event is generated and the **gc_ResultValue( )** function can be used to retrieve the reason for the failure. See the *Dialogic® Global Call API Library Reference* for error codes and the *gcerr.h* file for more information.

Note: The information above applies when using **gc_MakeCall( )** in asynchronous or synchronous mode. However, in synchronous mode, since the **gc_MakeCall( )** function must complete, the GCEV_MEDIADETECTED event is generated after the call is connected.

GCPR_MEDIADETECT and **GCPR_CALLPROGRESS** parameter settings for **gc_SetParm( )** actually allow the application to specify whether pre- or post-connect call analysis or both should be activated. This method for achieving this is shown in Table 2.

Table 2. Dialogic® Global Call API Call Progress Settings

	GCPR_CALLPROGRESS= GCPV_DISABLE	GCPR_CALLPROGRESS= GCPV_ENABLE (default)
GCPR_MEDIADETECT= GCPV_DISABLE (default)	No call progress	Pre-connect call progress **only**
GCPR_MEDIADETECT= GCPV_ENABLE	No call progress	Full call progress

As shown in this table, the default behavior (**GCPR_MEDIADETECT = GCPV_DISABLE**) disables media detection but actually activates pre-connect call progress for CAS protocols. To enable full CPA, set the **GCPR_MEDIADETECT** parameter to GCPV_ENABLE for the respective channel.

Note: For this Global Call media detection to work, a voice device must be attached to the line device and properly routed. Failure to do so will cause subsequent outgoing call attempts to fail.

The **GCPR_CALLPROGRESS** parameter can be used to enable or disable pre-connect call progress. When combined with **GCPR_MEDIADETECT**, this allows the application to specify whether to use pre-connect call progress only or full call progress. If **GCPR_CALLPROGRESS = GCPV_DISABLE**, there will be no call progress at all, regardless of the setting of **GCPR_MEDIADETECT**.

Table 3 explains call analysis support via the Global Call interface. The table applies to DM3 CAS protocols with flexible routing clusters, provided that a voice device is attached to the network device. Check on a protocol-by-protocol basis, as some might not support call analysis at all.

Table 3. Call Analysis Support on Dialogic® DM3 Boards with CAS

Call Analysis Feature	Support on Dialogic® DM3 Boards	How Obtained/Notes
Busy	Yes	Upon DISCONNECT event, call **gc_ResultValue( )**.
No ringback	No	
SIT	Yes	Upon DISCONNECT event, call **gc_ResultValue( )**.
No answer	Yes	Upon DISCONNECT event, call **gc_ResultValue( )**.
Cadence break	No	
Discarded	No	
NA	Yes	Use **GCPR_MEDIADETECT** parameter. Upon MEDIADETECTED event, call **gc_GetCallInfo( )**.
Unknown	Yes	Use **GCPR_MEDIADETECT** parameter. Upon MEDIADETECTED event, call **gc_GetCallInfo( )**.
PVD	Yes	Use **GCPR_MEDIADETECT** parameter. Upon MEDIADETECTED event, call **gc_GetCallInfo( )**.

Table 3. Call Analysis Support on Dialogic® DM3 Boards with CAS (Continued)

Call Analysis Feature	Support on Dialogic® DM3 Boards	How Obtained/Notes
PAMD	Yes	Use **GCPR_MEDIADETECT** parameter. Upon MEDIADETECTED event, call **gc_GetCallInfo( )**.
Fax	Yes	Use **GCPR_MEDIADETECT** parameter. Upon MEDIADETECTED event, call **gc_GetCallInfo( )**.
In progress	Yes	Use **GCPR_MEDIADETECT** parameter. Upon MEDIADETECTED event, call **gc_GetCallInfo( )**.

Note that the call analysis time-out parameters values apply, and they are configurable by the user. (They cannot be changed at run time.) The parameters are **CaSignalTimeout**, **CaAnswerTimeout**, and **CaPvdTimeout**; their values are found in the CHP section of the configuration (.config) file. However, they apply only to post-connect call analysis and are not used until the call moves from an initiated to a Proceeding, Alerting, or Connected state.

Another option for call analysis is provided by the Dialogic® Voice API, which provides post-connect call analysis on Dialogic DM3 Boards through the **dx_dial( )** function. Note that the Global Call method and the **dx_dial( )** method are mutually exclusive, so you must choose one or the other.

4.1.2 Call Analysis with Dialogic® Springware Boards

The **gc_GetCallInfo( )** function is used immediately following the receipt of a GCEV_CONNECTED event to retrieve this post-connect information notifying of the media type of the answering party. See the *Dialogic® Global Call API Library Reference* for more information.

Call analysis tones such as dial tone, ringback, busy, and fax are defined either in the firmware (global tone detection and global tone generation), or in the country dependent parameters (.cdp) file, or a combination of both. Tones defined in the firmware can be enabled or disabled by configuring parameters in the DX_CAP (call analysis parameter) data structure. Similarly, the DX_CAP data structure can be used to configure the voice detection algorithm that distinguishes answering machine or human speech. The default parameter values defined in the DX_CAP data structure can be changed by the **gc_LoadDxParm( )** function to fit the needs of your application. For a detailed description of enhanced call analysis (Perfect Call) and how to use call analysis, see the *Dialogic® Voice API Programming Guide*. For a detailed description of the **gc_LoadDxParm( )** function, see the *Dialogic® Global Call API Library Reference*.

Some example uses of call progress tones are as follows:

- By detecting the ringback tone, the Global Call API can count the rings and report a GCEV_DISCONNECTED event when the call is not answered within the specified number of rings.
- For telephone circuits that include analog links, the local line may not have access to all of the digital signaling information. If so, the user must modify the .cdp file accordingly to detect or generate the busy, ringback, or dial tone of the native country.

Global Tone Detection (GTD) Tone Considerations

Global Call will delete all tones and load internally required tones (used for call progress) under either of the following circumstances:

- If there is a voice device attached to the network device during **gc_OpenEx( )**
- When **gc_Attach( )** or **gc_AttachResource( )** is called, if at least one of the following statements is true:
  - A voice resource is being attached to a network device opened in the PDK library (either implicitly via **gc_OpenEx( )**, or explicitly via **gc_Attach( )** or **gc_AttachResource( )**).
  - Downloading of tones is enabled (**gc_SetParm(ldev, GCPR_LOADTONES, GCPV_ENABLE)**).

If Global Call deleted all tones during **gc_OpenEx( )**, **gc_Attach( )**, or **gc_AttachResource( )** as described above, then the application must reload any tones that it has loaded. It is recommended that the application not download tones for a voice device prior to calling **gc_OpenEx( )** if the voice device is specified in the **gc_OpenEx( )**, as the tones will be deleted. Similar considerations apply to **gc_Attach( )** and **gc_AttachResource( )**.

It is the application's responsibility to ensure that the internally required tones are available to the protocol during call setup. This can be done by either:

- Never deleting all tones, or
- If the application has deleted all tones while the voice resource is not attached, enabling downloading of tones

Caution: The application must not delete all tones while the voice resource is attached.

In any case, the application may not delete internally required tones during call setup.

Note: For PDK and ICAPI protocols, the tone IDs for any additional tones that must be redefined after calling **gc_Attach( )** or **gc_AttachResource( )** cannot be in the range from 101 to 189.

The overhead of downloading tones is expensive. Therefore, for any application that calls **gc_Attach( )** or **gc_AttachResource( )** several times on the same device (for example, when resource sharing), this overhead can be avoided by calling **gc_SetParm(ldev, GCPR_LOADTONES, GCPV_DISABLE)**. This **gc_SetParm( )** function should be called after the call to the **gc_Attach( )** or **gc_AttachResource( )** function, or after the call to the **gc_OpenEx( )** function if the voice device is specified in **gc_OpenEx( )**. It is then the application's responsibility not to delete all tones on the voice device.

4.1.3 Call Analysis Functionality for PDK Protocols

Call analysis functionality for PDK protocols is discussed in the following sections:

- Overview of Call Analysis When Using PDK Protocols
- Configuring Call Analysis in the Protocol .cdp File
- Enabling the GCEV_MEDIADETECTED Event
- Retrieving the Detected Media Type

4.1.3.1 Overview of Call Analysis When Using PDK Protocols

On both Dialogic® DM3 and Dialogic® Springware Boards, when using PDK protocols, media detection (i.e., call analysis) is completed in two parts: protocol and library.

There are parameters in the .cdp file that provide options for media detection with the protocol. The parameters are **PSL_CAMediaDetectOverride** (for Dialogic DM3 Boards) and **PSL_MakeCall_MediaDetect** (for Dialogic Springware Boards). Further information about these parameters is given in Section 4.1.3.2, "Configuring Call Analysis in the Protocol .cdp File", on page 30.

After the protocol sends the call analysis result to the library, the library determines whether to send GCEV_MEDIADETECTED to the application, independent of these PSL_ parameter settings. The GCEV_MEDIADETECTED event is disabled by default in the library, so the application must explicitly enable the event. See Section 4.1.3.3, "Enabling the GCEV_MEDIADETECTED Event", on page 31.

After receiving GCEV_MEDIADETECTED (or after receiving GCEV_CONNECTED if GCEV_MEDIADETECTED is not enabled), the **gc_GetCallInfo( )** function is used to retrieve information about the detected media type. See Section 4.1.3.4, "Retrieving the Detected Media Type", on page 32.

4.1.3.2 Configuring Call Analysis in the Protocol .cdp File

PDK protocols configure default call analysis operation through the use of two Protocol Service Layer (PSL) parameters in the protocol .cdp file (the parameter names are different for Dialogic® DM3 and Dialogic® Springware Boards):

- **PSL_CACallProgressOverride** (parameter for Dialogic DM3 Boards) **PSL_MakeCall_CallProgress** (parameter for Dialogic Springware Boards): Provides default options for call progress. Possible values are:
  - 0 (Always Off): Specifies that the call progress resource cannot be used by the protocol. This is the default value if this parameter is left undefined in the .cdp file.
  - 1 (Preferred): Specifies that the call progress resource is preferred by the protocol. This value is typically used for T1 and analog protocols. However, the protocol is able to function without call progress.
  - 2 (Pass-through): Specifies that the call progress resource is configured as specified dynamically by the application, for example, via **gc_MakeCall( )** when using Global Call. This value is typically used by E1 protocols.
- **PSL_CAMediaDetectOverride** (parameter for Dialogic DM3 Boards) **PSL_MakeCall_MediaDetect** (parameter for Dialogic Springware Boards): Provides options for media detection. Possible values are:
  - 1 (Preferred): Specifies that the media detection resource is preferred by the protocol. This setting is typically used for T1 and analog protocols. The protocol is able to function without media detection.
  - 2 (Pass-through): Specifies that the media detection resource is configured as specified dynamically by the application, for example, via **gc_MakeCall( )** or **gc_SetParm( )** when using Global Call. This value is typically used by E1 protocols. This is the default value if this parameter is left undefined in the .cdp file.

When call progress or media detection support PSL parameters are specified as pass-through values in the .cdp file, the application is permitted to define call analysis settings, for example via **gc_MakeCall( )** when using Global Call. More specifically:

- When the **PSL_CACallProgressOverride** (DM3) or **PSL_MakeCall_CallProgress** (Springware) parameter in the .cdp file is specified as 2 (Pass-through), the application may disable call progress (the default is enabled) in its call to **gc_MakeCall( )**.
- When the **PSL_CAMediaDetectOverride** (DM3) or **PSL_MakeCall_MediaDetect** (Springware) parameter in the .cdp file is **not** specified as 1 (Preferred, the default is 2, Pass-through), the application may enable media type detection (the default is disabled) in a call to **gc_MakeCall( )** or **gc_SetParm( )**.
- When call progress or media detection support PSL parameters are specified as pass-through values in the .cdp file, the application defines call analysis and/or media detection on a per call basis via the **gc_MakeCall( )** or **gc_SetParm( )** call.
- When call analysis behavior is not specified via PSL parameters in the .cdp file, the default behavior has call progress always disabled and media type detection disabled by default unless the application explicitly enables media type detection via the **gc_MakeCall( )** or **gc_SetParm( )**.
- If the call progress and/or media type detection parameters are specified in the .cdp file as 1 (Preferred) or 0 (Always Off), application setting requests (for example, the settings specified via **gc_MakeCall( )** or **gc_SetParm( )**) are ignored.

4.1.3.3 Enabling the GCEV_MEDIADETECTED Event

On Dialogic® DM3 and Dialogic® Springware Boards, PDK protocols support a method of call progress configuration using the **gc_SetConfigData( ) / gc_SetParm( )** function. The parameter used to specify call analysis (media detection) in this case is **GCPR_MEDIADETECT**. (See Table 2, “Dialogic® Global Call API Call Progress Settings”, on page 27.) This enables media type detection on a **per channel** basis.

When this method is used to enable media type detection, a GCEV_MEDIADETECTED event is returned to the application on media type detection so that the **gc_GetCallInfo( )** function can be used immediately to get information about the type of connection. The application does not have to wait for a GCEV_CONNECTED event.

Note that if this method of call progress configuration is **not** used and only **PSL_CAMediaDetectOverride / PSL_MakeCall_MediaDetect** is enabled for media detection, the application must receive a GCEV_CONNECTED event **before** the **gc_GetCallInfo( )** function can be used to get information about the type of connection. Even after the GCEV_CONNECTED event is received, the call information may not be available. (In this situation, **gc_GetCallInfo( )** returns GCCT_NA.) Consequently, the application may need to poll for the information.

On Dialogic Springware Boards, PDK protocols support another method of call analysis via the **gc_MakeCall( )** function. The **gc_MakeCall( )** function uses the **flags** parameter in the PDK_MAKECALL_BLK structure to determine if call progress and/or media type detection are enabled on a **per call** basis. The two flags are NO_CALL_PROGRESS and MEDIA_TYPE_DETECT. The default values are such that call progress is enabled and media type detection is disabled, but the bits in the **flags** parameter can be changed to enable/disable call progress and/or media type detection as required. (See PDK_MAKECALL_BLK in Chapter 9, "E1/T1 CAS/R2-Specific Data Structures".) If this method is used for media detection, the application must receive a GCEV_CONNECTED event **before** the **gc_GetCallInfo( )** function can be used as described above.

4.1.3.4 Retrieving the Detected Media Type

When the **gc_GetCallInfo( )** function is used to retrieve information about the detected media type, the **info_id** parameter to the **gc_GetCallInfo( )** function must be CONNECT_TYPE. The values that may be returned when the **info_id** parameter is CONNECT_TYPE include:

GCCT_CAD
: Connection due to cadence break

GCCT_PVD
: Connection due to voice detection

GCCT_PAMD
: Connection due to answering machine detection

GCCT_FAX
: Connection due to fax machine detection

GCCT_NA
: Connection type is not available

Whether a positive media detection result is sufficient to signal a call state change to the CONNECTED state is dependent upon the specific PDK protocol. For example, in PDK protocols where CAS signaling is required for identifying a connection, a signaling bit change must be received before signaling a CONNECTED call state change. For increased flexibility, a separate .cdp file parameter, **CDP_Connect_Upon_Media**, may be defined in the associated parameter file and used inside the protocol to enable the protocol to perform a call state change to the Connected state immediately upon positive media detection. This parameter is mostly of interest to T1 protocols.

4.1.4 Tone Definitions for PDK Protocols

On Dialogic® Springware Boards, call analysis and progress tones are mapped to US specified tones by default. PDK protocols also permit call analysis and progress tones to be customized for non-US defaults via **PSL_TONE_CP_xxx** (where xxx is the call analysis tone type, that is, BUSY, RINGBACK, etc.) parameters as specified in the protocol .cdp file.

The format of a tone definition in the .cdp file is as follows:

```
ALL TONE_t TONE_<NAME> = Frequency_1, Frequency_1_Deviation, Frequency_2, Frequency_2_Deviation,
Amplitude_1, Amplitude_2, OnTime, OnTime_Deviation, OffTime, OffTime_Deviation, Mode, Repeat
Count
```

There are two basic types of tone detection for both single and dual tones: edge detection and cadence detection.

Tone detection using the edge detection algorithm provides notification either when the energy in the specified frequency band(s) exceeds the threshold (leading-edge detection) or no longer exceeds the threshold (trailing-edge detection). Edge detection is identified by assigning a value of zero (0) to the **On Time** parameter. See Table 4 below.

Tone detection using the cadence detection algorithm provides notification when the energy in the specified frequency band(s) exceeds threshold and meets the timing requirements of the specified ring cadence. Cadence detection, like edge detection, can provide notification either when the cadence completes the specified number of cycles (**Repeat Count** parameter) or when the cadence ceases after ringing the specified number of cycles. Cadence detection is identified by assigning a non-zero value to the **On Time** parameter.

Another tone detection feature is the ability to debounce the leading edge of the tone. Rather than notifying the protocol immediately when the leading edge of the tone is detected, the protocol can specify to wait for a period of time (debounce time) before the tone signal is delivered to the protocol, that is, debouncing. This type of tone detection can be specified in the tone template as:

- **On Time**: plus half the debounce time
- **On Time Deviation**: minus half the debounce time
- **Off Time**: 0
- **Off Time Deviation**: 0
- **Repeat Count**: 0

Note: Many Dialogic Springware Boards cannot detect dual tones with frequency components closer than 65 Hz. In these instances, use a single tone template with the specified frequency band (that is, Frequency1 +/- Frequency1 Deviation) encompassing both dual tone ranges.

The meaning of each argument of a tone definition is explained in Table 4.

Table 4. TONE_t Signal Definition Parameters

Parameter Number	Name	Description	Detect/ Generate	Edge/ Cadence Detection
1	Frequency 1	Frequency of first tone (in Hz)	Detect, Generate	Edge, Cadence
2	Frequency 1 Deviation	Frequency deviation for first tone (in Hz) **Note:** The minimum recommended value for this parameter is 50.	Detect	Edge, Cadence
3	Frequency 2	Frequency of second tone (in Hz)	Detect, Generate	Edge, Cadence
4	Frequency 2 Deviation	Frequency deviation for second tone (in Hz) **Note:** The minimum recommended value for this parameter is 50.	Detect	Edge, Cadence
5	Amplitude 1	Amplitude of first tone (in dB)	Generate	Neither
6	Amplitude 2	Amplitude of second tone (in dB)	Generate	Neither
7	On Time	On duration (in milliseconds) **Note:** The minimum recommended value is 50.	Detect, Generate	Cadence

Table 4. TONE_t Signal Definition Parameters (Continued)

Parameter Number	Name	Description	Detect/ Generate	Edge/ Cadence Detection
8	On Time Deviation	On time deviation (in milliseconds) **Note:** The minimum recommended value is 50.	Detect	Cadence
9	Off Time	Off duration (in milliseconds) **Note:** The minimum recommended value is 50.	Detect, Generate	Cadence
10	Off Time Deviation	Off time deviation (in milliseconds) **Note:** The minimum recommended value is 50.	Detect	Cadence
11	Mode	Detection notification. For Dialogic **Springware** Boards, the values and their meanings are: • 0 – Report a tone match at the termination of the tone. For edge detection, this is the trailing edge. For cadence detection, this is the termination of the cadence after the specified number of cycles. • 1 – Report a tone match at the beginning of the tone. For edge detection, this is the leading edge. For cadence detection, this is the onset of cadence detection. For Dialogic **DM3** Boards, the values and their meanings are: • 0 – Report a tone match at the termination of the tone. For edge detection, this is the trailing edge. For cadence detection, this is after cadence has ended. • 1 – Report a tone match at the beginning of the tone. For edge detection, this is the leading edge. For cadence detection, this is after the first pulse has been detected and at the beginning of the second pulse. (Normal cadence detection for DM3 Boards requires at least 2 pulses.) • 2 – Report a tone match at both the beginning and end. • 3 – Disable tone detection. This is used for definitions intended only for tone generation. • 4 – Report a tone match at the beginning of the tone. For edge detection, this is the leading edge. For cadence detection, this is after the first pulse has been detected. (Unlike mode 1, this mode does not require 2 pulses.)	Detect	Edge, Cadence
12	Repeat Count	Repetition count (the number of repetitions on cycles)	Detect, Generate	Cadence

If TONE_x is previously defined, TONE_y may be set equal to TONE_x in the following manner:

```
ALL TONE_t TONE_y = TONE_x
```

The following are examples of tone declarations in a .cdp file:

```
/*
This defines the ringback tone. The currently defined tone is a
tone (440Hz+480Hz) on for 0.25 secs and off for 0.25 secs and a
ring count of 1
*/
R4 TONE_t TONE_RINGBACK = 440,0,480,0,0,0,250,0,250,0,0,1

/*
This identifies the KP tone for ANI.
*/
R4 TONE_t TONE_ANI_KP = 1100,0,1700,0,0,0,100,0,0,0,0,1
```

4.1.5 Call Analysis Functionality for ICAPI Protocols

Note: The information in this section is applicable to Dialogic® Springware Boards only. Dialogic® DM3 Boards do not use ICAPI protocols.

Global Call call analysis uses global tone detection (GTD) and timers. Some of the country dependent parameters (.cdp) files define tone templates for recognition of call progress tones. The tone IDs defined match the protocol parameter numbers (for example, parameter $103 creates tone ID# 103). See the *Dialogic® Voice API Programming Guide* for information about working with and building tone templates.

Parameter $1, $6, or $13 in the .cdp file defines the maximum time (in seconds) for a call to be answered. Within that interval, a busy tone and ringback tone can be detected. If the timer expires, the GCEV_DISCONNECTED event is reported to the application.

Two separate busy tones can be defined to accommodate two different call progress failure tones (that is, busy and out-of-order). Busy tones are defined in parameters $103 and $104 using the following format:

```
$103: - <frequency 1> <deviation> <frequency 2> <deviation>
%01: - <on time> <on deviation> <off time> <off deviation>
%02: - <number of cycles before detect>
```

Frequency is expressed in Hz; time duration is expressed in 10 ms units; unspecified values are set to 0. The deviation value for frequency 1 or 2 specifies the allowable variation in Hz. The %01 parameter relates to cadence detection. Cadence detection analyzes the audio signal on the line to detect a repeating pattern of sound (on time) and silence (off time). The deviation value for cadence detection is the allowable variation in 10 ms units. The %02 parameter specifies the number of times that the cadence on/off pattern must be detected before classifying the tone detected.

To comment out a tone template, insert a ";" (semicolon) as the first character in all three lines of the definition. If either of the busy tones is detected, the GCEV_DISCONNECTED event is reported to the application.

A ringback tone is defined in parameter $105 using the format defined above. The maximum allowable time between successive rings is defined in parameter $3 in 10 ms units. ICAPI starts a timer after receiving a ringback TONEOFF event. Typically, Connect is indicated by line signaling. However, if the network cannot indicate a Connect via line signaling, then Connect can be indicated if the next TONEON event does not arrive before the ICAPI timer expires.

To disable Connect detection, set parameter $3 to 0. Global Call will still be able to count the rings and report the GCEV_DISCONNECTED event if the maximum number of rings is reached. The maximum number of rings is set in parameter $1.

The ringback tone heard on any specific call depends on the specific CO that is serving the called party, not the local CO. If the ringback tone is not known, the recommendation is to remove this tone from the country dependent parameters (.cdp) file.

Only the call progress tone definitions in the .cdp file are used by the Global Call API. The R1 and R2 tone definitions are used only if you disable R2 MF support in the *icapi.cfg* file by setting the $17 parameter to 1.

The following are examples of the definitions of busy tones $103 and $104 and ringback tone $105 in the .cdp file:

```
******************************
*    TID # 103  BUSY      *
******************************
$103 BUSY       : 450       35
%01 cadence     : 50 10  50 10
%02 cycle       : 2

******************************
*    TID # 104  SBUSY     *
******************************
$104 SBUSY      :450        35
%01 cadence     : 25  5  25  5
%02 cycle       : 3

******************************
*    TND # 105  RINGBACK  *
******************************
$105 RINGBACK   : 450     35
%01 cadence     :  80
```

See the *Dialogic® Voice API Programming Guide* for information about using cadence, cadence detection, and tone definitions for determining the progress of outbound calls.

In addition, the following outbound parameters in the .cdp file may need to be modified when using these call progress tones:

- Number of ringback tones before returning GCEV_CALLSTATUS event with a GCRV_NOANSWER result value (typically, parameter $1 or $5)
- Default maximum time in seconds for a call to be answered (typically, parameter $1, $6, or $13)

After the .cdp file is modified as described above, whenever one of the defined conditions is detected on a channel, the **gc_MakeCall( )** function is terminated with a busy, no answer, or time-out result/error value.

Note: For ICAPI protocols, the filename specified after @0 in the .cdp file must also be specified in the *country.c* file used in Linux applications.

4.2 CAS Pattern Signal Declarations

CAS signals are defined in the cdp file for each protocol. They are typically set to default values based on protocol specifications, but can be tuned if needed. The following sections describe the formats of CAS signal declarations for different types of signals:

- CAS_SIGNAL_TRANS_t
- CAS_SIGNAL_PULSE_t
- CAS_SIGNAL_TRAIN_t

Notes: ***1.*** When editing CAS signals in the .cdp file, CAS signal *x* may be set equal to CAS signal *y*, for example: `All CAS_SIGNAL_TRANS_t CAS_x = CAS_y`

2. In the signal definitions:

- When the term "minimum" is used, this implies the information is used for detection and represents a minimum time for which the associated signal must occur.
- When the term "maximum" is used, this implies the information is used for detection and represents the maximum time that the associated signal may occur.
- When the term "nominal" is used, this implies the information is used for generation and represents the actual time to transmit the associated signal.

4.2.1 CAS_SIGNAL_TRANS_t

CAS_SIGNAL_TRANS_t signal declarations represent a transition from one signaling pattern to another. The format of a CAS_SIGNAL_TRANS_t signal definition in the .cdp file is:

```
CAS_SIGNAL_TRANS_t format = PreTrans, PostTrans, PreTransInterval, PostTransInterval,
PreTransIntervalNominal, PostTransIntervalNominal
```

Some examples are:

```
R4 CAS_SIGNAL_TRANS_t CAS_BLOCK = 00xx,11xx,50,50,80,80
R4 CAS_SIGNAL_TRANS_t CAS_UNBLOCK = 11xx,00xx,50,50,80,80
ALL CA_ SIGNAL_TRANS_t CAS_SEIZE = xxx, 10xx
```

The meaning of each argument is explained in Table 5.

Table 5. CAS_SIGNAL_TRANS_t Signal Definition Parameters

Parameter Number	Name	Description
1	PreTrans	$B_aB_bB_cB_d$ where B_i represents a signaling bit (0, 1, - = don't care, or x = don't care) for bit i, where i = a, b, c, or d. **Note:** Although T1 signaling does not have c and d bits, they must be specified for T1 protocols as don't care values.
2	PostTrans	$B_aB_bB_cB_d$ (See description of PreTrans.)
3	PreTransInterval	Minimum time for the duration of the pre-transition interval. If the value is -1 or not present, then the global timing parameter PSL_PreTransInterval is used.
4	PostTransInterval	Minimum time for the duration of the post-transition interval. 0 is allowed. If the value is -1 or is not present, then the global timing parameter PSL_PostTransInterval is used.
5	PreTransIntervalNominal	Nominal time for the duration of the pre-transition interval. If the value is -1 or is not present, then the global timing parameter PSL_PreTransIntervalNominal is used. **Note:** For Dialogic DM3 Boards, this value is always ignored, and the PreTransInterval value is used.
6	PostTransIntervalNominal	Nominal time for the duration of the post-transition interval. If the value is -1 or is not present, then the global timing parameter PSL_PostTransIntervalNominal is used. **Note:** For Dialogic DM3 Boards, this value is always ignored, and the PostTransInterval value is used.
Notes: Time intervals are specified in units of 1 millisecond. The actual granularity is implementation dependent, as is the maximum value. Due to implementation restrictions, no time value should be less than 20 milliseconds (except where 0 is allowed).		

4.2.2 CAS_SIGNAL_PULSE_t

CAS_SIGNAL_PULSE_t signal declarations represent a transition from one signaling pattern to another and then back to the original signaling pattern. The format of a CAS_SIGNAL_PULSE_t signal definition in the .cdp file is:

```
CAS_SIGNAL_PULSE_t format = OffPulse, OnPulse, PrePulseInterval, PostPulseInterval,
PrePulseIntervalNominal, PostPulseIntervalNominal, PulseIntervalMin, PulseIntervalNominal,
PulseIntervalMax
```

Some examples are:

```
R4 CAS_SIGNAL_PULSE_t CAS_WINK = 00xx,11xx,50,50,80,80,200,250,300
R4 CAS_SIGNAL_PULSE_t CAS_SEIZE_ACK = 00xx,11xx,50,50,80,80,200,250,300
```

The meaning of each argument is explained in Table 6.

Table 6. CAS_SIGNAL_PULSE_t Signal Definition Parameters

Parameter Number	Name	Description
1	OffPulse	$B_aB_bB_cB_d$ where B_i represents a signaling bit (0, 1, - = don't care, or x = don't care) for bit i, where i = a, b, c, or d. **Note:** Although T1 signaling does not have c and d bits, they must be specified for T1 protocols as don't care values.
2	OnPulse	$B_aB_bB_cB_d$ (See description of OffPulse.)
3	PrePulseInterval	Minimum time for the duration of the pre-pulse interval. 0 is allowed.
4	PostPulseInterval	Minimum time for the duration of the post-pulse interval. 0 is allowed.
5	PrePulseIntervalNominal	Nominal time for the duration of the pre-pulse. 0 is allowed. **Note:** For Dialogic DM3 Boards, this value is always ignored, and the PrePulseInterval value is used.
6	PostPulseIntervalNominal	Nominal time for the duration of the post-pulse. 0 is allowed. **Note:** For Dialogic DM3 Boards, this value is always ignored, and the PostPulseInterval value is used.
7	PulseIntervalMin	Minimum time for the duration of the pulse.
8	PulseIntervalNominal	Nominal time for the duration of the pulse.
9	PulseIntervalMax	Maximum time for the duration of the pulse.
Notes: Time intervals are specified in units of 1 millisecond. The actual granularity is implementation dependent, as is the maximum value. Due to implementation restrictions, no time value should be less than 20 milliseconds (except where 0 is allowed).		

4.2.3 CAS_SIGNAL_TRAIN_t

CAS_SIGNAL_TRAIN_t signal declarations represent a "train," that is, a sequence of pulses. The format of a CAS_SIGNAL_TRAIN_t signal definition in the .cdp file is:

```
CAS_SIGNAL_TRAIN_t format = OffPulse, OnPulse, PreTrainInterval, PostTrainInterval,
PreTrainIntervalNominal, PostTrainIntervalNominal, PulseIntervalMin, PulseIntervalNominal,
PulseIntervalMax, InterPulseIntervalMin, InterPulseIntervalNominal, InterPulseIntervalMax
```

The meaning of each argument is explained in Table 7.

Table 7. CAS_SIGNAL_TRAIN_t Signal Definition Parameters

Parameter Number	Name	Description
1	OffPulse	$B_aB_bB_cB_d$ where B_i represents a signaling bit (0, 1, - = don't care, or x = don't care) for bit i, where i = a, b, c, or d. **Note:** Although T1 signaling does not have c and d bits, they must be specified for T1 protocols as don't care values.
2	OnPulse	$B_aB_bB_cB_d$ (See description of OffPulse.)
3	PreTrainInterval	Minimum time for the duration of the pre-train interval. 0 is allowed.
4	PostTrainInterval	Minimum time for the duration of the post-train interval. Must be greater than or equal to InterPulseIntervalMax + 20.
5	PreTrainIntervalNominal	Nominal time for the duration of the pre-train interval. **Note:** For Dialogic DM3 Boards, this value is always ignored, and the PreTrainInterval value is used.
6	PostTrainIntervalNominal	Nominal time for the duration of the post-train interval. **Note:** For Dialogic DM3 Boards, this value is always ignored, and the PostTrainInterval value is used.
7	PulseIntervalMin	Minimum time for the duration of the pulse.
8	PulseIntervalNominal	Nominal time for the duration of the pulse.
9	PulseIntervalMax	Maximum time for the duration of the pulse.
10	InterPulseIntervalMin	Minimum time for the duration of the interval between pulses.
11	InterPulseIntervalNominal	Nominal time for the duration of the interval between pulses.
12	InterPulseIntervalMax	Maximum time for the duration of the interval between pulses.
Notes: Time intervals are specified in units of 1 millisecond. The actual granularity is implementation dependent, as is the maximum value. Due to implementation restrictions, no time value should be less than 20 milliseconds (except where 0 is allowed).		

4.3 Dynamic Trunk Configuration

When using Dialogic® DM3 Boards, the Dialogic® Global Call API provides the ability to perform the following dynamic configuration operations at run time:

- 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.

4.3.1 Setting the Line Type and Coding for a Trunk

Note: This feature is only applicable when using Dialogic® DM3 Boards.

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 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_SETCONFIGDATAFAIL 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_SETCONFIGDATAFAIL(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;
    }
}
```

4.3.2 Specifying the Protocol for a Trunk

Note: This feature is only applicable when using Dialogic® DM3 Boards.

The protocol used by a trunk can be dynamically configured by using the **gc_SetConfigData( )** function after devices have been opened. 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("pdk_ar_r2_io")
- **data** = "<protocol_name>", for example, "pdk_ar_r2_io" (a null-terminated string). For CAS/R2 protocols, this is the name of the CDP file (without the .cdp extension) of the protocol variant being selected. This protocol variant must already be downloaded, i.e., it must already be specified in the *pdk.cfg* file.

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_SETCONFIGDATAFAIL 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 trunk to operate with the pdk_ar_r2_io protocol.

```
static int MAX_PROTOCOL_LEN=20;
GC_PARM_BLKP ParmBlkp = NULL;
long id;
char protocol_name[]="pdk_ar_r2_io";

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_SETCONFIGDATAFAIL(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;
    }
}
```

4.4 Resource Association

In E1 CAS and T1 robbed bit protocols, a combination of line signaling and audio tones are used to establish a call. The line signaling is controlled by a network time slot device, or resource, and the tones are controlled by a voice channel (voice resource). Voice channel, voice resource, and tone resource are used interchangeably in this manual when discussing Dialogic® Global Call Software functionality.

Typically, in E1 CAS or T1 robbed bit environments, a Global Call line device consists of a network time slot resource and a voice resource. When the same voice resource is always used for a given network time slot, then this configuration is called a dedicated voice resource. The Global Call line device ID is a single ID that represents the combination of the voice and network resources that work together to establish and to tear-down calls.

In configurations with more network time slot resources than available voice (or tone) resources, the application may share these available voice resources among the time slots (resource sharing). When voice resources are shared, the Global Call line device ID represents a network time slot after issuing a **gc_OpenEx( )** function. However, before issuing a **gc_MakeCall( )** or a

gc_WaitCall( ) function, a voice resource must be attached to the Global Call line device using the **gc_Attach( )** function and then routed to the line device's network time slot. The **gc_Attach( )** function tells the Global Call protocol handler which voice channel will be used to establish the call. Once the call is established (answered), the application can use this voice resource for other calls by first detaching the voice resource using the **gc_Detach( )** function from the current line device and then attaching this voice resource to another line device using the **gc_Attach( )** function. The **gc_Detach( )** function must not be used to detach the voice resource until the call is in the connected state.

See Section 4.5, "Resource Allocation and Routing", on page 45 for more information.

4.5 Resource Allocation and Routing

E1 CAS and T1 robbed bit protocols require tone generation and detection capability and therefore require a voice or tone resource for setting up a call. Application development considerations for using dedicated voice (or tone) resources or shared voice (or tone) resources in an E1 CAS and T1 robbed bit environment are discussed in the following sections:

- Dedicated Voice Resources
- Shared Voice Resources

4.5.1 Dedicated Voice Resources

Applications requiring voice resources during the entire call (for example, voice-mail and announcements) must have enough voice channels to dedicate one channel to each network interface time slot. The Dialogic® Global Call API simplifies applications written to handle E1 CAS and T1 robbed bit protocols using dedicated voice resource configurations. To use Global Call functionality to set up dedicated resources, the application must pass both the network time slot and the voice channel to the **gc_OpenEx( )** function. Global Call uses this information to automatically:

- Open the network board device, if not previously opened (the board device is used internally by Global Call)
- Open both the voice channel and the network time slot
- Route the voice channel and network time slot together (full duplex) (CT Bus configurations only)
- Associate the voice channel and the network time slot by issuing an internal **gc_Attach( )** function

For CT Bus applications, applications using dedicated voice resources (a voice resource dedicated to a network resource) do not need to route the voice and network resources together nor issue the **gc_Attach( )** function before making a call or when handling a pending call. For applications using shared voice resources, the voice resource must be attached to a network resource before call establishment. After call establishment, this voice resource may be detached and then attached to a different network resource.

To perform activities such as routing and voice store and forward, etc., use the **gc_GetResourceH( )** function to obtain the voice and network handles associated with a line device. For example, before playing a file, you can retrieve the voice handle using the **gc_GetResourceH( )** function. If needed, you may route other resources to the network interface (for example, to send a fax) and reroute the voice channel back to the network interface before setting up or waiting for another call. You must route the same voice channel back to the associated network interface channel because these two resources were internally attached when opened.

The following example illustrates the function calls that apply when using dedicated voice resources.

Dedicated Voice Resources Example

```
            .
            .
            #define MAXCHAN  30
            struct linebag{
               LINEDEV   ldev;
               CRN       crn;
               INT       state;
            }port[MAXCHAN+1]
            .
            .
            /* Open a Global Call device with a voice channel and a network time slot */
1 ---->     if (gc_OpenEx(&linedev, ":N_dtiB1T1:P_br_r2_o:V_dxxxB1C1", 0,
                (void*)&port[port_index]) == GC_SUCCESS) {
                     /*
                      * Wait for GCEV_UNBLOCKED event.
                      */
                     .
                     .
                     /* Make an outgoing call */
2 ---->              if (gc_MakeCall(linedev, &crn, "123456", NULL, 0,
                         EV_ASYNC) == GC_SUCCESS) {
                              /*
                               * Wait for GCEV_CONNECTED event.
                               */
                     } else {
                          /* Process error from gc_MakeCall( ) */
                     }
            } else {
                  /* Process error from gc_Open( ) */
            }
            .
            .
```

Legend:

1 The **gc_OpenEx( )** function:
- Opens a Global Call line device using time slot 1 of dtiB1, opens voice channel dxxxB1C1, and configures the line device to use outbound Brazilian R2 protocol
- Opens the time slot and voice channel automatically
- Opens the network board device automatically, if not already opened to monitor the alarm
- Sets the user attribute, **usrattr**, (void*)&port[port_index] into the channel information structure

CT Bus time slot routing and attaching are done automatically. The function need only be called once for a time slot/voice channel pair.

2 The **gc_MakeCall( )** function is invoked once for each outbound call.

4.5.2 Shared Voice Resources

Applications requiring voice resources for a limited portion of the call, typically during call setup, may share voice resources among the available network time slots. For example, using a Dialogic® D/320SC Voice Board and two Dialogic® DTI/300SC (E1 interface) Network Boards, 32 voice channels may be able to handle the audio portions of the call control for 60 network interface time slots. This savings in hardware requires more complexity in writing the application, which must manage both the voice and network resources, and places limits on your call throughput. The number of calls that can be established simultaneously is limited to the number of voice resources in the system.

For voice resource sharing configurations, you need only specify the network time slot and protocol in the **gc_OpenEx( )** function. This function uses the Dialogic® Global Call API library to open the network time slot device. Your application must also open the voice device and then route and attach the necessary resources before these resources are needed for signaling. You must explicitly open the voice device by issuing a **dx_open( )** function to open the voice device selected. For routing these devices, use the native time slot routing functions or the CT Bus **nr_scroute( )** and **nr_scunroute( )** functions provided for the voice and network devices. For example, route the devices using the routing functions provided by the network and voice libraries, and then use the **gc_Attach( )** and **gc_Detach( )** functions to associate or disassociate a voice channel and a Global Call line device and therefore a network time slot. When the above sequence of operations completes, use the **gc_MakeCall( )** or **gc_WaitCall( )** function, as appropriate.

After a call is answered, the voice resource can be detached from the network time slot using the **gc_Detach( )** function and routed to another network time slot using the **nr_scunroute( )** and **nr_scroute( )** functions.

The following example illustrates the function calls that apply when using shared voice resources.

Shared Voice Resources Example

```
1 ---->    if (gc_OpenEx(&linedev, ":N_dtiB1T1:P_br_r2_o", 0,
              (void*)&port[port_index]) == GC_SUCCESS) {
                       /* process error */
           }
```

```
2 ---->    if (gc_GetNetworkH(linedev, &networkh) != GC_SUCCESS)
           {
                        /* process error */
           }


3 ---->    if ((voiceh = dx_open("dxxxB1C1", 0)) == -1)
           {
                       /* process error */
           }


4 ---->    printf("******* %d: Calling Attach %d\n", index, voiceh);
           if (gc_Attach(linedev, voiceh, EV_SYNC) != GC_SUCCESS)
           {
                        /* process error */
           }


5 ---->    if (nr_scroute(networkh, SC_DTI, voiceh, SC_VOX,
                  SC_FULLDUP) == -1)
               {
                          /* process error */
               }
                 /* Wait for GCEV_UNBLOCKED event */


6 ---->    if (gc_MakeCall(linedev, &crn, "123456", NULL, 0, EV_ASYNC)

                     != GC_SUCCESS)
               {
                         /* process error */
               }
           .
           .
           /*
            * Wait for GCEV_CONNECTED event. Voice resource may be detached
            * if necessary after receiving this event.
            */


7 ---->    if (gc_Detach(linedev, voiceh, EV_SYNC) != GC_SUCCESS)
               {
                        /* process error */
               }


8 ---->    if (nr_scunroute(networkh, SC_DTI, voiceh, SC_VOX, SC_FULLDUP) == -1)
           {
                        /* process error */

           }
```

Legend:

1 The **gc_OpenEx( )** function:

- Opens a Global Call line device using time slot 1 of dtiB1 using outbound Brazilian R2 protocol
- Opens the network board device automatically, if not already opened
- Sets the user attribute, **usrattr**, (void*)&port[port_index] into the channel information structure

The specified network time slot device is opened. This function need only be called once for a time slot.

2 The **gc_GetNetworkH( )** function retrieves the network device handle.

3 The **dx_open( )** function opens a voice device and gets a voice device handle.
4 The **gc_Attach( )** function logically connects voice and network resources.
5 The **nr_scroute( )** function routes voice and network resources together.
6 The **gc_MakeCall( )** function is invoked each time a call is to be made.
7 The **gc_Detach( )** function disassociates the voice resource from the Global Call line device.
8 The **nr_scunroute( )** function unroutes the voice and network resources.

4.6 Alarm Handling

Alarm handling using the Dialogic® Global Call API is different depending on the board architecture (DM3 or Springware). The following sections provide information about handling alarms in each architecture:

- Alarm Handling for Dialogic® DM3 Boards
- Alarm Handling for Dialogic® Springware Boards

4.6.1 Alarm Handling for Dialogic® DM3 Boards

When using Dialogic® DM3 Boards, alarms are recognized, and can also be transmitted, on a span (trunk) basis. Once an alarm is detected, all open channels on that span receive a GCEV_BLOCKED event. When the alarm is cleared, open channels receive a GCEV_UNBLOCKED event.

The **gc_SetEvtMsk( )** function can be used to mask events on a line device. Using the **gc_SetEvtMsk( )** function on a line device for a time slot sets the mask for the specified time slot only and does not apply to all time slots on the same trunk as is the case when using Dialogic® Springware Boards.

The set of Dialogic® Global Call API functions that comprise the Global Call Alarm Management System (GCAMS) interface are supported with the following restrictions:

- Using GCAMS, the application has the ability to set which detected alarms are blocking and non-blocking as described in the *Dialogic® Global Call API Programming Guide*. However, this capability applies on a span basis only. Changing which alarms are blocking and non-blocking for one time slot results in changing which alarms are blocking and non-blocking for all time slots on the span.
- Using the **gc_GetAlarmParm( )** and **gc_SetAlarmParm( )** functions to retrieve and set specific alarm parameters, for example alarm triggers, is not supported.
- The **gc_TransmitAlarms( )** and **gc_StopTransmitAlarms( )** functions can be used to start and stop the transmission of alarms to the remote side. Table 8 gives the alarms that can be transmitted on E1 and T1 interfaces.

Table 8. Alarms That Can Be Transmitted on E1 and T1 Interfaces on Dialogic® DM3 Boards

E1 Alarm	T1 Alarm	Description	Equivalent 0x1626 Parameter Value in CONFIG Files Used for Trunk Preconditioning ‡
DEA_REMOTE †	YELLOW †	Remote alarm indication (RAI)	2
DEA_UNFRAMED1 †	BLUE †	Alarm indication signal (AIS)	1
DEA_SIGNALALL1 †	—	Signaling all 1s alarm (a multi-frame alarm)	Not applicable
DEA_DISTANTMF †	—	Distant multi-frame alarm	Not applicable

† Defines that can be used in the alarm_number field of the ALARM_FIELD structure when using the **gc_TransmitAlarms( )** and **gc_StopTransmitAlarms( )** functions to start and stop the transmission of specific alarms.
‡ Trunk preconditioning is the ability to place board interface trunks in an alarm state during board initialization. See the Dialogic® DM3 Configuration Guide for more information.

The following list shows the detected (incoming) alarms that are supported on **E1** for Dialogic DM3 Boards. The dagger symbol (†) next to an alarm name indicates that the alarm is blocking by default. The default can be changed using **gc_SetAlarmConfiguration( )**. For alarms where a default threshold value is shown, the default can be changed in the .config file for the board as explained in the Dialogic® DM3 Configuration Guide.

DTE1_BPVS
: Bipolar violation count saturation. The default threshold value is 255 and the range is 0 to 255.

DTE1_CECS
: CRC4 error count saturation. The default threshold value is 255 and the range is 0 to 255.

DTE1_CRC_CFA†
: Time slot 16 CRC failure

DTE1_CRC_CFAOK
: Time slot 16 CRC failure recovered

DTE1_ECS
: Frame sync bit error count saturation. The default threshold value is 0 and the range is 0 to 255.

DTE1_FSERR
: Received frame sync error

DTE1_FSERROK
: Received frame sync error recovered

DTE1_LOOPBACK_CFA
: Diagnostic mode on the line trunk

DTE1_LOOPBACK_CFAOK
: Diagnostic mode on the line trunk recovered

DTE1_LOS
: Received loss of signal

DTE1_LOSOK
: Received loss of signal recovered

DTE1_MFSERR
: Received multi-frame sync error

DTE1_MFSERROK
: Received multi-frame sync error recovered

DTE1_RDMA
: Received distant multi-frame alarm

DTE1_RDMAOK
: Received distant multi-frame alarm recovered

DTE1_RED†
: Received red alarm

DTE1_REDOK
: Received red alarm recovered

DTE1_RLOS
: Received loss of sync

DTE1_RLOSOK
: Received loss of sync recovered

DTE1_RRA†
: Received remote alarm

DTE1_RRAOK
: Received remote alarm recovered

DTE1_RSA1
: Received signaling all 1's

DTE1_RSA1OK
: Received signaling all 1's recovered

DTE1_RUA1
: Received unframed all 1's

DTE1_RUA1OK
: Received unframed all 1's recovered

The following list shows the detected (incoming) alarms that are supported on **T1** for Dialogic DM3 Boards. The dagger symbol (†) next to an alarm name indicates that the alarm is blocking by default. The default can be changed using **gc_SetAlarmConfiguration( )**. For alarms where a default threshold value is shown, the default can be changed in the .config file for the board as explained in the Dialogic® DM3 Configuration Guide.

DTT1_BPVS
: Bipolar violation count saturation. The default threshold value is 255 and the range is 0 to 255.

DTT1_ECS
: Frame sync bit error count saturation. The default threshold value is 0 and the range is 0 to 255.

DTT1_FERR
: Two out of four consecutive frame bits (F bit) in error. The default threshold value is 0 and the range is 0 to 255.

DTT1_LOOPBACK_CFA
: Diagnostic mode on the line trunk

DTT1_LOOPBACK_CFAOK
: Diagnostic mode on the line trunk recovered

DTT1_LOS
: Initial loss of signal detected

DTT1_LOSOK
: Signal restored

DTT1_OOF
: Out of frame error count saturation. The default threshold value is 0 and the range is 0 to 255.

DTT1_RBL
: Received blue alarm

DTT1_RBLOK
: Received blue alarm restored

DTT1_RCL
: Received carrier loss

DTT1_RCLOK
: Received carrier loss restored

DTT1_RED†
: Received a red alarm condition

DTT1_REDOK
: Red alarm condition recovered

DTT1_RLOS
: Received loss of sync

DTT1_RLOSOK
: Received loss of sync restored

DTT1_RYEL†
: Received yellow alarm

DTT1_RYELOK
: Received yellow alarm restored

4.6.2 Alarm Handling for Dialogic® Springware Boards

As described in the *Dialogic® Global Call API Library Reference*, the GCEV_BLOCKED event indicates that a line is blocked and the application cannot issue call-related function calls, and the GCEV_UNBLOCKED event indicates that the line has become unblocked. For example, an alarm condition has occurred or has been cleared, respectively. These events are generated on every opened line device associated with the trunk on which the alarm occurs, if the event is enabled.

These events are enabled by default. The application may disable and enable the events by using the **gc_SetEvtMsk( )** function.

Setting the event mask on any line device that represents a time slot will result in setting the mask to the same value on all time slot level line devices on the same trunk. Additionally, setting the event mask on a line device that represents the board will have the same effect (that is, it will set the mask for all time slot level line devices on that trunk).

When an alarm occurs on a Dialogic® Global Call line device, the application must call the **dx_stopch( )** function to stop any application initiated voice processing, such as **dx_play( )** and **dx_record( )**, that is associated with that line device. The application should wait for the receipt of the GCEV_UNBLOCKED event that signals the end of the alarm condition; then the application can proceed with its call processing (for example, making or receiving calls).

Alarm notification can be configured for time slot devices using the Global Call Alarm Management System (GCAMS). The Global Call functions that comprise the GCAMS interface for alarm management are supported. See the *Dialogic® Global Call API Programming Guide* for more information on GCAMS and the *Dialogic® Global Call API Library Reference* for more information on the GCAMS functions.

The **gc_TransmitAlarms( )** and **gc_StopTransmitAlarms( )** functions can be used to start and stop the transmission of alarms to the remote side. Table 9 gives the alarms that can be transmitted on E1 and T1 interfaces.

Table 9. Alarms That Can Be Transmitted on E1 and T1 Interfaces on Dialogic® Springware Boards

E1 Alarm	T1 Alarm	Description
DEA_REMOTE †	YELLOW †	Remote alarm indication (RAI)
DEA_UNFRAMED1 †	BLUE †	Alarm indication signal (AIS)
DEA_SIGNALALL1 †	—	Signaling all 1s alarm (a multi-frame alarm)
DEA_DISTANTMF †	—	Distant multi-frame alarm
† Defines that can be used in the alarm_number field of the ALARM_FIELD structure when using the **gc_TransmitAlarms( )** and **gc_StopTransmitAlarms( )** functions to start and stop the transmission of specific alarms.		

The following list shows the detected (incoming) alarms that are supported on **E1** for Dialogic Springware Boards. The dagger symbol (†) next to an alarm name indicates that the alarm is blocking by default. The default can be changed using **gc_SetAlarmConfiguration( )**.

DTE1_BPVS†
: Bipolar violation count saturation

DTE1_BPVSOK
: Bipolar violation count saturation recovered

DTE1_CECS†
: CRC4 error count saturation

DTE1_CECSOK
: CRC4 error count saturation recovered

DTE1_DPM†
: Driver performance monitor failure

DTE1_DPMOK
: Driver performance monitor failure recovered

DTE1_ECS†
: Error count saturation

DTE1_ECSOK
: Error count saturation recovered

DTE1_FSERR†
: Received frame sync error

DTE1_FSERROK
: Received frame sync error recovered

DTE1_LOS†
: Received loss of signal

DTE1_LOSOK
: Received loss of signal recovered

DTE1_MFSERR†
: Received multi-frame sync error

DTE1_MFSERROK
: Received multi-frame sync error recovered

DTE1_RDMA†
: Received distant multi-frame alarm

DTE1_RDMAOK
: Received distant multi-frame alarm recovered

DTE1_RED
: Received red alarm

DTE1_REDOK
: Received red alarm recovered

DTE1_RLOS†
: Received loss of sync

DTE1_RLOSOK
: Received loss of sync recovered

DTE1_RRA†
: Received remote alarm

DTE1_RRAOK
: Received remote alarm recovered

DTE1_RSA1†
: Received signaling all 1’s

DTE1_RSA1OK
: Received signaling all 1’s recovered

DTE1_RUA1†
Received unframed all 1's

DTE1_RUA1OK
Received unframed all 1's recovered

The following list shows the detected (incoming) alarms that are supported on **T1** for Dialogic Springware Boards. The dagger symbol (†) next to an alarm name indicates that the alarm is blocking by default. The default can be changed using **gc_SetAlarmConfiguration( )**.

DTT1_B8ZSD†
Bipolar eight zero substitution detected

DTT1_B8ZSD
Bipolar eight zero substitution detected recovered

DTT1_BPVS†
Bipolar violation count saturation

DTT1_BPVSOK
BPVS restored

DTT1_DPM†
Driver performance monitor

DTT1_DPMOK
Driver performance monitor restored

DTT1_ECS†
Error count saturation

DTT1_ECSOK
Error count saturation recovered

DTT1_FERR†
Frame bit error

DTT1_FERROK
Frame bit error restored

DTT1_LOS†
Initial loss of signal detected

DTT1_LOSOK
Signal restored

DTT1_OOF†
Out of frame error; count saturation

DTT1_OOFOK
Out of frame restored

DTT1_RBL†
Received blue alarm

DTT1_RBLOK
Received blue alarm recovered

DTT1_RCL†
Received carrier loss

DTT1_RCLOK
Received carrier loss restored

DTT1_RED†
Received a red alarm condition

DTT1_REDOK
Red alarm condition recovered

DTT1_RLOS†
Received loss of sync

DTT1_RLOSOK
Received loss of sync restored

DTT1_RYEL†
Received yellow alarm

DTT1_RYELOK
Received yellow alarm restored

4.7 Run-Time Configuration of the PDKRT Call Control Library

Note: The information in this section is applicable to Dialogic® Springware Boards only.

Table 10 shows the parameters of the PDKRT call control library that can be configured using the real time configuration management (RTCM) functions. The **gc_GetConfigData( )** function can be used to retrieve the target object configuration, and the **gc_SetConfigData( )** function can be used to update the target object configuration.

Note: Since these parameters are statically defined, the **gc_QueryConfigData( )** is not applicable.

Table 10. Configurable PDKRT Call Control Library Parameters

Set ID	Parm ID	Target Object Type	Description	Data Type	Access Attribute*
GCSET_ CALLINFO	CALLINFO TYPE	GCTGT_CCLIB_ CRN	Calling info type (alternative to **gc_GetCallInfo( )**)	string	GC_R_O
	CATEGORY_DIGIT	GCTGT_CCLIB_ CRN	Category digit type (alternative to **gc_GetCallInfo( )**)	char	GC_R_O
*Access attributes are: GC_W_I: update GC_R_O: retrieve only GC_W_N: update only at null state GC_W_X: not available					

Table 10. Configurable PDKRT Call Control Library Parameters (Continued)

Set ID	Parm ID	Target Object Type	Description	Data Type	Access Attribute*
	CONNECT_ TYPE	GCTGT_CCLIB_ CRN	Connect type (alternative to **gc_GetCallInfo( )**)	char	GC_R_O
GCSET_ PARM	GCPR_ CALLING PARTY	GCTGT_CCLIB_ CHAN	Calling party (alternative to **gc_GetParm( )**/ **gc_SetParm( )**)	string	GC_W_I
	GCPR_ LOADTONES	GCTGT_CCLIB_ CHAN	Load tones (alternative to **gc_GetParm( )**/ **gc_SetParm( )**)	short	GC_W_I
	GCPR_ MEDIADETECT	GCTGT_CCLIB_ CHAN	Set Media Detect (alternative to **gc_SetParm( )**)	short	GC_W_I
GCSET_ ORIG_ ADDR	GCPARM_ ADDR_DATA	GCTGT_CCLIB_ CHAN	Calling number (alternative to **gc_SetCallingNum( )**)	string	GC_W_I
*Access attributes are: GC_W_I: update GC_R_O: retrieve only GC_W_N: update only at null state GC_W_X: not available					

4.8 Run-Time Configuration of PDK Protocol Parameters

Note: The information in this section is applicable to Dialogic® Springware Boards only.

Configurable PDK protocol parameters are grouped into two sets:

- Protocol state information (PSI) variable parameters
- Protocol service layer (PSL) variable parameters

Note: To avoid errors, the PSI and PSL parameters of a GCTGT_PROTOCOL_CHAN channel are allowed to be changed only when the channel object does not have an active call.

PSI variable parameters are interpreted by the PDK run-time component (PDKRT).The names of the PSI variable parameters (beginning with CDP_) are found in the .cdp file. The PSI parameters that can be accessed via **gc_GetConfigData( )**, **gc_SetConfigData( )**, and **gc_QueryConfigData( )** are protocol dependent. Refer to the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for further information.

The PSL variable parameters are not available to the protocol state machine, but rather are used by the protocol services layer to control the behavior of various network and voice functions. The names of the PSL variable parameters begin with PSL_ and SYS_. No variation in the names is allowed. These parameters are required to control protocol parameters (e.g., timing) or they may control the behavior of the underlying implementation. In the latter case, the parameters will most likely have a platform tag. All of these parameter names must begin with PSL. The PSL parameters that can be accessed via **gc_GetConfigData( )**, **gc_SetConfigData( )**, and **gc_QueryConfigData( )** are shown in Table 11.

Table 11. PSL and SYS Parameters

PSL Variable Name	Data Type
PSL_AcceptCallDefaultNumOfRings	Integer
PSL_AnswerCallDefaultNumOfRings	Integer
PSL_MakeCall_CallProgress	Integer
PSL_MakeCall_MediaDetect	Integer
PSL_DefaultMakeCallTimeout	Integer
PSL_DXCAS_HOOKFLASH_DURATION	Integer
SYS_FEATURES	String
SYS_PSINAME	String

Table 12 shows the Set ID and Parm ID for these parameter types.

Table 12. Configurable PDK Protocol Parameters

Set ID	Parm ID	Target Object Type	Explanation	Access Attribute**
PDKSET_ PSI_VAR *	Dynamically assigned	GCTGT_ PROTOCOL_ SYSTEM, GCTGT_PROTOCOL_ CHAN	Protocol state information (PSI) variable parameters	GC_W_N
PDKSET_ SERVICE_ VAR	Dynamically assigned	GCTGT_ PROTOCOL_ SYSTEM, GCTGT_PROTOCOL_ CHAN	Protocol service layer (PSL) variable parameter and system parameters	GC_W_N
*Indicates that CAS pattern signals and tones cannot be accessed. **Access attributes are: GC_W_I: update GC_R_O: retrieve only GC_W_N: update only at null state GC_W_X: not available				

The PDK GCTGT_PROTOCOL_SYSTEM target object is not available until the first **gc_OpenEx( )** function is called to run this protocol.

The Global Call application can call **gc_GetConfigData( )** to retrieve protocol configuration information or **gc_SetConfigData( )** to set protocol configuration information. Since these parameters are protocol dependent, their parameters are dynamically assigned when a protocol is loaded into the PDKRT. Therefore, a Global Call application must call **gc_QueryConfigData( )** to find the parameter information (set ID, parm ID, and value data type, etc.) first. For more information about these functions, refer to the *Dialogic® Global Call API Programming Guide*.

The pair (target object type, target object ID) supporting **gc_QueryConfigData( )** to find PDKRT protocol parameter information can be one of the following:

- (GCTGT_PROTOCOL_SYSTEM, Global Call protocol ID)
- (GCTGT_PROTOCOL_CHAN, Global Call line device ID)

For a given protocol, although the GCTGT_PROTOCOL_SYSTEM target object and GCTGT_PROTOCOL_CHAN target object share the same set ID and parm ID for PSI variables, they can have different values. When a new GCTGT_PROTOCOL_CHAN target object is opened, it gets a copy of the current PSI variable configuration of GCTGT_PROTOCOL_SYSTEM target object. Under this situation, changes to the GCTGT_PROTOCOL_SYSTEM target object configuration will not affect the configuration of the GCTGT_PROTOCOL_CHAN target object. But the GCTGT_PROTOCOL_SYSTEM target object shares the same PSL variable configuration with other GCTGT_PROTOCOL_CHAN target objects.

The following example shows how to set the **CDP_ANI_ENABLED** parameter for channel ldev running a PDK protocol at the NULL state in asynchronous mode.

Note: Error handling is not shown.

```
    GC_PARM t_SourceParm, t_DestParm;
    GC_PARM_ID t_ParmIDSt;
    char t_name[20] = "CDP_ANI_ENABLED";
    long request_id;
    LINEDEV ldev;
    GC_PARM_BLK * t_pParmBlk = NULL;

/* first find the parameter info by calling gc_QueryConfigData() function */
    t_SourceParm.padress = t_name;              /* Pass the PSI variable name */
    memset(&t_ParmIDSt, 0, sizeof(GC_PARM_ID));
    t_DestParm.pstruct = &t_ParmIDStruct;       /* Pass desired the parm info */
    gc_QueryConfigData(GCTGT_PROTOCOL_CHAN, ldev, &t_SourceParm,
                       GCQUERY_PARM_NAME_TO_ID, &t_DestParm);

/* Call GC utility function to insert a parameter data to GC_PARM_BLK */
    gc_util_insert_parm_val(&t_pParmBlk, t_ParmIDStruct.set_ID,
                            t_ParmIDStruct.parm_ID, sizeof(int), 10);

    /* Call gc_SetConfigData() function to set the "CDP_ANI_ENABLE" */
    gc_SetConfigData(GCTGT_PROTOCOL_CHAN, ldev, t_pParmBlk, 0,
                     GCUPDATE_ATNULL, &request_id, EV_ASYNC);
    ...
    /* Call GC utility function to release the memory after using the GC_PARM_BLK */
    gc_util_delete_parm_blk(t_pParmBlk);
```

4.9 Determining the Protocol Version

Note: The information in this section is applicable to Dialogic® Springware Boards only.

The following software code demonstrates how you can determine the Dialogic® Global Call protocol version you are running.

```
#include <gclib.h>
#include <gcerr.h>
#include <srllib.h>
```

```
int main()
{
     LINEDEV     ldev;
     GC_PARM     parm;
     int         retcode;
     METAEVENT   metaevent;
     parm.paddress = NULL;
     int         mode;

ifdef _WIN32
    mode = SR_STASYNC|SR_POLLMODE;
#else
    mode = SR__POLLMODE;
#endif

     if (sr_setparm(SRL_DEVICE, SR_MODELTYPE, &mode) == -1)
      {
        // Error processing
      }

     gc_Start(NULL);
     retcode = gc_Open(&ldev,":P_na_an_io:N_dtiB1T1:V_dxxxB1C1", 0);
     if (retcode != GC_SUCCESS)
     {
          // Error processing
     }

     sr_waitevt(50);
     retcode = gc_GetMetaEvent(&metaevent);
     if (retcode != GC_SUCCESS)
     {
         // Error processing
     }
     if (metaevent.flags & GCME_GC_EVENT)
     {
          if (metaevent.evttype == GCEV_UNBLOCKED)
          {
               if (gc_GetParm(ldev, GCPR_PROTVER, &parm) == GC_SUCCESS)
               {
                    printf("The protocol version: %s\n", parm.paddress);
               }
               else
               {
                     // Error processing
               }
          }
     }

     gc_Close(ldev);
     gc_Stop();
     return(0);
}
```

4.10 Run-Time Control of Single or Double Hookflash on Consultation Drop for FXS/LS Protocol

Note: The information in this section is applicable to Dialogic® DM3 Boards only. For information about boards supported and the features supported on each board, see the Release Guide and Release Update for your Dialogic® Software release.

Run-time control of sending either a single or double hookflash when dropping a consultation call on a supervised transfer is supported for Dialogic® DM3 Boards using the United States T1 FXS/LS Bidirectional protocol.

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

Run-time control of single or double hookflash is implemented using the **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. The **GCPARM_CONSDROP_HKFLASH_OVERRIDE** parmID 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

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.

4.11 Retrieving Line Signaling Access

Note: The information in this section is applicable to Dialogic® DM3 Boards only.

The **gc_Extension( )** function can be used to retrieve the current transmit/receive ABCD signaling bits on a particular channel. For this feature, the **gc_Extension( )** function should use GCTGT_GCLIB_CHAN as target type, the Global Call device handle for the line device as the target ID, and DM3CC_EXID_TXRX_SIGBITS_GET as the extension ID.

The following example shows how to retrieve the signaling bits. The format of the response is explained below.

```
#include <iostream.h>
#include "srllib.h"
#include "gclib.h"
#include "gcerr.h"
#include "dm3cc_parm.h"

/* Some macros to get the signaling bits */
#define GET_TX_BITS(x)        (( x & 0xF0 ) >> 4 )
#define GET_RX_BITS(x)        (( x & 0xF ))

LINEDEV       g_channel;
GC_PARM_BLKP  g_pblkp = NULL;
GC_PARM_DATAP g_parmp = NULL;
METAEVENT     g_EvtData;
int           g_TxABCDbits;
int           g_RxABCDbits;
int           g_SignalingBits;
```

```
void main( void )
{
    gc_Start( NULL );
    if( gc_OpenEx( &g_channel, ":N_dtiB1T1:V_dxxxB1C1:P_dm3", 0, NULL ) != GC_SUCCESS )
    {
        gc_Stop();
        return;
    }

     /* Wait for GCEV_UNBLOCKED event */

    gc_Extension( GCTGT_GCLIB_CHAN, g_channel, DM3CC_EXID_TXRX_SIGBITS_GET, g_pblkp,
                  NULL, EV_ASYNC );

    /* Wait for GCEV_EXTENSIONCMPLT event */

    g_parmp = gc_util_next_parm( &(((EXTENSIONEVTBLK *)g_EvtData.extevtdatap)->parmblk ),
                                 NULL );

    if( g_parmp == NULL )
    {
        cout << "No parameters in event GC_PARM_BLK." << endl;
    }
    else
    {
        g_SignalingBits = *((int *)parmp->value_buf );
        g_TxABCDbits    = GET_TX_ BITS( g_SignalingBits );
        g_RxABCDbits    = GET_RX_BITS( g_SignalingBits );

        cout << "Signaling Bits:" << endl;
        cout << "    Transmit ABCD Bits = " << g_TxABCDbits << "." << endl;
        cout << "    Receive ABCD Bits  = " << g_RxABCDbits << "." << endl;
    }

    gc_ResetLineDev( g_channel, EV_SYNC );
    gc_Close( g_channel );
    gc_Stop();
}
```

The response is the GCEV_EXTENSIONCMPLT event, which will contain a GC_PARM_DATA pointer that is structured as follows:

```
typedef struct
{
    unsigned short      set_ID;            /* Set ID (two bytes long)*/
    unsigned short      parm_ID;           /* Parameter ID (two bytes long) */
    unsigned char       value_size;        /* Size of value_buf in bytes */
    unsigned char       value_buf[1];      /* Address to the parm value buffer */
}GC_PARM_DATA, *GC_PARM_DATAP;
```

The fields of GC_PARM_DATA will be set to the following parameters:

- set_ID = CCSET_SIG_BITS
- parm_ID = CCPARM_CURRENT_STATE
- value_size = 0x1
- value_buf[1] = see Table 13

Table 13. Bit Positioning in GC_PARM_DATA value_buf Element

Bit No.	7	6	5	4	3	2	1	0
Value	A_{TX}	B_{TX}	C_{TX}	D_{TX}	A_{RX}	B_{RX}	C_{RX}	D_{RX}

Setting the Initial Bit Pattern

In addition to using Global Call functions to retrieve the bit values, you can set the initial bit pattern that is sent on the line when the board is downloaded. To do this, add or change the following parameter in the CHP section in the .config file for the firmware:

```
[CHP]
SetParm=0x1316,0xfd    ! Initial Bit Pattern on the line - should be 0xF<pattern>, where
<pattern> is the ABCD bit values. The default is 0xfd -> ABCD=1101 (blocking pattern for E1)
```

This allows the application to know what the initial bit pattern is whenever the board is downloaded.

E1/T1 CAS/R2 Protocols 5

This chapter describes the E1/T1 CAS/R2 protocols supported by the Dialogic® Global Call Software. Topics include:

- Protocols Supported. 67
- Protocol File Naming Conventions 68
- Protocol Components 69

Notes: ***1.*** See the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for more information about using the protocols and the country dependent parameter (CDP) files, including detailed procedures for configuring country dependent parameters and for downloading the protocol and CDP file.

2. With newer releases of Dialogic® System Release Software, the Dialogic® Global Call protocols can be installed as part of the System Release or with a Service Update for the System Release. You do not have to install the Global Call protocols separately as in the past.

3. The development of the ICAPI protocols supported by Global Call has been capped. Customers should migrate to equivalent protocols developed using the Protocol Development Kit (PDK). New protocol development as well as existing protocol support will be on the PDK. ICAPI protocols are supported only on Dialogic® Springware Boards. PDK protocols are supported on both Dialogic® DM3 Boards and Springware Boards.

5.1 Protocols Supported

The Dialogic® Global Call protocols available are listed in the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*. For the most up-to-date list of available protocols, contact your nearest Dialogic Sales Office.

The protocol and parameters used at the application's interface to the PTT must complement those used by the local central office (CO). To maintain compatibility with the local PTT, Dialogic provides .cdp country dependent parameter files that can be modified to satisfy local requirements. User selectable options allow customization of the country dependent parameters to fit a particular application or configuration within a country (for example, switches within the same country may use the same protocol but may require different parameter values for local use). These parameters (for example, the number of DNIS digits, time-outs, party calling number, idle patterns, signaling patterns, and protocol-specific definitions) are specified in the .cdp file and may be modified at configuration time (that is, at any time before starting your application). See the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for additional information.

5.2 Protocol File Naming Conventions

When a protocol is installed on your system, several files are installed, including the protocol module(s) and country dependent parameter files. For most protocols, the files are named according to the conventions in Table 14.

Table 14. Protocol File Naming Conventions

File Name	Description
ccl_cc_tt_d.hot and *ccl_cc_tt_d.qs*	PDK protocol module (Dialogic® DM3 Boards)
ccl_cc_tt_d.psi	PDK protocol module (Dialogic® Springware Boards)
ccl_cc_tt_d.so or *ccl_cc_tt_ffff_d.so*	ICAPI protocol module for Linux (Dialogic® Springware Boards)
ccl_cc_tt_d.dll or *ccl_cc_tt_ffff_d.dll*	ICAPI protocol module for Windows® (Dialogic® Springware Boards)
ccl_cc_tt_d.cdp or *ccl_cc_tt_ffff_d.cdp*	Country dependent parameter file (Dialogic® DM3 Boards and Dialogic® Springware Boards)

where:

- **ccl** indicates the call control library for which the protocol is written, for example, pdk represents the PDKRT call control library. For the ICAPI call control library, **ccl** is blank.
- **cc** is a 2-character ISO country code, regional code (for example, es = Spain, fr = France, mx = Mexico, na = North America, etc.), or an indication of a switch-specific protocol.
- **tt** is a 2-character protocol type. Valid types are:
  - **em:** a T1 protocol using E&M signaling with support for DTMF digits only
  - **mf:** a T1 protocol using E&M signaling with support for MF digits
  - **r2:** a protocol using R2 MFC signaling
  - **r1:** a protocol using R1 MFC signaling
  - **e1:** a pulse, MF SOCOTEL, or other E1 protocol
  - **sw:** a protocol that is switch specific
  - **ls:** a loop start protocol
- **d** is a 1- or 2-character direction indicator. Valid directions are:
  - **i:** inbound
  - **o:** outbound
  - **io:** inbound/outbound
- **ffff** is optional and defines a special software or hardware feature supported by the protocol; 1 to 4 characters. If the protocol type is “sw”, then this field provides additional information about the switch.

  Note: Requires ICAPI call control library level 2, or else a compatibility error, EGC_COMPATIBILITY, will be generated when the application attempts to load the protocol.

The protocol name used in the **devicename** parameter of the **gc_OpenEx( )** function is the root name of the .cdp file. (On Dialogic® DM3 Boards, the protocol is determined at board initialization time and not when a Global Call device is opened. For compatibility, the **gc_OpenEx( )** protocol name may be specified for Dialogic DM3 Boards, but it is not used.)

Most ICAPI protocol releases use separate protocol modules to handle the inbound and the outbound portions of a protocol. For example, Table 15 describes the files included for the Argentina R2 ICAPI protocol.

Table 15. Sample ICAPI Protocol File Set

Description	Protocol Files	
	Linux	Windows®
Inbound protocol module	*ar_r2_i.so*	*ar_r2_i.dll*
Outbound protocol module	*ar_r2_o.so*	*ar_r2_o.dll*
Inbound country dependent parameters	*ar_r2_i.cdp*	*ar_r2_i.cdp*
Outbound country dependent parameters	*ar_r2_o.cdp*	*ar_r2_o.cdp*

PDK protocols are bidirectional protocols. For example, Table 16 describes the files included with the Argentina R2 PDK protocol.

Table 16. Sample PDK Protocol File Set

Description	Protocol Files
	Linux and Windows®
Bidirectional protocol module (Dialogic® DM3 Boards)	*pdk_r2_io.hot, pdk_r2_io.qs*
Bidirectional protocol module (Dialogic® Springware Boards)	*pdk_r2_io.psi*
Bidirectional country dependent parameters (Dialogic® DM3 Boards and Dialogic® Springware Boards)	*pdk_ar_r2_io.cdp*

5.3 Protocol Components

Each protocol requires specific firmware parameter file(s) to be downloaded to the voice and network boards:

- Protocol Modules
- Country Dependent Parameter (.cdp) Files

5.3.1 Protocol Modules

These files contain protocol specific information and are dynamically linked to the application as needed.

PDK protocols are supported on both Dialogic® DM3 Boards and Dialogic® Springware Boards. For Dialogic DM3 Boards, the protocol modules are .hot and .qs files. For Dialogic Springware

Boards, the protocol module is a protocol state information (.psi) file, a binary file that is interpreted by the PDK run-time component (PDKRT).

ICAPI protocols are supported on Dialogic Springware Boards only. The protocol modules for Linux are .so files. The protocol modules for Windows® are .dll files.

5.3.2 Country Dependent Parameter (.cdp) Files

These files contain country specific and protocol specific parameters for use by the Dialogic® Global Call Software. Country dependent parameter (.cdp) files may be customized. Descriptions of the country dependent parameters most likely to be modified for a protocol are provided in the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*.

For ICAPI protocols, the special parameter @0 identifies the protocol to be run. This parameter specifies the name of the protocol module (ignoring the filename extension and without the path) to be run by the application. Two variations of the same protocol can be run if two .cdp files point to the same protocol module filename after @0.

The .cdp file should be located only under the installation directory:

- For Linux: $INTEL_DIALOGIC_CFG
- For Windows®: %INTEL_DIALOGIC_CFG%

Building Dialogic® Global Call E1/T1 CAS/R2 Applications — 6

This chapter describes the E1/T1 CAS/R2 specific header files and libraries required when building applications.

- Header Files 71
- Required Libraries 71
- Required System Software 71

6.1 Header Files

When compiling Dialogic® Global Call applications for the E1/T1 CAS/R2 technology, it may be necessary to include the following header files in addition to the standard Global Call header files, which are listed in the *Dialogic® Global Call API Library Reference* and *Dialogic® Global Call API Programming Guide*:

For Dialogic® DM3 Boards

dm3cc_parm.h
: required when using Dialogic DM3 Boards

For Dialogic® Springware Boards

gcpdkrt.h
: required when using PDK error codes, the PDK_MAKECALL_BLK structure for call analysis, or logging via the **gc_Start( )** function

icapi.h
: required when using ICAPI error codes and features

6.2 Required Libraries

When building Dialogic® Global Call applications for the E1/T1 CAS/R2 technology, it is not necessary to link any libraries other than the standard Global Call library, *libgc.lib.*

6.3 Required System Software

The Dialogic® System Software must be installed on the development system. See the Software Installation Guide for your Dialogic® software release for further information.

Debugging Dialogic® Global Call E1/T1 CAS/R2 Applications 7

The Dialogic® Global Call debugging utilities are described in this chapter.

- Introduction 73
- Debugging Applications That Use PDK Protocols 73
- Debugging Applications That Use ICAPI Protocols 78

Note: The information in this chapter is applicable to Dialogic® Springware Boards only. For information about the pdktrace tool used with Dialogic® DM3 Boards, see the Diagnostics Guide for your Dialogic® software release. The pdktrace tool requires Dialogic® Global Call Protocols Version 4.1 or later.

7.1 Introduction

The Dialogic® Global Call Software includes powerful debugging capabilities for troubleshooting protocol-related problems, including the ability to generate a detailed log file. These debugging tools should not be used during normal operations or when running an application for an extended period of time since they increase the processing load on the system and they can quickly generate a large log file.

Note: Only run the debugging and logging utilities on a limited number of channels at a time to avoid the possibility of losing events.

7.2 Debugging Applications That Use PDK Protocols

This section discusses the following topics:

- Enabling and Disabling the Logging
- Populating and Using a CCLIB_START_STRUCT
- Defining the GC_PDK_START_LOG Environment Variable

7.2.1 Enabling and Disabling the Logging

The Dialogic® Global Call PDKRT (Protocol Development Kit Run Time) provides a rich set of logging features that are useful to protocol developers and implementers of the engine and call control libraries. The application may add additional log records to the log file when logging is enabled.

Notes: ***1.*** It is recommended to use logging on an as-needed basis. Logging uses significant resources and can reduce the performance of the Global Call PDKRT call control library. Full logging (debug logging) enabled on many channels can reduce performance to such a degree that time-critical operations are affected and the behavior of a protocol may be altered.

2. The LogView tool is required to view the log file.

The PDKRT call control library provides a service for capturing error and debug information in a log file. Enabling and disabling logging is achieved using the **gc_Start( )** function. Once logging is enabled, the **gc_StartTrace( )** function can be used to enable logging on each individual channel. See Section 8.2.22, "gc_Start( ) and gc_Stop( ) Variances for E1/T1 CAS/R2", on page 103 and Section 8.2.23, "gc_StartTrace( ) Variances for E1/T1 CAS/R2", on page 104 for more information.

The parameters that control the logging mechanism can be set by:

- Populating and using a CCLIB_START_STRUCT. See Section 7.2.2, "Populating and Using a CCLIB_START_STRUCT", on page 74.
- Defining the GC_PDK_START_LOG environment variable. See Section 7.2.3, "Defining the GC_PDK_START_LOG Environment Variable", on page 78.

When both methods are used, the CCLIB_START_STRUCT takes precedence over the GC_PDK_START_LOG environment variable.

Note: Two applications should not use the same log file.

7.2.2 Populating and Using a CCLIB_START_STRUCT

The following code shows an example of how to define a CCLIB_START_STRUCT, populate the fields, and use it to enable logging when issuing the **gc_Start( )** function.

```
GC_START_STRUCT t_GcStart;
CCLIB_START_STRUCT t_PdkStart;
t_PdkStart.cclib_name = "GC_PDKRT_LIB";
t_PdkStart.cclib_data = "filename: pdktest.log;
loglevel: ENABLE_DEBUG;
service: R2MF_ENABLE | CAS_ENABLE;
cachedump: WHEN_FULL | THREAD_ON;
channel: B1C1, B2C2-4;
cachesize: 10;
maxfilesize: 0;
mindiskfree: 20";
t_GcStart.num_cclibs = 1;
t_GcStart.cclib_list = (void *)
    (& t_PdkStart);
int t_result = gc_Start((GC_START_STRUCTP)& t_GcStart);
```

Note: The example above shows all the possible fields in a **cclib_data** string. In practice, you only need to specify the values of fields that are different than the default values.

The length of the filename must be less than 8 characters.

The value of the **cclib_name** field must be GC_PDKRT_LIB and the **cclib_data** field should have the following format:

```
"field name 1 : field value 1; field name 2 : field value 2; ..."
```

where the allowable field names and values are given in Table 17.

Table 17. cclib_data Fields and Values

Field Name	Field Values	Default Value
filename	Log file name	gc_pdk.log
loglevel	See Table 18.	ENABLE_FATAL or 5
service	See Table 19.	ALL_SERVICES
cachedump	See Table 20.	WHEN_FULL or 1
cachesize	Any positive integer	1 (number of records in cache)
channel	See Table 21.	B*C*
maxfilesize	Integer	0 (Megabytes)
mindiskfree	Integer	20 (Megabytes)

The fields can be defined in any sequence. If any field is not defined or defined incorrectly (either in name or value), then the default value is used for logging. The actual values of the fields are posted as the first record of the log file. In this way, when a log file is received, the user knows how logging was configured (that is, which log level and services were enabled, what the cache size and cache dump conditions were when it was generated).

The following examples show how to set the **cclib_data** string:

- The example below shows all the possible fields. In practice, you only have to specify the values of fields that are different than the default values.

```
cclib_data = "filename: pdktest.log;
loglevel: ENABLE_DEBUG;
service: R2MF_ENABLE;
cachedump: WHEN_FULL|THREAD_ON;
channel: B1C1, B2C2-4;
cachesize: 10;
maxfilesize: 0;
mindiskfree: 20"
```

- For simplicity and to avoid errors, use only the values of fields that are different than the default values. For example, to specify a log file name called *mylog.log* that includes all log entries, use the following **cclib_data** string:

```
cclib_data = "filename: mylog.log; loglevel: ENABLE_DEBUG"
```

The following tables show the allowable values for the **loglevel**, **service**, **cachedump**, and **channel** fields respectively. The values of **loglevel**, **service**, and **cachedump** can be numbers or symbols. (If hex format is used, the prefix 0x should be used.) Consequently, before these values are passed to the LOG_INIT, the values must be examined and converted from symbols to numbers, if necessary. The value symbol of **service** and **cachedump** can be a bit mask.

Table 18 shows the valid values for the **loglevel** parameter.

Table 18. Loglevel Parameter Values

loglevel	Valid Value	Description
ENABLE_FATAL (default)	5	Only fatal errors are logged. A fatal error is an error that will make the program run abnormally or will stop the program. For example, in *channelimpl.cpp*, **dx_open( )** returns INVALID_VOICEH. It is expected that an exception will be thrown and the log cache will be dumped to a file if possible.
ENABLE_WARNING	4	All levels above ALERT are logged. An error occurs that may make the program run abnormally. For example, in *channelimpl.cpp*, the new local state is not ChanState_InService while the reason is Wait Call. An exception may be thrown, but log cache will not be dumped to a file automatically.
ENABLE_ALERT	3	All levels above INFO are logged. There is a problem, generally not an error, that the user should know about.
ENABLE_INFO	2	All levels above DEBUG are logged. Important information that the user needs to be aware of is logged. For example, in *channelimpl.cpp*, issuing a **gc_StartTrace( )** and **gc_StopTrace( )** determines if logging for a specific channel is on or off. This kind of information is a level higher than DEBUG.
ENABLE_DEBUG	1	All levels are logged. This gives the most detailed information to help debug protocols or code step-by-step. For example, in *channelimpl.cpp*, a call to any of the GC_PDK_C_XXX functions should be logged at this level. Most routine logging should use this level.
Note: Values are in decimal but can also be specified in hex using a 0x prefix.		

Table 19 shows the valid values for the **service** parameter.

Table 19. Service Parameter Values

service	Valid Value	Description
ALL_SERVICES (default)	0xFFFFFFFF (65535)	All services are enabled.
USRAPP_ENABLE	0x00000001 (1)	Only USRAPP service enabled.
GCAPI_ENABLE	0x00000002 (2)	Only GCAPI service enabled.
GCXLTR_ENABLE	0x00000004 (4)	Only GCXLTR service enabled.
LINEADMIN_ENABLE	0x00000008 (8)	Only LINEADMIN service enabled.
CHANNEL_ENABLE	0x00000010 (16)	Only CHANNEL service enabled.
LOADER_ENABLE	0x00000020 (32)	Only LOADER service enabled.
CALL_ENABLE	0x00000040 (64)	Only CALL service enabled.
R2MF_ENABLE	0x00000080 (128)	Only R2 MF service enabled.
TONE_ENABLE	0x00000100 (256)	Only TONE service enabled.
CAS_ENABLE	0x00000200 (512)	Only CAS service enabled.
TIMER_ENABLE	0x00000400 (1024)	Only TIMER service enabled.
Note: Values prefixed with 0x are hexadecimal values. Decimal values are shown in parentheses.		

Table 19. Service Parameter Values (Continued)

service	Valid Value	Description
SDL_ENABLE	0x00000800 (2048)	Only SDL service enabled.
SRL_ENABLE	0x00001000 (4096)	Only SRL service enabled.
ERRHNDLR_ENABLE	0x00002000 (8192)	Only ERRHNDLR service enabled.
LOGGER_ENABLE	0x00004000 (16384)	Only LOGGER service enabled.
RTCM_ENABLE	0x00008000 (32768)	Only RTCM service enabled.
GCLIB_ENABLE	0x00010000 (65546)	Only GCLIB service enabled.
Note: Values prefixed with 0x are hexadecimal values. Decimal values are shown in parentheses.		

Table 20 shows the valid values for the **cachedump** parameter.

Table 20. Cachedump Parameter Values

cachedump	Valid Value	Description
ON_FATAL	0x0000 (bit 1 = 0)	The cache memory will be dumped to the log file once there is a log record with a FATAL level.
WHEN_FULL (default)	0x0001 (bit 1 = 1)	The cache memory will be dumped to the log file once the log cache is full as determined by the **cachesize** parameter. For example, if **cachesize** is 10, the log cache is dumped to a file when it contains 10 log records.
THREAD_OFF (default)	0x0000 (bit 2 = 0)	The dump operation will be executed by the calling thread.
THREAD_ON	0x0002 (bit 2 = 1)	The dump operation will be executed by a separate cache dumping thread.
Note: Values prefixed with 0x are hexadecimal values.		

Table 21 shows some examples of the **channel** parameter.

Table 21. Sample Channel Parameter Values

Example Value	Boards and Channels Enabled for Logging
B*C* (default)	All boards and all channels
B-1C-1	Only board number = -1 and channel number = -1.
B1C*	All channels on board 1.
B1C-1	Only board 1 level.
B1C1	Channel 1 on board 1.
B1C1-5	Channels 1 to 5 on board 1.
B1C1,20	Channels 1 and 20 on board 1.
B1-4C*	All channels of boards 1 to 4.
B1C2, B2C2,20-22	Channel 2 on board 1, channels 2, 20, 21, and 22 on board 2.

7.2.3 Defining the GC_PDK_START_LOG Environment Variable

The GC_PDK_START_LOG environment variable can also be used to enable and configure logging.

The following examples show how to set the GC_PDK_START_LOG environment variable in Windows®:

- The following is an example of a GC_PDK_START_LOG environment variable definition showing all the possible field values in the environment variable. In practice, you only have to specify the values of fields that are different than the default values.

```
set GC_PDK_START_LOG="filename : pdktest.log;
loglevel: ENABLE_DEBUG; services: ALL_SERVICES;
cachedump : WHEN_FULL | THREAD_ON; channel : B1C1, B2C2-4;
cachesize : 10; maxfilesize : 0; mindiskfree : 20"
```

- For simplicity and to avoid errors, use only the values of fields that are different than the default values. For example, to specify a log file name called *mylog.log* that includes all log entries, use the following GC_PDK_START_LOG environment variable definition:

```
set GC_PDK_START_LOG = "filename: mylog.log; loglevel: ENABLE_DEBUG"
```

  This definition is equivalent to the logging configuration used in Section 7.2.2, "Populating and Using a CCLIB_START_STRUCT", on page 74 and the definition for each field is also the same as described in that section.

The setting of the environment variable to enable PDK logging in Linux is:

```
export GC_PDK_START_LOG="filename:gc_pdk.log;loglevel:ENABLE_DEBUG;
service:ALL_SERVICES;cachedump:WHEN_FULL|THREAD_OFF;cachesize:1;maxfilesize:2"
```

7.3 Debugging Applications That Use ICAPI Protocols

The parameters shown in Table 22 are available in the *icapi.cfg* file as debugging tools. Unless otherwise instructed, these parameters should retain their original settings.

The *icapi.cfg* file is located in the following directory:

- For Linux: $INTEL_DIALOGIC_CFG
- For Windows®: %INTEL_DIALOGIC_CFG%

When logging is enabled, the log file generated is *icapi.log.<pid>*, where pid = the process identification number.

For Linux applications, the log file is generated by compiling the *country.c* file with the symbol DEBUG defined and then setting the parameters $11 and $12 in the *icapi.cfg* file as indicated in the following table. To write additional information directly to the ICAPI log file, use the **rs_log_printf( )** function. This function works like the **fprintf( )** function except that a file descriptor is not used.

For Windows® applications, the log file is generated by setting parameters $11 and $12 in the *icapi.cfg* file as indicated in Table 22.

Table 22. icapi.cfg File Parameters

Parameter	Description
$11	Logging utility (default = 0): • Set to 0 to ignore parameters $12, $13 and $15. • Set to 1 to enable logging, either to the screen (set $13 parameter to 1) or to the *icapi.log.<pid>* file to track all the events that occur at the device selected for monitoring (parameter $12). This setting enables the debug tools associated with the protocol. These tools help to locate the source of a protocol problem. • (Windows® only) Set to 2 to enable logging to a memory buffer and to generate an *icapi.inf* file. The *icapi.inf* file contains the memory address where the debug information is stored. **Note:** Enabling logging is not recommended during normal operation due to the increased host processor loading.
$12	Number of the channel to be monitored (default = 0): • A value of 0 means monitor all opened devices. • A value of -1 means do not monitor any device. • Entering a channel number designates the channel to be monitored.
$13	Echo on screen (default = 0): • Set to 0 to ignore parameter. • Set to 1 to send the debug information to the screen.
$14	Disable DTI Wait Call function (default = 0): • The 0 default value causes the DTI Wait Call firmware function to wait for an incoming call at the board firmware level. • A value of 1 causes the DTI Wait Call firmware function to wait for an incoming call at the ICAPI call control library level. The value selected is protocol-dependent; do not change the default value unless instructed to do so in the documentation for your protocol.
$15	(Linux only) Size of debug memory (default = 1; that is, 1 = 1 event or action in memory) The debug memory saves passed actions or events to a buffer. The built-in debug function does not use this feature. Change this parameter only if you implement your own debug function and you need a larger circular buffer than 1 event or action. • Set to 1 to store one action or event in the buffer. • Set to 0 to ignore feature (default).
$18	Enables cadenced tones, such as ringback and busy, to be played using the firmware rather than using host-based function calls such as **dx_playtone( )** and **sleep( )**. • Set to 0 to disable firmware cadence tones (default). • Set to 1 to enable firmware cadence tones.

Any unspecified parameter defaults to 0. If parameters $13 and $15 are set to 0, they are ignored.

Parameters $16 and $17 (not shown in Table 22) are for backwards compatibility only and should not be changed.

E1/T1 CAS/R2-Specific Function Information 8

This chapter describes the Dialogic® Global Call API functions that have additional functionality or perform differently when used with E1/T1 CAS/R2 technology. The function descriptions are presented alphabetically and contain information that is specific to E1/T1 CAS/R2 applications. Generic function description information (that is, information that is not technology-specific) is provided in the *Dialogic® Global Call API Library Reference*.

Topics in this chapter include:

- Dialogic® Global Call Functions Supported by E1/T1 CAS/R2. 81
- Dialogic® Global Call Function Variances for E1/T1 CAS/R2. 88

8.1 Dialogic® Global Call Functions Supported by E1/T1 CAS/R2

The following is a list of the Dialogic® Global Call functions that indicates the level of support when used with E1/T1 CAS/R2 technology. The list indicates whether the function is supported, not supported, or supported with variances.

gc_AcceptCall( )
: Supported with variances described in Section 8.2.1, "gc_AcceptCall( ) Variances for E1/T1 CAS/R2", on page 89.

gc_AcceptInitXfer( )
: Not supported.

gc_AcceptModifyCall( )
: Not supported.

gc_AcceptXfer( )
: Not supported.

gc_AlarmName( )
: Supported.

gc_AlarmNumber( )
: Supported.

gc_AlarmNumberToName( )
: Supported.

gc_AlarmSourceObjectID( )
: Supported.

gc_AlarmSourceObjectIDToName( )
: Supported.

gc_AlarmSourceObjectName( )
Supported.

gc_AlarmSourceObjectNameToID( )
Supported.

gc_AnswerCall( )
Supported with variances described in Section 8.2.2, "gc_AnswerCall( ) Variances for E1/T1 CAS/R2", on page 89.

gc_Attach( ) (deprecated)
Supported.

gc_AttachResource( )
Supported.

gc_BlindTransfer( )
Supported with variances described in Section 8.2.3, "gc_BlindTransfer( ) Variances for E1/T1 CAS/R2", on page 91.

gc_CallAck( )
For Dialogic® Springware Boards: Supported with variances described in Section 8.2.4, "gc_CallAck( ) Variances for E1/T1 CAS/R2", on page 92. For Dialogic® DM3 Boards: Not supported.

gc_CallProgress( )
Not supported.

gc_CCLibIDToName( )
Supported.

gc_CCLibNameToID( )
Supported.

gc_CCLibStatus( ) (deprecated)
Supported.

gc_CCLibStatusAll( ) (deprecated)
Supported.

gc_CCLibStatusEx( )
Supported.

gc_Close( )
Supported with variances described in Section 8.2.5, "gc_Close( ) Variances for E1/T1 CAS/R2", on page 92.

gc_CompleteTransfer( )
Supported with variances described in Section 8.2.6, "gc_CompleteTransfer( ) Variances for E1/T1 CAS/R2", on page 93.

gc_CRN2LineDev( )
Supported.

gc_Detach( )
Supported with variances described in Section 8.2.7, "gc_Detach( ) Variances for E1/T1 CAS/R2", on page 93.

gc_DropCall( )
Supported with variances described in Section 8.2.8, "gc_DropCall( ) Variances for E1/T1 CAS/R2", on page 93.

gc_ErrorInfo( )
Supported.

gc_ErrorValue( ) (deprecated)
Supported.

gc_Extension( )
Supported with variances described in Section 8.2.9, "gc_Extension( ) Variances for E1/T1 CAS/R2", on page 94.

gc_GetAlarmConfiguration( )
Supported.

gc_GetAlarmFlow( )
Supported.

gc_GetAlarmParm( )
For Dialogic® Springware Boards: Supported. For Dialogic® DM3 Boards: Not supported.

gc_GetAlarmSourceObjectList( )
Supported.

gc_GetAlarmSourceObjectNetworkID( )
Supported.

gc_GetANI( ) (deprecated)
Supported.

gc_GetBilling( )
Not supported.

gc_GetCallInfo( )
Supported with variances described in Section 8.2.10, "gc_GetCallInfo( ) Variances for E1/T1 CAS/R2", on page 94.

gc_GetCallProgressParm( )
For Dialogic® Springware Boards: Supported (PDKRT only). For Dialogic® DM3 Boards: Not supported.

gc_GetCallState( )
Supported.

gc_GetConfigData( )
For Dialogic® Springware Boards: Supported (PDKRT only). For Dialogic® DM3 Boards: Not supported.

gc_GetCRN( )
Supported.

gc_GetCTInfo( )
For Dialogic® Springware Boards: Supported (PDKRT only). For Dialogic® DM3 Boards: Supported.

gc_GetDNIS( ) (deprecated)
Supported.

gc_GetFrame( )
Not supported.

gc_GetInfoElem( )
Not supported.

gc_GetLineDev( )
Supported.

gc_GetLineDevState( )
For Dialogic® Springware Boards: Not supported. For Dialogic® DM3 Boards: Supported.

gc_GetMetaEvent( )
Supported.

gc_GetMetaEventEx( )
Supported (Windows® extended asynchronous mode only).

gc_GetNetCRV( )
Not supported.

gc_GetNetworkH( ) (deprecated)
Supported.

gc_GetParm( )
Supported with variances described in Section 8.2.11, "gc_GetParm( ) Variances for E1/T1 CAS/R2", on page 96.

gc_GetResourceH( )
Supported.

gc_GetSigInfo( )
Not supported.

gc_GetUserInfo( )
Not supported.

gc_GetUsrAttr( )
Supported.

gc_GetVer( )
For Dialogic® Springware Boards: Supported. For Dialogic® DM3 Boards: Not supported.

gc_GetVoiceH( ) (deprecated)
Supported.

gc_GetXmitSlot( )
For Dialogic® Springware Boards: Supported (PDKRT only). For Dialogic® DM3 Boards: Supported.

gc_HoldACK( )
Not supported.

gc_HoldCall( )
Supported with variances described in Section 8.2.12, "gc_HoldCall( ) Variances for E1/T1 CAS/R2", on page 97.

gc_HoldRej( )
Not supported.

gc_InitXfer( )
Not supported.

gc_InvokeXfer( )
Not supported.

gc_LinedevToCCLIBID( )
Supported.

gc_Listen( )
For Dialogic® Springware Boards: Supported (PDKRT only). For Dialogic® DM3 Boards: Supported.

gc_LoadDxParm( )
For Dialogic® Springware Boards: Supported (PDKRT only). For Dialogic® DM3 Boards: Not supported.

gc_MakeCall( )
Supported with variances described in Section 8.2.13, "gc_MakeCall( ) Variances for E1/T1 CAS/R2", on page 97.

gc_Open( ) (deprecated)
Supported.

gc_OpenEx( )
Supported with variances described in Section 8.2.14, "gc_OpenEx( ) Variances for E1/T1 CAS/R2", on page 99.

gc_QueryConfigData( )
For Dialogic® Springware Boards: Supported (PDKRT only). For Dialogic® DM3 Boards: Not supported.

gc_RejectInitXfer( )
Not supported.

gc_RejectModifyCall( )
Not supported.

gc_RejectXfer( )
Not supported.

gc_ReleaseCall( ) (deprecated)
Supported.

gc_ReleaseCallEx( )
Supported.

gc_ReqANI( )
Not supported.

gc_ReqModifyCall( )
Not supported.

gc_ReqMoreInfo( )
For Dialogic® Springware Boards: Supported (PDKRT only). For Dialogic® DM3 Boards: Not supported.

gc_ReqService( )
Not supported.

gc_ResetLineDev( )
For Dialogic® Springware Boards: Supported with variances described in Section 8.2.15, "gc_ResetLineDev( ) Variances for E1/T1 CAS/R2", on page 101. For Dialogic® DM3 Boards: Supported.

gc_RespService( )
Not supported.

gc_ResultInfo( )
Supported.

gc_ResultMsg( ) (deprecated)
Supported.

gc_ResultValue( ) (deprecated)
Supported.

gc_RetrieveAck( )
Not supported.

gc_RetrieveCall( )
Supported with variances described in Section 8.2.16, "gc_RetrieveCall( ) Variances for E1/T1 CAS/R2", on page 101.

gc_RetrieveRej( )
Not supported.

gc_SendMoreInfo( )
For Dialogic® Springware Boards: Supported (PDKRT only). For Dialogic® DM3 Boards: Not supported.

gc_SetAlarmConfiguration( )
Supported.

gc_SetAlarmFlow( )
Supported.

gc_SetAlarmNotifyAll( )
Supported.

gc_SetAlarmParm( )
For Dialogic® Springware Boards: Supported. For Dialogic® DM3 Boards: Not supported.

gc_SetAuthenticationInfo( )
Not supported.

gc_SetBilling( )
For Dialogic® Springware Boards: Supported with variances described in Section 8.2.17, "gc_SetBilling( ) Variances for E1/T1 CAS/R2", on page 101. For Dialogic® DM3 Boards: Not supported.

gc_SetCallingNum( ) (deprecated)
Supported.

gc_SetCallProgressParm( )
For Dialogic® Springware Boards: Supported (PDKRT only). For Dialogic® DM3 Boards: Not supported.

gc_SetChanState( )
Supported with variances described in Section 8.2.18, "gc_SetChanState( ) Variances for E1/T1 CAS/R2", on page 102.

gc_SetConfigData( )
For Dialogic® Springware Boards: Supported (PDKRT only). For Dialogic® DM3 Boards: Supported.

gc_SetEvtMsk( ) (deprecated)
For Dialogic® Springware Boards: Supported with variances described in Section 8.2.19, "gc_SetEvtMsk( ) Variances for E1/T1 CAS/R2", on page 102. For Dialogic® DM3 Boards: Supported.

gc_SetInfoElem( )
Not supported.

gc_SetParm( )
Supported with variances described in Section 8.2.20, "gc_SetParm( ) Variances for E1/T1 CAS/R2", on page 102.

gc_SetupTransfer( )
Supported with variances described Section 8.2.21, "gc_SetupTransfer( ) Variances for E1/T1 CAS/R2", on page 103.

gc_SetUserInfo( )
Not supported.

gc_SetUsrAttr( )
Supported.

gc_SipAck( )
Not supported.

gc_SndFrame( )
Not supported.

gc_SndMsg( )
Not supported.

gc_Start( )
For Dialogic® Springware Boards: Supported with variances described in Section 8.2.22, "gc_Start( ) and gc_Stop( ) Variances for E1/T1 CAS/R2", on page 103. For Dialogic® DM3 Boards: Supported.

gc_StartTrace( )
For Dialogic® Springware Boards: Supported with variances described in Section 8.2.23, "gc_StartTrace( ) Variances for E1/T1 CAS/R2", on page 104. For Dialogic® DM3 Boards: Not supported.

gc_Stop( )
For Dialogic® Springware Boards: Supported with variances described in Section 8.2.22, "gc_Start( ) and gc_Stop( ) Variances for E1/T1 CAS/R2", on page 103. For Dialogic® DM3 Boards: Supported.

gc_StopTrace( )
For Dialogic® Springware Boards: Supported (PDKRT only). For Dialogic® DM3 Boards: Not supported.

gc_StopTransmitAlarms( )
Supported.

gc_SwapHold( )
Supported with variances described in Section 8.2.24, "gc_SwapHold( ) Variances for E1/T1 CAS/R2", on page 104.

gc_TransmitAlarms( )
Supported.

gc_UnListen( )
For Dialogic® Springware Boards: Supported (PDKRT only). For Dialogic® DM3 Boards: Supported.

gc_util_copy_parm_blk( )
Supported.

gc_util_delete_parm_blk( )
Supported.

gc_util_find_parm( )
Supported.

gc_util_find_parm_ex( )
Supported.

gc_util_insert_parm_ref( )
Supported.

gc_util_insert_parm_ref_ex( )
Supported.

gc_util_insert_parm_val( )
Supported.

gc_util_next_parm( )
Supported.

gc_util_next_parm_ex( )
Supported.

gc_WaitCall( )
Supported.

8.2 Dialogic® Global Call Function Variances for E1/T1 CAS/R2

The Dialogic® Global Call function variances that apply when using E1/T1 CAS/R2 technology are described in the following sections. See the *Dialogic® Global Call API Library Reference* for generic (technology-independent) descriptions of the Global Call API functions.

- gc_AcceptCall( ) Variances for E1/T1 CAS/R2
- gc_AnswerCall( ) Variances for E1/T1 CAS/R2
- gc_BlindTransfer( ) Variances for E1/T1 CAS/R2

- gc_CallAck( ) Variances for E1/T1 CAS/R2
- gc_Close( ) Variances for E1/T1 CAS/R2
- gc_CompleteTransfer( ) Variances for E1/T1 CAS/R2
- gc_Detach( ) Variances for E1/T1 CAS/R2
- gc_DropCall( ) Variances for E1/T1 CAS/R2
- gc_Extension( ) Variances for E1/T1 CAS/R2
- gc_GetCallInfo( ) Variances for E1/T1 CAS/R2
- gc_GetParm( ) Variances for E1/T1 CAS/R2
- gc_HoldCall( ) Variances for E1/T1 CAS/R2
- gc_MakeCall( ) Variances for E1/T1 CAS/R2
- gc_OpenEx( ) Variances for E1/T1 CAS/R2
- gc_ResetLineDev( ) Variances for E1/T1 CAS/R2
- gc_RetrieveCall( ) Variances for E1/T1 CAS/R2
- gc_SetBilling( ) Variances for E1/T1 CAS/R2
- gc_SetChanState( ) Variances for E1/T1 CAS/R2
- gc_SetEvtMsk( ) Variances for E1/T1 CAS/R2
- gc_SetParm( ) Variances for E1/T1 CAS/R2
- gc_SetupTransfer( ) Variances for E1/T1 CAS/R2
- gc_Start( ) and gc_Stop( ) Variances for E1/T1 CAS/R2
- gc_StartTrace( ) Variances for E1/T1 CAS/R2
- gc_SwapHold( ) Variances for E1/T1 CAS/R2

8.2.1 gc_AcceptCall( ) Variances for E1/T1 CAS/R2

The **gc_AcceptCall( )** function optionally responds to an inbound call request by providing an indication to the remote end that a call was received but not yet answered. This function causes ringback to be generated.

The **gc_AcceptCall( )** function uses the **rings** parameter to specify the number of rings to wait before terminating the function, that is, before the Dialogic® Global Call API sends the GCEV_ACCEPT event to the application.

- For PDK protocols, if the **rings** parameter is set to 0, the value of the **PSL_AcceptCallDefaultNumOfRings** parameter in the country dependent parameters (.cdp) file is used.
- For ICAPI protocols (Dialogic® Springware Boards only), if the **rings** parameter is set to 0, the value specified in parameter $9 of the country dependent parameters (.cdp) file is used.

8.2.2 gc_AnswerCall( ) Variances for E1/T1 CAS/R2

gc_AnswerCall( ) function variances for E1/T1 CAS/R2 are discussed in the following topics:

- Use of the rings Parameter

- Run-Time Control of Double Answer

8.2.2.1 Use of the rings Parameter

The **gc_AnswerCall( )** function indicates to the remote end that the connection is established (call has been answered). The **rings** parameter specifies the number of rings to wait before terminating the **gc_AnswerCall( )** function, that is, before answering the call.

- For PDK protocols, if the **rings** parameter is set to 0, the value of the **PSL_AnswerCallDefaultNumOfRings** parameter in the country dependent parameters (.cdp) file is used.
- For ICAPI protocols (Dialogic® Springware Boards only), if the **rings** parameter is set to 0, the value specified in parameter $9 of the country dependent parameters (.cdp) file is used.

8.2.2.2 Run-Time Control of Double Answer

Double answer is a feature supported in some protocols for blocking collect calls.

Double answer signaling can be *statically* enabled or disabled by setting the **CDP_DOUBLE_ANSWER_FLAG** parameter in the CDP file. This setting applies to all the calls on the channels and cannot be controlled on a call-by-call basis.

The **gc_AnswerCall( )** function provides a method of rejecting collect calls on a call-by-call basis. The following Dialogic® Boards currently support this feature:

- Dialogic® DM/V-A Media Boards
- Dialogic® DM/V-B Media Boards
- Dialogic® D/300JCT-E1 Media Boards
- Dialogic® D/600JCT-1E1 Media Boards
- Dialogic® D/600JCT-2E1 Media Boards

Double answer can be triggered on a call-by-call basis by issuing **gc_AnswerCall( )** with the number of rings ORed with the **GC_DBL_ANSWER** define (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 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.

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);
}
```

8.2.3 gc_BlindTransfer( ) Variances for E1/T1 CAS/R2

The **gc_BlindTransfer**( ) function is only valid for applications using PDK protocols that support call hold and transfer. Check the **sys_features** parameter in the .cdp file for a value of Feature_Transfer. See the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for more information.

8.2.4 gc_CallAck( ) Variances for E1/T1 CAS/R2

Note: The variances described in this section apply when using Dialogic® Springware Boards only. The **gc_CallAck( )** function is not supported when using Dialogic® DM3 Boards.

The **gc_CallAck( )** function may be called before issuing a **gc_AcceptCall( )** or a **gc_AnswerCall( )** function to indicate to the network if more information is desired before completing the call. This function is used to request additional DDI digits from the network. After using this function, call the **gc_GetCallInfo( )** function to retrieve the digits. The **gc_GetCallInfo( )** function will return all DDI digits collected from the network (including both the digits already received and those returned by the network in response to the **gc_CallAck( )** function call). Using the **gc_CallAck( )** function for this service is described in the *Dialogic® Global Call API Library Reference*.

The valid range of values for the **gc_CallAck( )** function **info_len** field is from 1 to GCDG_MAXDIGIT. If more than GCDG_MAXDIGIT digits are required, or if an unknown number of digits is to be requested, set the **info_len** field to GCDG_NDIGIT.

The value GCDG_PARTIAL may be ORed with the number of digits field if the application needs to call the **gc_CallAck( )** function again for this call (that is, if the application needs additional DDI digits before accepting or rejecting the call).

8.2.5 gc_Close( ) Variances for E1/T1 CAS/R2

The **gc_Close( )** function only affects the link between the calling process and the device. For CAS protocols, if a voice resource is currently assigned to the specified line device, the voice resource will be closed. To keep the voice resource open for other operations, use the **gc_Detach( )** function to detach the voice resource from the line device before issuing the **gc_Close( )** function.

Functionality of **gc_Close( )** is different for Dialogic® Springware Boards and Dialogic® DM3 Boards with regards to stopping the protocol.

Dialogic® Springware Board-specific variances

Dialogic® Springware Boards stop the protocol after **gc_Close( )**.

Dialogic® DM3 Board-specific variances

Dialogic® DM3 Boards do not stop the protocol after **gc_Close( )**.

On Dialogic DM3 Boards, **gc_Close( )** typically sets the protocol out of service; the protocol is not stopped until the board is stopped. Therefore, when a Dialogic DM3 Board uses a protocol that includes the **CDP_ProtocolStopsOffhook** parameter, which determines the state of the hook switch signaling (on-hook or off-hook) when the protocol stops after **gc_Close( )**, this parameter has no effect.

8.2.6 gc_CompleteTransfer( ) Variances for E1/T1 CAS/R2

The **gc_CompleteTransfer( )** function is only valid for applications using PDK protocols that support call hold and transfer. Check the **sys_features** parameter in the .cdp file for a value of Feature_Transfer. See the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for more information.

8.2.7 gc_Detach( ) Variances for E1/T1 CAS/R2

The **gc_Detach( )** function logically disconnects a voice channel from a line device. It is the responsibility of the application to make sure that there is a voice resource available while the **gc_WaitCall( )** function is active and that the current Global Call call state is Null or Idle. Furthermore, the **gc_Detach( )** function can only be called in the Null, Idle, or Connected states.

8.2.8 gc_DropCall( ) Variances for E1/T1 CAS/R2

The **gc_DropCall( )** function supports the following values for its **cause** parameter:

GC_CALL_REJECTED
: Call is not accepted

GC_NETWORK_CONGESTION
: Cannot establish connection due to volume of traffic on network

GC_NORMAL_CLEARING
: Normal end of call

GC_SEND_SIT
: Sends a special information tone

GC_UNASSIGNED_NUMBER
: Invalid called party number

GC_USER_BUSY
: Called party is busy

Note: You must use the **dx_stopch( )** function to terminate any application-initiated voice functions, such as **dx_play( )** or **dx_record( )**, before calling **gc_DropCall( )**.

Some protocols do not support all **gc_DropCall( )** causes for dropping a call. Any unsupported cause(s) is automatically mapped to the most appropriate cause. This approach facilitates developing protocol independent applications.

From the Accepted state, some protocols do not support a forced release of the line; that is, issuing a **gc_DropCall( )** function after a **gc_AcceptCall( )** function. Refer to the Protocol Limitations section in the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for your protocol. If a forced release is attempted, the function fails and an error is returned. To recover, the application should issue a **gc_AnswerCall( )** function followed by **gc_DropCall( )** and **gc_ReleaseCall( )** functions. However, anytime a GCEV_DISCONNECTED event is received in the Accepted state, the **gc_DropCall( )** function can be issued.

After the **gc_AnswerCall( )** function is issued, the application must wait for a GCEV_ANSWER event. Otherwise the **gc_DropCall( )** function is ignored, no error is returned, and no drop call action is taken.

When using ICAPI protocols (Dialogic® Springware Boards only), the **gc_DropCall( )** function occasionally results in the generation of the GCEV_DROPCALL event followed by a GCEV_BLOCKED event. The generation of the GCEV_BLOCKED event is most likely if the **gc_DropCall( )** function is issued before the call is connected. The reason for the GCEV_BLOCKED event is that the remote side does not recognize the disconnection in a timely manner. When the GCEV_BLOCKED event occurs, call-related Global Call functions should not be issued until a GCEV_UNBLOCKED event is detected on the respective device.

In some protocols, a **gc_DropCall( )** command on a call in the Accepted state requires a momentary transition to the Connected state. This may result in a charge being registered for the call.

8.2.9 gc_Extension( ) Variances for E1/T1 CAS/R2

Dialogic® DM3 Board-specific variances

The **gc_Extension( )** function can be used to access the functionality of the Direct Signaling protocol. The Direct Signaling protocol is not a call control protocol; it is used strictly to give applications direct control over the signaling patterns on a line, as a means to allow the application to implement its own protocols. The Direct Signaling protocol allows the application to generate and detect signaling patterns. Applications can call the **gc_Extension( )** function to generate up to 11 distinct CAS patterns, and through the GCEV_EXTENSION event, be notified when one of the patterns is detected by the protocol. For details about the Direct Signaling protocol and the **gc_Extension( )** function, see the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide*.

8.2.10 gc_GetCallInfo( ) Variances for E1/T1 CAS/R2

gc_GetCallInfo( ) function variances for E1/T1 CAS/R2 are discussed in the following topics:

- Use of the CONNECT_TYPE info_id Parameter
- Use of the CALLINFOTYPE info_id Parameter

8.2.10.1 Use of the CONNECT_TYPE info_id Parameter

For E1 CAS and T1 robbed bit protocols that support enhanced call analysis (call progress), the **gc_GetCallInfo( )** CONNECT_TYPE **info_id** parameter contains the type of connection as returned by the function. These connection types are:

GCCT_CAD
: Connection due to cadence break

GCCT_PVD
: Connection due to voice detection

GCCT_PAMD
: Connection due to answering machine detection

GCCT_FAX
: Connection due to fax machine detection

GCCT_NA
: Connection type is not available

PDK protocols provide support for enhanced call analysis.

For protocols that do not support enhanced call progress analysis, the **gc_GetCallInfo( )** function with the CONNECT_TYPE parameter specified will return a CONNECT_TYPE value of GCCT_NA (not available).

8.2.10.2 Use of the CALLINFOTYPE info_id Parameter

For E1 CAS protocols that support the CALLINFOTYPE **info_id** parameter, a call information string containing either CHARGE, NO CHARGE, or CHARGE WITH CLEARING FROM INBOUND is returned by the parameter; check your protocol in the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for applicability. This allows the application to know which billing type was received when the lines are available for call establishment. B tones are sent to indicate whether the line is available, and also to indicate the type of billing for the call.

The following Dialogic® Boards currently support this feature:

- Dialogic® DM/V-A Media Boards
- Dialogic® DM/V-B Media Boards
- Dialogic® D/300JCT-E1 Media Boards
- Dialogic® D/600JCT-1E1 Media Boards
- Dialogic® D/600JCT-2E1 Media Boards

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. Table 23 shows the mapping of group B tones to billing type string returned.

Table 23. gc_GetCallInfo( ) Billing Type Strings Returned

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), Table 24 shows the mappings that are used for assigning cause values to the GCEV_DISCONNECT event.

Table 24. gc_GetCallInfo( ) Cause Values for GCEV_DISCONNECT

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 Dialogic® DM3 Boards: GCRV_UNALLOCATED For Dialogic® Springware Boards: GCRV_NOT_INSERVICE	For Dialogic® DM3 Boards: "Number not allocated" For Dialogic® Springware Boards: "Number not in service"
GrpB - SIT	For Dialogic® DM3 Boards: GCRV_SIT_UNKNOWN For Dialogic® Springware Boards: GCRV_CEPT	For Dialogic® DM3 Boards: "Unknown SIT detected" For Dialogic® 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".

8.2.11 gc_GetParm( ) Variances for E1/T1 CAS/R2

The **gc_GetParm( )** function retrieves the value of the specified parameter for a line device.

Dialogic® Springware Board-specific variances

In addition to the GCPR_CALLINGPARTY parameter, which is common across all technologies and documented in the *Dialogic® Global Call API Library Reference*, the following parameters are supported:

- GCPR_LOADTONES
- GCPR_MEDIADETECT

See Section 8.2.20, "gc_SetParm( ) Variances for E1/T1 CAS/R2", on page 102 for more information on the meaning of these parameters.

Dialogic® DM3 Board-specific variances

In addition to the GCPR_CALLINGPARTY parameter, which is common across all technologies and documented in the *Dialogic® Global Call API Library Reference*, the following parameters are supported:

- GCPR_CALLPROGRESS
- GCPR_MEDIADETECT
- GCPR_MINDIGITS

See Section 8.2.20, “gc_SetParm( ) Variances for E1/T1 CAS/R2”, on page 102 for more information on the meaning of these parameters.

8.2.12 gc_HoldCall( ) Variances for E1/T1 CAS/R2

The **gc_HoldCall( )** function is only valid for applications using PDK protocols that support call hold and transfer. Check the **sys_features** parameter in the .cdp file for a value of Feature_Hold. See the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for more information.

8.2.13 gc_MakeCall( ) Variances for E1/T1 CAS/R2

gc_MakeCall( ) function variances for E1/T1 CAS/R2 are discussed in the following topics:

- Use of the timeout Parameter
- Other gc_MakeCall( ) Considerations

8.2.13.1 Use of the timeout Parameter

When using E1 CAS or T1 robbed bit line devices, the **timeout** parameter in the **gc_MakeCall( )** function is supported when the **mode** parameter is set to either EV_SYNC or EV_ASYNC.

For ICAPI protocols (Dialogic® Springware Boards only), when the **mode** parameter is set to EV_ASYNC, the **timeout** parameter overrides the time-out parameter ($13) value and the outbound number of ringback tones parameter ($1) in most protocol country dependent parameters (.cdp) files.

- If the **timeout** parameter is set to 0, then the time-out and ringback parameters in the .cdp file are used to set the time-out conditions.
- If the **timeout** parameter is set to a value larger than a protocol time-out value, a protocol time-out may occur first, which will cause the **gc_MakeCall( )** function to fail. The protocol time-out is configured in the .cdp file.
- If the **timeout** value is reached before the remote end answers the call, the application is notified of this condition and should respond as described in the **gc_MakeCall( )** function description in the *Dialogic® Global Call API Library Reference*.
- If all **timeout** values are set to 0, no time-out condition will apply.

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.

8.2.13.2 Other gc_MakeCall( ) Considerations

If your T1 robbed bit circuit is provisioned for Feature Group A, your application should call the **gc_MakeCall( )** function with a null dial string.

When using R2 protocols, a “#” in the dial string is not supported.

If a protocol error occurs during dialing and the default call progress enabled governs, then an error code or an event is returned as described in the **gc_MakeCall( )** function description in the *Dialogic® Global Call API Library Reference*. If call progress is disabled and a protocol error occurs during dialing, then a GCRV_BUSY result value or an EGC_BUSY error is returned.

For drop and insert applications, call progress is typically disabled to enable the application to complete the dialing sequence, listen for voice or ringback on the line, and:

- if ringback is detected, to transition to the Alerting state
- if voice is detected, to transition to the Connected (call answered) state and to implement voice cut-through immediately

This methodology enables the application to pass signaling from the remote end (outbound line) to the caller on the inbound line. If call progress is not disabled, then the GCEV_ALERTING event and the GCEV_ANSWERED event may be received from the outbound line for an unacceptable amount of time after the dialing sequence was completed. During this period of time, the caller could misinterpret the silence on the line as a disconnect or a failure, and then hang up and redial. For further information, see the tips for programming drop and insert applications in the *Dialogic® Global Call API Programming Guide*.

In an E1 environment, the GCEV_ALERTING event is generated when the equivalent of ringback is recognized. For almost all E1 protocols, this is a required part of the protocol, so E1 applications will receive the GCEV_ALERTING event by default.

In a T1 environment, the GCEV_ALERTING event is generated when the ringback is recognized. However, not all inbound applications will generate a ringback tone; for example, the PDK US MF protocol has disabled ringback tone generation by default to minimize call setup time. (Detecting the ringback tone can take several tenths of a second.) If the outbound application does not wish to use the detection of the ringback tone to generate the GCEV_ALERTING event, the **CDP_OUT_Send_Alerting_After_Dialing** parameter in the *pdk_us_mf_io.cdp* file should be set to 1 (default is 0). That way, if call progress is enabled, GCEV_ALERTING is sent after dialing is initiated rather than when ringback is detected.

Since GCEV_ALERTING is an optional event triggered by the inbound side, all applications must be able to handle not receiving the GCEV_ALERTING event.

When the **gc_MakeCall( )** function sets up a call, the default is to enable call analysis (call progress). To change the enabled call progress default when making a call on Dialogic® Springware Boards, use **PDK_MAKECALL_BLK** for PDK protocols and **IC_MAKECALL_BLK** for ICAPI protocols as discussed in Chapter 9, “E1/T1 CAS/R2-Specific Data Structures”. (These structures do not apply to Dialogic® DM3 Boards, which use **gc_SetParm( )** parameters to change call progress configuration as discussed in Section 4.1.1, “Call Analysis with Dialogic® DM3 Boards”, on page 26.)

8.2.14 gc_OpenEx( ) Variances for E1/T1 CAS/R2

The **gc_OpenEx( )** function is used to open both network board and channel (time slot) devices. This generic call control function initializes the specified time slot on the specified trunk. A line device ID will be returned to the application. The E1 or T1 feature of this function specifies the voice device as part of the **devicename** parameter.

gc_OpenEx( ) function variances for E1/T1 CAS/R2 are discussed in the following topics:

- Conventions for Specifying the devicename Parameter
- Other gc_OpenEx( ) Considerations
- Handling GCEV_BLOCKED and GCEV_UNBLOCKED Events

8.2.14.1 Conventions for Specifying the devicename Parameter

A device is specified by the **devicename** parameter using a format that includes protocol specific information.

The format for the fields used to specify this parameter is:

```
:N_<network_device_name>:P_<protocol_name>:V_<voice_channel_name>
```

The prefixes (N_, P_, and V_) are used for parsing purposes. These fields may appear in any order. The fields within the **devicename** parameter must each begin with a colon.

The conventions described below allow the Dialogic® Global Call API to map subsequent calls made on specific line devices or CRNs to interface-specific libraries.

<network_device_name>

This field is required. It may be a board name or a time slot name:

- If <network_device_name> is a board name, use the format: dtiB<number of board>.
- If <network_device_name> is a time slot name, use the format: dtiB<number of board>T<number of time slot>.

<protocol_name>

This field is required on Dialogic® Springware Boards. It specifies the protocol to use. Use the root file name of the country dependent parameters (.cdp) file.

On Dialogic® DM3 Boards, the protocol is determined at board initialization time and not when a Global Call device is opened. For compatibility, the <protocol_name> field may be specified, but it is not used.

<voice_channel_name>

This field is optional depending on your application (see Section 4.5, "Resource Allocation and Routing", on page 45). It specifies the name of the voice channel to be associated with the device being opened. Use the following format: dxxxB<virtual board number>C<channel number>.

Note: Attachment to different types of Dialogic® DM3 voice devices is dependent on the protocol downloaded. For example, if one board has ISDN for protocols and another has T1 CAS, the T1 CAS network devices cannot be attached to the voice devices on the ISDN board. See the *Dialogic® Global Call API Programming Guide* for further information.

8.2.14.2 Other gc_OpenEx( ) Considerations

For E1 CAS or T1 robbed bit applications, always specify a network resource (board or time slot level) and a protocol. A voice resource (<voice_channel_name>) may also be specified for E1 CAS or T1 robbed bit operations. When a voice resource is specified, Global Call automatically opens the voice device and internally attaches the voice device to the line device.

When using the CT Bus and a voice resource is specified, the **gc_OpenEx( )** function routes the voice and network resources together.

When the voice resource is not specified, the application must perform these functions (open device, route, attach); see Section 4.5, "Resource Allocation and Routing", on page 45 for details.

When a network resource is specified, the **gc_OpenEx( )** function internally issues a **dt_open( )** function. Likewise, when a voice resource is specified, the **gc_OpenEx( )** function internally issues a **dx_open( )** function. The corresponding network or voice device handle may be retrieved using the **gc_GetResourceH( )** function. These lower level device handles may be useful for routing or for playing or recording a file.

If a **gc_OpenEx( )** function fails with an error value of EGC_DXOPEN, then the internally issued **dx_open( )** function failed. If a **gc_OpenEx( )** function fails with an error value of EGC_DTOPEN, then the internally issued **dt_open( )** function failed.

8.2.14.3 Handling GCEV_BLOCKED and GCEV_UNBLOCKED Events

At the firmware level, when using Dialogic® Springware Boards, the line is considered **unblocked** until otherwise informed (that is, some event occurs to change the state). From the Global Call perspective, the line is considered **blocked** until otherwise informed. To reconcile this difference in behavior, the Global Call Software generates the required GCEV_UNBLOCKED event as part of the **gc_OpenEx( )** functionality with Dialogic Springware Boards.

When using Dialogic Springware Boards, if a blocking alarm exists on the line when an application tries to open a device, the **gc_OpenEx( )** function will complete, generating the GCEV_UNBLOCKED event, before the firmware detects that the alarm exists, which would trigger the generation of a GCEV_BLOCKED event. This means that the application temporarily sees a GCEV_UNBLOCKED event even though an alarm exists on the line. The application must be capable of handling a GCEV_BLOCKED event at any time, even milliseconds after a GCEV_UNBLOCKED event.

8.2.15 gc_ResetLineDev( ) Variances for E1/T1 CAS/R2

Dialogic® Springware Board-specific variances

For applications that use PDK protocols, the **gc_ResetLineDev( )** function cannot be called while there is an alarm on the line.

For applications that use ICAPI protocols, the **gc_ResetLineDev( )** function is not supported in synchronous mode. If the application calls **gc_ResetLineDev( )** on a line device in synchronous mode (that is, with the mode parameter set to EV_SYNC), the function fails silently.

Dialogic® DM3 Board-specific variances

There are no restrictions on using **gc_ResetLineDev( )** with Dialogic® DM3 Boards.

8.2.16 gc_RetrieveCall( ) Variances for E1/T1 CAS/R2

The **gc_RetrieveCall( )** function is only valid for applications using PDK protocols that support call hold and transfer. Check the **sys_features** parameter in the .cdp file for a value of Feature_Hold. See the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for more information.

8.2.17 gc_SetBilling( ) Variances for E1/T1 CAS/R2

Note: The variances described in this section apply when using Dialogic® Springware Boards only. The **gc_SetBilling( )** function is not supported when using Dialogic® DM3 Boards.

On Dialogic Springware Boards, the **gc_SetBilling( )** function sets different billing rates on a per call basis. For example:

- To charge the call, use **gc_SetBilling(crn, GCR_CHARGE, NULL, EV_SYNC)**.
- To select no-charge for the call, use **gc_SetBilling(crn, GCR_NOCHARGE, NULL, EV_SYNC)**.

The **gc_SetBilling( )** function is called after the GCEV_OFFERED event arrives and before issuing a **gc_AcceptCall( )** or **gc_AnswerCall( )** function.

Not all protocols support this feature; see the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for protocol specific limitations.

The **mode** parameter must be set to EV_SYNC. Asynchronous mode (EV_ASYNC) is not supported for this function.

8.2.18 gc_SetChanState( ) Variances for E1/T1 CAS/R2

The GCLS_INSERVICE and GCLS_OUT_OF_SERVICE states are the only valid service states that can be used to set the state of a line in an E1 CAS or T1 robbed bit environment.

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.

8.2.19 gc_SetEvtMsk( ) Variances for E1/T1 CAS/R2

Dialogic® Springware Board-specific variances

On Dialogic® Springware Boards using PDK protocols, all of the mask parameter values are supported. See the **gc_SetEvtMsk( )** function reference page in the *Dialogic® Global Call API Library Reference* for more information.

On Dialogic Springware Boards using ICAPI protocols, the following mask parameter values are supported:

- GCMSK_ALERTING
- GCMSK_BLOCKED
- GCMSK_UNBLOCKED

Dialogic® DM3 Board-specific variances

There are no restrictions on using **gc_SetEvtMsk( )** with Dialogic® DM3 Boards. All of the mask parameter values are supported. See the **gc_SetEvtMsk( )** function reference page in the *Dialogic® Global Call API Library Reference* for more information.

8.2.20 gc_SetParm( ) Variances for E1/T1 CAS/R2

The **gc_SetParm( )** function sets the default parameters and all channel information associated with the specific line device. In addition to the GCPR_CALLINGPARTY parameter, which is common across all technologies and documented in the *Dialogic® Global Call API Library Reference*, the parameters listed in Table 25 are supported.

Table 25. Parameters Supported, gc_GetParm( ) and gc_SetParm( )

Parameter	Level	Description	Supported on
GCPR_CALLPROGRESS	channel	Enables or disables call progress; enabled by default. If this parameter is disabled, post-connect call progress is also disabled, regardless of the setting of GCPR_MEDIADETECT.	Dialogic® DM3 Boards
GCPR_LOADTONES	channel	Enables or disables downloading of predefined call progress tones to the firmware. These tones are predefined in the E1 CAS or T1 robbed bit specific configuration files and are used for call progress. The tones are downloaded during execution of the **gc_Attach( )** or **gc_AttachResource( )** function.	Dialogic® Springware Boards
GCPR_MEDIADETECT	channel	Enables or disables post-connect call progress or media detection; disabled by default.	Dialogic® DM3 Boards and Dialogic® Springware Boards
GCPR_MINDIGITS	channel	Specifies the minimum number of digits to receive before a call is offered to the application.	Dialogic® DM3 Boards

For further information about the GCPR_CALLPROGRESS, GCPR_LOADTONES, and GCPR_MEDIADETECT parameters, see Section 4.1, "Call Progress and Call Analysis", on page 25.

8.2.21 gc_SetupTransfer( ) Variances for E1/T1 CAS/R2

The **gc_SetupTransfer**( ) function is only valid for applications using PDK protocols that support call hold and transfer. Check the **sys_features** parameter in the .cdp file for a value of Feature_Transfer. See the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for more information.

8.2.22 gc_Start( ) and gc_Stop( ) Variances for E1/T1 CAS/R2

Dialogic® Springware Board-specific variances

On Dialogic® Springware Boards using PDK protocols, the **gc_Start**( ) function is used to access the error and debug logging capabilities of the PDKRT call control library. See Section 7.2, "Debugging Applications That Use PDK Protocols", on page 73 for more information.

On Dialogic Springware Boards using ICAPI protocols, when the **gc_Start**( ) function is called, a log file, if enabled, is created. This file logs debug information for all ICAPI call control libraries for all open channels. The log file remains open until the **gc_Stop**( ) function is called. This allows channels to be opened, closed, and reopened multiple times without overwriting or otherwise affecting the continuity of the log file. See Section 7.3, "Debugging Applications That Use ICAPI Protocols", on page 78 for more information.

Dialogic® DM3 Board-specific variances

There are no restrictions on using **gc_Start( )** and **gc_Stop( )** with Dialogic® DM3 Boards.

8.2.23 gc_StartTrace( ) Variances for E1/T1 CAS/R2

Note: The variances described in this section apply when using Dialogic® Springware Boards only. The **gc_StartTrace( )** function is not supported when using Dialogic® DM3 Boards.

When using PDK protocols, the **gc_StartTrace( )** function can be used to enable logging on individual channels. This function has no effect unless the name of the log file and the logging level have been set using the **gc_Start( )** function. The **gc_StartTrace( ) filename** parameter is ignored. The name of the log file is specified in the PDK_START_STRUCT data structure. See Section 7.2, "Debugging Applications That Use PDK Protocols", on page 73 for more information.

8.2.24 gc_SwapHold( ) Variances for E1/T1 CAS/R2

The **gc_SwapHold( )** function is only valid for applications using PDK protocols that support call hold and transfer. Check the **sys_features** parameter in the .cdp file for a value of Feature_Transfer. See the *Dialogic® Global Call Country Dependent Parameters (CDP) for PDK Protocols Configuration Guide* for more information.

E1/T1 CAS/R2-Specific Data Structures 9

This chapter describes the data structures that are specific to E1/T1 CAS/R2 technology.

- IC_MAKECALL_BLK 106
- PDK_MAKECALL_BLK 107

Note: These data structures are used with Dialogic® Springware Boards only.

IC_MAKECALL_BLK

```
typedef struct ic_makecall_blk{
   unsigned long         flags;
   void                  *v_rfu_ptr;
   unsigned long         ul_rfu[4];
}IC_MAKECALL_BLK;
```

Description

For Dialogic® Springware Boards using ICAPI protocols, the IC_MAKECALL_BLK structure contains information used by the **gc_MakeCall( )** function when setting up a call. When the **gc_MakeCall( )** function sets up a call, the default is to enable call analysis (call progress). This default can be changed on a call basis by setting the **flags** parameter in the IC_MAKECALL_BLK data structure.

Field Descriptions

The fields of the IC_MAKECALL_BLK data structure are described as follows:

flags
: Controls call analysis on a per call basis. The flags included are:
  - NO_CALL_PROGRESS – Set to 0 to enable call analysis (default). Set to 1 to disable call analysis.

*v_rfu_ptr
: Reserved for future use.

ul_rfu[4]
: Reserved for future use.

PDK_MAKECALL_BLK

```
typedef struct pdk_makecall_blk{
   unsigned long         flags;
   void                  *v_rfu_ptr;
   unsigned long         ul_rfu[4];
}PDK_MAKECALL_BLK;
```

■ Description

For Dialogic® Springware Boards using PDK protocols, the PDK_MAKECALL_BLK structure contains information used by the **gc_MakeCall( )** function when setting up a call. When the **gc_MakeCall( )** function sets up a call, the default is to enable call analysis (call progress). This default can be changed on a call basis by setting the **flags** parameter in the PDK_MAKECALL_BLK data structure.

Note: Control of call progress and media detection at **gc_MakeCall( )** time works only when the following parameters in the .cdp file are set to allow application control:

```
/* Set to 0 to disable, 1 to enable, and 2 to allow app control */
R4 INTEGER_t PSL_MakeCall_CallProgress = 0
DM3 INTEGER_t PSL_CACallProgressOverride = 0

/* Set 1 to enable, 2 to allow app control */
R4 INTEGER_t PSL_MakeCall_MediaDetect  = 2
DM3 INTEGER_t PSL_CAMediaDetectOverride = 2
```

■ Field Descriptions

The fields of the PDK_MAKECALL_BLK data structure are described as follows:

flags
: 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.

*v_rfu_ptr
: Reserved for future use.

ul_rfu[4]
: Reserved for future use.

■ Example

```
/* 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;
}
```

E1/T1 CAS/R2-Specific Event Cause Values 10

This chapter lists the supported E1/T1 CAS/R2-specific event cause values, which are retrieved by **gc_ResultValue( )** and **gc_ResultInfo( )**, and provides a description of each value.

Note: The information in this chapter is applicable to Dialogic® DM3 Boards only.

Table 26 lists the E1/T1 CAS/R2 call control library cause values supported by Dialogic DM3 Boards.

Table 26. 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.		

Table 27 lists the firmware-related cause values supported by Dialogic DM3 Boards.

Table 27. 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
Note: The cause values in this table are ORed with the value 0xC0, which identifies them as firmware-related cause values.		

Table 27. Firmware-Related Cause Values When Using Dialogic® DM3 Boards (Continued)

Cause Value (Decimal)	Cause Value (Hex)	Description
09	0x09	Forward
10	0x0A	Incompatible
11	0x0B	Incoming call
12	0x0C	New call
13	0x0D	No answer from user
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
Note: The cause values in this table are ORed with the value 0xC0, which identifies them as firmware-related cause values.		

Table 27. Firmware-Related Cause Values When Using Dialogic® DM3 Boards (Continued)

Cause Value (Decimal)	Cause Value (Hex)	Description
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.		

Supplementary Reference Information 11

This chapter lists references to publications about E1/T1 CAS/R2 technology.

For additional information about E1 or T1 telephony, see the following publications:

- R2 MF Signaling References
  - *Specifications of Signaling Systems R1 and R2*, International Telegraph and Telephone Consultative Committee (CCITT), Blue Book Vol. VI, Fascicle VI.4, ISBN 92-61-03481-0
  - *General Recommendations on Telephone Switching and Signaling*, International Telegraph and Telephone Consultative Committee (CCITT), Blue Book Vol. VI, Fascicle VI.1, ISBN 92-61-03451-9
- T1 Robbed Bit Signaling References
  - Bellamy, John, *Digital Telephony*, 2nd ed. New York: John Wiley & Sons, 1991
  - Fike, John L., and George Friend, *Understanding Telephone Electronics*, Indiana: Howard W. Sams & Company, 1988
  - Flanagan, William A., *The Guide to T-1 Networking*, 4th ed. New York, Telecom Library Inc., 1990
  - *LATA Switching Systems Generic Requirements (LSSGR)*, Bellcore Technical Reference TR-TSY-000064, Issue 2, July 1987, and modules, Bellcore

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