



Dialogic[®] Voice API

Programming Guide

April 2009

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05-2332-005	January 2008	Made global changes to reflect Dialogic brand. Product Description chapter: Added information about silence compressed record in Play and Record Features section. Recording and Playback chapter: Added support for silence compressed record in Silence Compressed Record section.
05-2332-004	August 2006	Product Description chapter: Added support for speed control in Speed and Volume Control section. Speed and Volume Control chapter: Added support for speed control.
05-2332-003	December 2005	Product Description chapter: Updated TDM Bus Routing section to include information about Dialogic® digital network interface boards. Application Development Guidelines chapter: Added note about continuous speech processing (CSP) multiprocess support in Multithreading and Multiprocessing section. Added bullet about digits not always being cleared by <code>dx_clrldigbuf()</code> in Tone Detection Considerations section. Call Progress Analysis chapter: Updated dial tone detection row to 'yes' in Call Progress Analysis Support with <code>dx_dial()</code> table. Updated to add support for <code>ATDX_CRTNID()</code> and enhanced SIT sequences. Added eight new SIT sequences that can be returned by <code>ATDX_CRTNID()</code> for DM3 boards in Types of Tones section. Revised values of <code>TID_SIT_NC</code> (Freq of first segment changed from 950/1001 to 950/1020) and <code>TID_SIT_VC</code> (Freq of first segment changed from 950/1001 to 950/1020) in table of Special Information Tone Sequences (DM3); also added four new SIT sequences to this table. Added note about SIT sequences that cannot be modified in API Functions for Manipulating Tone Definitions section. Added note about SRL device mapper functions in Steps to Modify a Tone Definition on DM3 Boards section. Recording and Playback chapter: Added 128 Kbps (8 kHz, 16-bit) linear PCM to Voice Encoding Methods table. Added cross-reference to a related technical note in Transaction Record section. Global Tone Detection and Generation, and Cadenced Tone Generation chapter: Added note about the effect of exhausting the number of tone templates in Guidelines for Creating User-Defined Tones section. Added bullet about the effect of adding a tone with a frequency of zero in Guidelines for Creating User-Defined Tones section. [PTR 34546]

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05-2332-001	September 2004	Initial version of document.

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 provides instructions for developing applications on Linux and Windows® operating systems using the Dialogic® Voice API that is supplied with the Dialogic® Host Media Processing (HMP) Software product. This document is a companion guide to the *Dialogic® Voice API Library Reference*, which describes the voice functions, data structures, events, and error codes.

Applicability

This document version (05-2332-006) is published for Dialogic® Host Media Processing Software Release 3.0WIN and Dialogic® Host Media Processing Software Release 3.1LIN.

This document may also be applicable to other software releases (including service updates) on Linux or 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 software developers who choose to access the voice software. They may include any of the following:

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

How to Use This Publication

This publication assumes that you are familiar with the Linux or Windows® operating systems and the C programming language.

The information in this guide is organized as follows:

- [Chapter 1, “Product Description”](#) introduces the features of the voice library and provides a brief description of each feature.
- [Chapter 2, “Programming Models”](#) provides a brief overview of supported programming models.
- [Chapter 3, “Device Handling”](#) discusses topics related to devices, such as device naming concepts.
- [Chapter 4, “Event Handling”](#) provides information on functions used to handle events.
- [Chapter 5, “Error Handling”](#) provides information on handling errors in your application.
- [Chapter 6, “Application Development Guidelines”](#) provides programming guidelines and techniques for developing an application using the voice library.
- [Chapter 7, “Call Progress Analysis”](#) describes components of call progress analysis and discusses how one can use call progress analysis.
- [Chapter 8, “Recording and Playback”](#) discusses playback and recording features, such as encoding algorithms, and play and record API functions.
- [Chapter 9, “Speed and Volume Control”](#) explains how to control the speed and volume of playback recordings through API functions and data structures.
- [Chapter 10, “Global Tone Detection and Generation, and Cadenced Tone Generation”](#) describes these tone detection and tone generation features.
- [Chapter 11, “Building Applications”](#) discusses compiling and linking requirements such as include files and library files.

A glossary and index are provided for your reference.

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)

This chapter provides information on Dialogic® Voice API library features and capabilities. The following topics are covered:

- Overview 13
- Dialogic® R4 API 13
- Dialogic® Host Media Processing Software 14
- Call Progress Analysis 14
- Tone Generation and Detection Features 14
- Play and Record Features 15
- TDM Bus Routing 16

1.1 Overview

The voice software is a building block for creating computer telephony applications. It offers a comprehensive set of features such as dual-tone multifrequency (DTMF) detection, tone signaling, call progress analysis, playing and recording that supports a number of encoding methods, and much more.

The Dialogic® voice software consists of a C language library of functions, device drivers, and firmware.

The Dialogic® Voice API library is well integrated with other technology libraries provided by Dialogic such as fax, conferencing, and continuous speech processing. This architecture enables you to add new capability to your voice application.

For a list of voice features by product, see the Release Guide for your system release.

1.2 Dialogic® R4 API

The term R4 API (“System Software Release 4 Application Programming Interface”) describes the direct interface used for creating computer telephony application programs. The Dialogic® R4 API is a rich set of proprietary APIs for building computer telephony applications on Dialogic® products, including Dialogic® Host Media Processing (HMP) software. These APIs encompass technologies that include voice, conferencing, fax, and speech. This document describes the Dialogic® Voice API.

In this document, the term Dialogic® Voice API is used to refer to the Dialogic® R4 Voice API.

1.3 Dialogic® Host Media Processing Software

Dialogic® Host Media Processing (HMP) Software performs media processing tasks on general-purpose servers based on Dialogic® architecture without the need for specialized hardware. When installed on a system, Dialogic® HMP Software performs like a virtual Dialogic® DM3 board to the customer application, but all media processing takes place on the host processor. In this document, the term “board” represents the virtual Dialogic® DM3 board.

Dialogic® DM3 boards refer to products that are based on the next-generation Dialogic® DM3 mediastream architecture. These products succeed the original Dialogic® Springware products (also known as earlier-generation products).

1.4 Call Progress Analysis

Call progress analysis monitors the progress of an outbound call after it is dialed into the Public Switched Telephone Network (PSTN).

See [Chapter 7, “Call Progress Analysis”](#) for detailed information about this feature.

1.5 Tone Generation and Detection Features

In addition to DTMF and MF tone detection and generation, the following signaling features are provided by the voice library:

- [Global Tone Detection \(GTD\)](#)
- [Global Tone Generation \(GTG\)](#)
- [Cadenced Tone Generation](#)

1.5.1 Global Tone Detection (GTD)

Global tone detection allows you to define single or dual-frequency tones for detection on a channel-by-channel basis. Global tone detection and GTD tones are also known as **user-defined tone detection** and **user-defined tones**.

Use global tone detection to detect single- or dual-frequency tones outside the standard DTMF range of 0-9, a-d, *, and #. The characteristics of a tone can be defined and tone detection can be enabled using GTD functions and data structures provided in the voice library.

See [Chapter 10, “Global Tone Detection and Generation, and Cadenced Tone Generation”](#) for detailed information about global tone detection.

1.5.2 Global Tone Generation (GTG)

Global tone generation allows you to define a single- or dual-frequency tone in a tone generation template and to play the tone on a specified channel.

See [Chapter 10, “Global Tone Detection and Generation, and Cadenced Tone Generation”](#) for detailed information about global tone generation.

1.5.3 Cadenced Tone Generation

Cadenced tone generation is an enhancement to global tone generation. It allows you to generate a tone with up to 4 single- or dual-tone elements, each with its own on/off duration, which creates the signal pattern or cadence. You can define your own custom cadenced tone or take advantage of the built-in set of standard PBX call progress signals, such as dial tone, ringback, and busy.

See [Chapter 10, “Global Tone Detection and Generation, and Cadenced Tone Generation”](#) for detailed information about cadenced tone generation.

1.6 Play and Record Features

The following play and record features are provided by the Dialogic® Voice API library:

- [Play and Record Functions](#)
- [Speed and Volume Control](#)
- [Transaction Record](#)
- [Silence Compressed Record](#)
- [Streaming to Board](#)

1.6.1 Play and Record Functions

The Dialogic® Voice API library includes several functions and data structures for recording and playing audio data. These allow you to digitize and store human voice; then retrieve, convert, and play this digital information.

For more information about play and record features, see [Chapter 8, “Recording and Playback”](#). This chapter also includes information about voice encoding methods supported; see [Section 8.5, “Voice Encoding Methods”](#), on page 53. For detailed information about play and record functions, see the Dialogic® *Voice API Library Reference*.

1.6.2 Speed and Volume Control

The speed and volume control feature allows you to control the speed and volume of a message being played on a channel, for example, by entering a DTMF tone.

See [Chapter 9, “Speed and Volume Control”](#) for more information about this feature.

1.6.3 Transaction Record

The transaction record feature allows voice activity on two channels to be summed and stored in a single file, or in a combination of files, devices, and memory. This feature is useful in call center applications where it is necessary to archive a verbal transaction or record a live conversation.

See [Chapter 8, “Recording and Playback”](#) for more information on the transaction record feature.

1.6.4 Silence Compressed Record

The silence compressed record (SCR) feature enables recording with silent pauses eliminated. This results in smaller recorded files with no loss of intelligibility.

When the audio level is at or falls below the silence threshold for a minimum duration of time, silence compressed record begins. If a short burst of noise (glitch) is detected, the compression does not end unless the glitch is longer than a specified period of time.

See [Chapter 8, “Recording and Playback”](#) for more information.

1.6.5 Streaming to Board

The streaming to board feature allows you to stream data to a network interface in real time. Unlike the standard voice play feature (store and forward), data can be streamed in real time with little delay as the amount of initial data required to start the stream is configurable. The streaming to board feature is essential for applications such as text-to-speech, distributed prompt servers, and IP gateways.

For more information about this feature, see [Chapter 8, “Recording and Playback”](#).

1.7 TDM Bus Routing

A time division multiplexing (TDM) bus is a technique for transmitting a number of separate digitized signals simultaneously over a communication medium. TDM bus includes the CT Bus.

The CT Bus is an implementation of the computer telephony bus standard developed by the Enterprise Computer Telephony Forum (ECTF) and accepted industry-wide. The H.100 hardware specification covers CT Bus implementation using the PCI form factor. The CT Bus has 4096 bi-directional time slots.

On Dialogic® HMP Software, no physical TDM bus exists but its functionality is implemented in the software. Starting with Dialogic® HMP Software Release 2.0 for Windows®, Dialogic® digital network interface boards are supported. These boards have a bridge device that is capable of streaming data between HMP software and boards connected to the CT Bus. Additionally, the bridge device is capable of providing clocking to HMP software. The clocking provided to HMP software from a digital network interface board is derived from CT Bus clocking. For more information on HMP clocking, see the Configuration Guide associated with your software release.

For information on TDM bus routing functions, see the *Dialogic® Voice API Library Reference*.

Product Description

This chapter briefly discusses the Dialogic® Standard Runtime Library and supported programming models:

- Dialogic® Standard Runtime Library 19
- Asynchronous Programming Models 19
- Synchronous Programming Model 20

2.1 Dialogic® Standard Runtime Library

The Dialogic® Standard Runtime Library (SRL) provides a set of common system functions that are device independent and are applicable to all Dialogic® devices. The SRL consists of a data structure, event management functions, device management functions (called standard attribute functions), and device mapper functions. You can use the SRL to simplify application development, such as by writing common event handlers to be used by all devices.

When developing voice processing applications, refer to the Standard Runtime Library documentation in tandem with the voice library documentation. For more information on the Standard Runtime Library, see the *Dialogic® Standard Runtime Library API Library Reference* and *Dialogic® Standard Runtime Library API Programming Guide*.

2.2 Asynchronous Programming Models

Asynchronous programming enables a single program to control multiple voice channels within a single process. This allows the development of complex applications where multiple tasks must be coordinated simultaneously.

The asynchronous programming model uses functions that do not block thread execution; that is, the function continues processing under the hood. A Standard Runtime Library (SRL) event later indicates function completion.

Generally, if you are building applications that use any significant density, you should use the asynchronous programming model to develop field solutions.

For complete information on asynchronous programming models, see the *Dialogic® Standard Runtime Library API Programming Guide*.

2.3 Synchronous Programming Model

The synchronous programming model uses functions that block application execution until the function completes. This model requires that each channel be controlled from a separate process. This allows you to assign distinct applications to different channels dynamically in real time.

Synchronous programming models allow you to scale an application by simply instantiating more threads or processes (one per channel). This programming model may be easy to encode and manage but it relies on the system to manage scalability. Applying the synchronous programming model can consume large amounts of system overhead, which reduces the achievable densities and negatively impacts timely servicing of both hardware and software interrupts. Using this model, a developer can only solve system performance issues by adding memory or increasing CPU speed or both. The synchronous programming models may be useful for testing or very low-density solutions.

For complete information on synchronous programming models, see the *Dialogic® Standard Runtime Library API Programming Guide*.

This chapter describes the concept of a voice device and how voice devices are named and used.

- Device Concepts 21
- Voice Device Names 21

3.1 Device Concepts

The following concepts are important in understanding devices and device handling:

device

A device is a computer component controlled through a software device driver. A resource board, such as a voice resource, fax resource, and conferencing resource, and network interface board, contains one or more logical board devices. Each channel or time slot on the board is also considered a device.

device channel

A device channel refers to a data path that processes one incoming or outgoing call at a time (equivalent to the terminal equipment terminating a phone line).

device name

A device name is a literal reference to a device, used to gain access to the device via an `xx_open()` function, where “xx” is the prefix defining the device to be opened. For example, “dx” is the prefix for voice device and “fx” for fax device.

device handle

A device handle is a numerical reference to a device, obtained when a device is opened using `xx_open()`, where “xx” is the prefix defining the device to be opened. The device handle is used for all operations on that device.

virtual boards

The device driver views a single voice board with more than four channels as multiple emulated Dialogic® D/4x boards. These emulated boards are called virtual boards. For example, a system with 44 voice channels consists of 11 virtual boards.

3.2 Voice Device Names

The software assigns a device name to each device or each component on a board. A voice device is named **dxxxBn**, where **n** is the device number assigned in sequential order down the list of sorted voice boards. A device corresponds to a grouping of two or four voice channels.

For example, a system running Dialogic® Host Media Processing (HMP) software with 44 voice channels has 11 virtual board devices, where each device consists of four channels. Examples of board device names for voice boards are `dxxxB1` and `dxxxB2`.

Device Handling

A device name can be appended with a channel or component identifier. A voice channel device is named **dxxxBnCy**, where **y** corresponds to one of the voice channels. Examples of channel device names for voice boards are dxxxB1C1 and dxxxB1C2.

Use the Dialogic® Standard Runtime Library device mapper functions to retrieve information on all devices in a system.

For complete information on device handling, see the *Dialogic® Standard Runtime Library API Programming Guide*.

This chapter provides information on functions used to retrieve and handle events. Topics include:

- [Overview of Event Handling](#) 23
- [Event Management Functions](#) 23

4.1 Overview of Event Handling

An event indicates that a specific activity has occurred on a channel. The voice driver reports channel activity to the application program in the form of events, which allows the program to identify and respond to a specific occurrence on a channel. Events provide feedback on the progress and completion of functions and indicate the occurrence of other channel activities. Dialogic® Voice API library events are defined in the *dxlib.h* header file.

For a list of events that may be returned by the voice software, see the *Dialogic® Voice API Library Reference*.

4.2 Event Management Functions

Event management functions are used to retrieve and handle events being sent to the application from the firmware. These functions are contained in the Dialogic® Standard Runtime Library (SRL) and defined in *srlib.h*. The SRL provides a set of common system functions that are device independent and are applicable to all Dialogic® devices. For more information on event management and event handling, see the *Dialogic® Standard Runtime Library API Programming Guide*.

Event management functions include:

- `sr_enbhdr()`
- `sr_dishdr()`
- `sr_getevtdev()`
- `sr_getevtttype()`
- `sr_getevtflen()`
- `sr_getevtdatap()`

For details on SRL functions, see the *Dialogic® Standard Runtime Library API Library Reference*.

The event management functions retrieve and handle voice device termination events for functions that run in asynchronous mode, such as `dx_dial()` and `dx_play()`. For complete function reference information, see the *Dialogic® Voice API Library Reference*.

Event Handling

Each of the event management functions applicable to the voice boards are listed in the following tables. Table 1 lists values that are required by event management functions. Table 2 list values that are returned for event management functions that are used with voice devices.

Table 1. Voice Device Inputs for Event Management Functions

Event Management Function	Voice Device Input	Valid Value	Related Voice Functions
sr_enbhdr() Enable event handler	evt_type	TDX_PLAY	dx_play()
		TDX_PLAYTONE	dx_playtone()
		TDX_RECORD	dx_rec()
		TDX_GETDIG	dx_getdig()
		TDX_DIAL	dx_dial()
		TDX_CALLP	dx_dial()
		TDX_ERROR	All asynchronous functions
sr_dishdr() Disable event handler	evt_type	As above	As above

Table 2. Voice Device Returns from Event Management Functions

Event Management Function	Return Description	Returned Value	Related Voice Functions
sr_getvtdev() Get device handle	device	voice device handle	
sr_getvttype() Get event type		TDX_PLAY	dx_play()
		TDX_PLAYTONE	dx_playtone()
		TDX_RECORD	dx_rec()
		TDX_GETDIG	dx_getdig()
		TDX_DIAL	dx_dial()
		TDX_CALLP	dx_dial()
		TDX_CST	dx_setvtmsk()
	TDX_ERROR	All asynchronous functions	
sr_getvtlen() Get event data length	event length	sizeof (DX_CST)	
sr_getvtdatap() Get pointer to event data	event data	pointer to DX_CST structure	

This chapter discusses how to handle errors that can occur when running an application.

All Dialogic® Voice API library functions return a value to indicate success or failure of the function. A return value of zero or a non-negative number indicates success. A return value of -1 indicates failure.

If a Dialogic® Voice API library function fails, call the standard attribute functions **ATDV_LASTERR()** and **ATDV_ERRMSGP()** to determine the reason for failure. For more information on these functions, see the *Dialogic® Standard Runtime Library API Library Reference*.

If an extended attribute function fails, two types of errors can be generated. An extended attribute function that returns a pointer will produce a pointer to the ASCII string “Unknown device” if it fails. An extended attribute function that does not return a pointer will produce a value of **AT_FAILURE** if it fails. Extended attribute functions for the voice library are prefaced with “**ATDX_**”.

- Notes:**
1. The **dx_open()** and **dx_close()** functions are exceptions to the above error handling rules. On Linux, if these functions fail, the return code is -1, and the specific error is found in the **errno** variable contained in *errno.h*. On Windows®, if these functions fail, the return code is -1. Use **dx_fileerrno()** to obtain the system error value.
 2. If **ATDV_LASTERR()** returns the **EDX_SYSTEM** error code, an operating system error has occurred. On Linux, check the global variable **errno** contained in *errno.h*. On Windows®, use **dx_fileerrno()** to obtain the system error value.

For a list of errors that can be returned by a Dialogic® Voice API library function, see the *Dialogic® Voice API Library Reference*. You can also look up the error codes in the *dxxlib.h* file.

Error Handling

Application Development Guidelines

6

This chapter provides programming guidelines and techniques for developing an application using the Dialogic® Voice API library. The following topics are discussed:

- [General Considerations](#) 27
- [Additional Considerations](#) 30

6.1 General Considerations

The following considerations apply to all applications written using the Dialogic® Voice API:

- [Busy and Idle States](#)
- [Setting Termination Conditions for I/O Functions](#)
- [Setting Termination Conditions for Digits](#)
- [Clearing Structures Before Use](#)
- [Working with User-Defined I/O Functions](#)

See feature chapters for programming guidelines specific to a feature, such as call progress analysis, recording and playback, and so on.

6.1.1 Busy and Idle States

The operation of some library functions are dependent on the state of the device when the function call is made. A device is in an idle state when it is not being used, and in a busy state when it is dialing, stopped, being configured, or being used for other I/O functions. Idle represents a single state; busy represents the set of states that a device may be in when it is not idle. State-dependent functions do not make a distinction between the individual states represented by the term busy. They only distinguish between idle and busy states.

For more information on categories of functions and their description, see the *Dialogic® Voice API Library Reference*.

6.1.2 Setting Termination Conditions for I/O Functions

When an I/O function is issued, you must pass a set of termination conditions as one of the function parameters. Termination conditions are events monitored during the I/O process that will cause an I/O function to terminate. When the termination condition is met, a termination reason is returned by `ATDX_TERMMSK()`. If the I/O function is running in synchronous mode, the `ATDX_TERMMSK()` function returns a termination reason after the I/O function has completed.

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If the I/O function is running in asynchronous mode, the **ATDX_TERMMSK()** function returns a termination reason after the function termination event has arrived. I/O functions can terminate under several conditions as described later in this section.

You can predict events that will occur during I/O (such as a digit being received or the call being disconnected) and set termination conditions accordingly. The flow of control in a voice application is based on the termination condition. Setting these conditions properly allows you to build voice applications that can anticipate a caller's actions.

To set the termination conditions, values are placed in fields of a DV_TPT structure. If you set more than one termination condition, the first one that occurs will terminate the I/O function. The DV_TPT structures can be configured as a linked list or array, with each DV_TPT specifying a single terminating condition. For more information on the DV_TPT structure, which is defined in *srlib.h*, see the *Dialogic® Voice API Library Reference*.

The termination conditions are described in the following paragraphs.

byte transfer count

This termination condition applies when playing or recording a file with **dx_play()** or **dx_rec()**. The maximum number of bytes is set in the DX_IOTT structure. This condition will cause termination if the maximum number of bytes is used before one of the termination conditions specified in the DV_TPT occurs. For information about setting the number of bytes in the DX_IOTT, see the *Dialogic® Voice API Library Reference*.

dx_stopch() occurred

The **dx_stopch()** function will terminate any I/O function, except **dx_dial()** (with call progress analysis disabled), and stop the device. See the **dx_stopch()** function description for more detailed information about this function.

end of file reached

This termination condition applies when playing a file. This condition will cause termination if -1 has been specified in the *io_length* field of the DX_IOTT, and no other termination condition has occurred before the end of the file is reached. For information about setting the DX_IOTT, see the *Dialogic® Voice API Library Reference*. When this termination condition is met, a TM_EOD termination reason is returned from **ATDX_TERMMSK()**.

maximum delay between digits (DX_IDDTIME)

This termination condition monitors the length of time between the digits being received. A specific length of time can be placed in the *tp_length* field of a DV_TPT. If the time between receiving digits is more than this period of time, the function terminates. The amount of time can be specified in 100 msec units (default) or 10 msec units. 10 msec units can be specified in the *tp_flags* field of the DV_TPT. When this termination condition is met, a TM_IDDTIME termination reason is returned from **ATDX_TERMMSK()**.

This termination condition is only supported by the **dx_getdig()** function.

maximum digits received (DX_MAXDTMF)

This termination condition counts the number of digits in the channel's digit buffer. If the buffer is not empty before the I/O function is called, the digits that are present in the buffer when the function is initiated are counted as well. The maximum number of digits to receive is set by placing a number from 1 to 31 in the *tp_length* field of a DV_TPT. This value specifies the number of digits allowed in the buffer before termination. When this termination condition is met, a TM_MAXDTMF termination reason is returned from **ATDX_TERMMSK()**.

maximum length of silence (DX_MAXSIL)

This termination condition is enabled by setting the `tp_length` field of a DV_TPT. The specified value is the length of time that continuous silence will be detected before it terminates the I/O function. The amount of time can be specified in 100 msec units (default) or 10 msec units. 10 msec units can be specified in the `tp_flags` field of the DV_TPT. When this termination condition is met, a TM_MAXSIL termination reason is returned from **ATDX_TERMMSK()**.

specific digit received (DX_DIGMASK)

Digits received during an I/O function are collected in a channel's digit buffer. If the buffer is not empty before an I/O function executes, the digits in the buffer are treated as being received during the I/O execution. This termination condition is enabled by specifying a digit bit mask in the `tp_length` field of a DV_TPT structure. If any digit specified in the bit mask appears in the digit buffer, the I/O function will terminate. When this termination condition is met, a TM_DIGIT termination reason is returned from **ATDX_TERMMSK()**.

Using more than one DX_IOTT structure for detecting different digits is not supported. Instead, use one DX_IOTT structure, set DX_DIGMASK in the `tp_termno` field, and bitwise-OR "DM_1 | DM_2" in the `tp_length` field.

maximum function time (DX_MAXTIME)

A time limit may be placed on the execution of an I/O function. The `tp_length` field of a DV_TPT can be set to a specific length of time in 100 msec units. The I/O function will terminate when it executes longer than this period of time. The amount of time can be specified in 100 msec units (default) or 10 msec units. 10 msec units can be specified in the `tp_flags` field of the DV_TPT. When this termination condition is met, a TM_MAXTIME termination reason is returned from **ATDX_TERMMSK()**.

DX_MAXTIME is not supported by tone generation functions such as **dx_playtone()** and **dx_playtoneEx()**.

user-defined digit received (DX_DIGTYPE)

User-defined digits received during an I/O function are collected in a channel's digit buffer. If the buffer is not empty before an I/O function executes, the digits in the buffer are treated as being received during the I/O execution. This termination condition is enabled by specifying the digit and digit type in the `tp_length` field of a DV_TPT structure. If any digit specified in the bit mask appears in the digit buffer, the I/O function will terminate. When this termination condition is met, a TM_DIGIT termination reason is returned from **ATDX_TERMMSK()**.

user-defined tone on/off event detected (DX_TONE)

This termination condition is used with global tone detection. Before specifying a user-defined tone as a termination condition, the tone must first be defined using the GTD **dx_bld...()** functions, and tone detection on the channel must be enabled using the **dx_addtone()** or **dx_enbtone()** function. To set tone on/off to be a termination condition, specify DX_TONE in the `tp_termno` field of the DV_TPT. You must also specify DX_TONEON or DX_TONEOFF in the `tp_data` field. When this termination condition is met, a TM_TONE termination reason is returned from **ATDX_TERMMSK()**.

6.1.3 Setting Termination Conditions for Digits

To specify a timeout for **dx_getdig()** if the first digit is not received within a specified time period, use the DX_MAXTIME termination condition in the DV_TPT structure.

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To specify an additional timeout if subsequent digits are not received, use the `DX_IDDTIME` (interdigit delay) termination condition and the `TF_FIRST` flag in the `DV_TPT` structure. The `TF_FIRST` flag specifies that the timer will start after the first digit is received; otherwise the timer starts when the `dx_getdig()` function is called.

6.1.4 Clearing Structures Before Use

Two library functions are provided to clear structures. `dx_clracap()` clears `DX_CAP` structures and `dx_clrtppt()` clears `DV_TPT` structures. See the *Dialogic® Voice API Library Reference* for details.

It is good practice to clear the field values of any structure before using the structure in a function call. Doing so will help prevent unintentional settings or terminations.

6.1.5 Working with User-Defined I/O Functions

Two library functions are provided to enable you to install user-defined I/O functions (also called user I/O functions or UIO): `dx_setuio()` and `dx_setdevuio()`. For details on these functions, see the *Dialogic® Voice API Library Reference*.

The following cautions apply when working with user I/O functions:

- Do not include sleeps, critical sections, or any other delays in the user I/O function.
- Do not call any other Dialogic function inside the user I/O function. One exception is the `ec_getblkinfo()` function which is called from within a user I/O function. For more information on this function, see the *Dialogic® Continuous Speech Processing API Library Reference*.

The reason for these cautions is as follows. On Dialogic® Springware boards, while the user I/O function is executing, the Dialogic® Standard Runtime Library (SRL) is blocked and cannot process further messages from the driver. Data will be lost if the driver cannot hand off messages to the SRL. On Dialogic® DM3 boards, you may see chopped audio or underruns. On all boards, be aware that the risk of underruns increases as density rises.

6.2 Additional Considerations

The following information provides programming guidelines and considerations for developing voice applications:

- [Multithreading and Multiprocessing](#)
- [Device Initialization Hint](#)
- [Tone Detection Considerations](#)

6.2.1 Multithreading and Multiprocessing

The Dialogic® Voice API supports multithreading and multiprocessing on the board level but not on the channel level.

The following restrictions apply:

- A channel can only be opened in one process at a time; the same channel cannot be used by more than one process concurrently. However, multiple processes can access different sets of channels. Ensure that each process is provided with a unique set of devices to manipulate.
- If a channel is opened in process A and then closed, process B is allowed to open the same channel. However, you should avoid this type of sequence. Since closing a channel is an asynchronous operation, there is a small gap between the time when the `xx_close()` function returns in process A and the time when process B is allowed to open the same channel. If process B opens the channel too early, unpredictable results may occur.
- Multiple processes that define tones (GTD or GTG) do not share tone definitions in the firmware. For example, if you define tone A in process 1 for channel `dxxxB1C1` on a Dialogic® DM3 board and the same tone A in process 2 for channel `dxxxB1C1` on the same Dialogic® DM3 board, two firmware tones are consumed on the board. In other words, the same tone defined from different processes is not shared in the firmware; hence this limits the number of tones that can be created overall. For more information, see [Chapter 10, “Global Tone Detection and Generation, and Cadenced Tone Generation”](#).

It is recommended that you develop your application using a single thread per span or a single thread per board rather than a single thread per channel. For more information on programming models and performance considerations, see the *Dialogic® Standard Runtime Library API Programming Guide*.

Note: The continuous speech processing architecture allows a voice channel to be shared between processes (or applications) on Dialogic® JCT boards, on Dialogic® DM3 boards, or on Dialogic® Host Media Processing (HMP) (starting with Dialogic® Host Media Processing Software Release 1.3 for Windows®), providing one process does the play activity and the other process does the record/stream activity. Other CSP scenarios are **not** supported, such as playing or recording/streaming from both processes. For details, refer to the application note, *Telephony Application Architectures for Dialogic® Boards with DM3 Architecture*, located at

http://www.dialogic.com/products/tdm_boards/media_processing/docs/9380an.pdf.

6.2.2 Device Initialization Hint

The `xx_open()` functions for the voice (`dx`), Global Call (`gc`), network (`dt`), and fax (`fx`) APIs are asynchronous. This should usually have no impact on an application, except in cases where a subsequent function calls on a device that is still initializing, that is, is in the process of opening. In such cases, the initialization must be finished before the follow-up function can work. The function won't return an error, but it is blocked until the device is initialized.

For instance, if your application calls `dx_open()` followed by `dx_getfeaturelist()`, the `dx_getfeaturelist()` function is blocked until the initialization of the device is completed internally, even though `dx_open()` has already returned success. In other words, the initialization (`dx_open()`) may appear to be complete, but, in truth, it is still going on in parallel.

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With some applications, this may cause slow device-initialization performance. You can avoid this problem in one of several ways, depending on the type of application:

- In multithreaded applications, you can reorganize the way the application opens and then configures devices. The recommendation is to do as many `xx_open()` functions as possible (grouping the devices) in one thread arranging them in a loop before proceeding with the next function. For example, you would have one loop through the grouping of devices do all the `xx_open()` functions first, and then start a second loop through the devices to configure them, instead of doing one single loop where an `xx_open()` is immediately followed by other API functions on the same device. With this method, by the time all `xx_open()` commands are completed, the first channel will be initialized, so you won't experience problems.
This change is not necessary for all applications, but if you experience poor initialization performance, you can gain back speed by using this hint.
- Develop your application using a single thread per span or a single thread per board. This way, device initialization can still be done in a loop, and by the time the subsequent function is called on the first device, initialization on that device has completed.

6.2.3 Tone Detection Considerations

The following consideration applies to tone detection:

- Digits will not always be cleared by the time the `dx_clrdigbuf()` function returns, because processing may continue on the board even after the function returns. For this reason, careful consideration should be given when using this function before or during a section where digit detection or digit termination is required; the digit may be cleared after the function has returned or possibly during the next function call.

This chapter provides detailed information about the call progress analysis feature. The following topics are discussed:

- Call Progress Analysis Overview 33
- Call Progress and Call Analysis Terminology..... 34
- Call Progress Analysis Components 34
- Using Call Progress Analysis 35
- Call Progress Analysis Tone Detection 39
- Media Tone Detection 44
- Default Call Progress Analysis Tone Definitions 45
- Modifying Default Call Progress Analysis Tone Definitions 46
- Call Progress Analysis Errors 49

7.1 Call Progress Analysis Overview

Call progress analysis monitors the progress of an outbound call after it is dialed into the Public Switched Telephone Network (PSTN).

By using call progress analysis (CPA) you can determine for example:

- whether the line is answered and, in many cases, how the line is answered
- whether the line rings but is not answered
- whether the line is busy
- whether there is a problem in completing the call

The outcome of the call is returned to the application when call progress analysis has completed.

Caution: If your application also uses the Dialogic® Global Call API, see the Global Call documentation set for call progress analysis considerations specific to Global Call. The Dialogic® Global Call API is a common signaling interface for network-enabled applications, regardless of the signaling protocol needed to connect to the local telephone network. Call progress analysis support varies with the protocol used.

7.2 Call Progress and Call Analysis Terminology

A distinction is made between activity that occurs before a call is connected and after a call is connected. The following terms are used:

call progress (pre-connect)

This term refers to activity to determine the status of a call connection, such as busy, no ringback, no dial tone, and can also include the frequency detection of Special Information Tones (SIT), such as operator intercept. This activity occurs before a call is connected.

call analysis (post-connect)

This term refers to activity to determine the destination party's media type, such as voice detection, answering machine detection, fax tone detection, modem, and so on. This activity occurs after a call is connected.

call progress analysis

This term refers to the feature set that encompasses both call progress and call analysis.

7.3 Call Progress Analysis Components

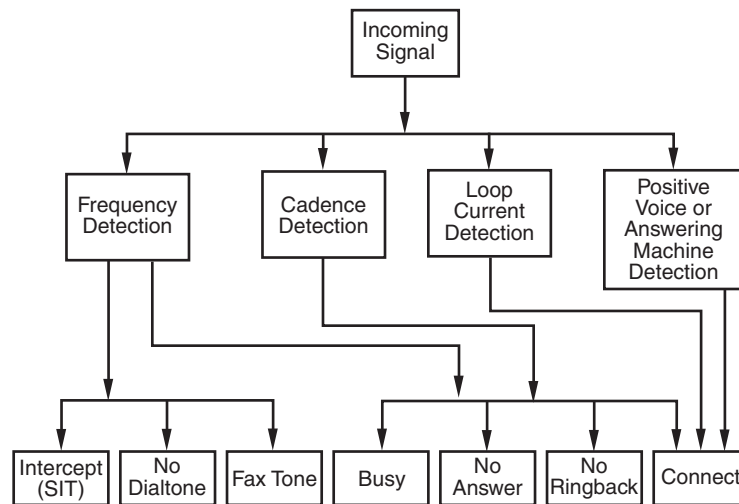
Call progress analysis uses the following techniques or components to determine the progress of a call as applicable:

- cadence detection (pre-connect part of call progress analysis)
- frequency detection (pre-connect part of call progress analysis)
- loop current detection (pre-connect part of call progress analysis)
- positive voice detection (post-connect part of call progress analysis)
- positive answering machine detection (post-connect part of call progress analysis)
- fax tone detection (post-connect part of call progress analysis)

Figure 1 illustrates the components of call progress analysis. These components can all operate simultaneously.

Note: Note that loop current detection is not applicable to Dialogic® Host Media Processing (HMP) software.

Figure 1. Call Progress Analysis Components



7.4 Using Call Progress Analysis

The following topics provide information on how to use call progress analysis:

- [Call Progress Analysis Rules](#)
- [Overview of Steps to Initiate Call Progress Analysis](#)
- [Setting Up Call Progress Analysis Parameters in DX_CAP](#)
- [Executing a Dial Function](#)
- [Determining the Outcome of a Call](#)
- [Obtaining Additional Call Outcome Information](#)

7.4.1 Call Progress Analysis Rules

The following rules apply to the use of call progress analysis using Dialogic® HMP voice devices:

- For IP ports using IP protocols:
 - Pre-connect is typically provided by the protocol via the Dialogic® Global Call API or via a native IP protocol API.
 - The `dx_dial()` function is not recommended to be used for pre-connect. In general this function is not required to be used since pre-connect call progress is typically not provided inband in the media stream. *However, the `dx_dial()` function may be used if inband pre-connect call progress is available.*

Call Progress Analysis

– The **dx_dial()** function may be used for post-connect call analysis.

Table 3 provides information on call progress analysis scenarios supported with the **dx_dial()** function.

Table 3. Call Progress Analysis Support with dx_dial()

CPA Feature	dx_dial() support
Busy	Yes
No ringback	Yes
SIT frequency detection	Yes
No answer	Yes
Cadence break	Yes
Loop current detection	No
Dial tone detection	Yes
Fax tone detection	Yes
Positive Voice Detection (PVD)	Yes
Positive Answering Machine Detection (PAMD)	Yes

7.4.2 Overview of Steps to Initiate Call Progress Analysis

Review the information in [Section 7.4.1, “Call Progress Analysis Rules”](#), on page 35. If you choose to use the voice API for call progress analysis, perform the following procedure to initiate an outbound call with call progress analysis:

1. Set up the call analysis parameter structure (DX_CAP), which contains parameters to control the operation of call progress analysis, such as positive voice detection and positive answering machine detection.
2. Call **dx_dial()** to start call progress analysis during the desired phase of the call.
3. Use the **ATDX_CPTERM()** extended attribute function to determine the outcome of the call.
4. Obtain additional termination information as desired using extended attribute functions.

Each of these steps is described in more detail next. For a full description of the functions and data structures described in this chapter, see the *Dialogic® Voice API Library Reference*.

7.4.3 Setting Up Call Progress Analysis Parameters in DX_CAP

The call progress analysis parameters structure, DX_CAP, is used by **dx_dial()**. It contains parameters to control the operation of call progress analysis features, such as positive voice detection (PVD) and positive answering machine detection (PAMD). To customize the parameters for your environment, you must set up the DX_CAP structure before calling a dial function.

To set up the DX_CAP structure for call progress analysis:

1. Execute the **dx_clrca**() function to clear the DX_CAP and initialize the parameters to 0. The value 0 indicates that the default value will be used for that particular parameter. **dx_dial**() can also be set to run with default call progress analysis parameter values, by specifying a NULL pointer to the DX_CAP structure.
2. Set a DX_CAP parameter to another value if you do not want to use the default value. The ca_intflg field (intercept mode flag) of DX_CAP enables and disables the following call progress analysis components: SIT frequency detection, positive voice detection (PVD), and positive answering machine detection (PAMD). Use one of the following values for the ca_intflg field:
 - DX_OPTDIS. Disables Special Information Tone (SIT) frequency detection, PAMD, and PVD. This setting provides call progress without SIT frequency detection.
 - DX_OPTNOCON. Enables SIT frequency detection and returns an “intercept” immediately after detecting a valid frequency. This setting provides call progress with SIT frequency detection.
 - DX_PVDENABLE. Enables PVD and fax tone detection. Provides PVD call analysis only (no call progress).
 - DX_PVDOPTNOCON. Enables PVD, DX_OPTNOCON, and fax tone detection. This setting provides call progress with SIT frequency detection and PVD call analysis.
 - DX_PAMDENABLE. Enables PAMD, PVD, and fax tone detection. This setting provides PAMD and PVD call analysis only (no call progress).
 - DX_PAMDOPTEN. Enables PAMD, PVD, DX_OPTNOCON, and fax tone detection. This setting provides full call progress and call analysis.

Note: DX_OPTEN and DX_PVDOPTEN are obsolete. Use DX_OPTNOCON and DX_PVDOPTNOCON instead.

7.4.4 Executing a Dial Function

To use call progress analysis, call **dx_dial**() with the **mode** function argument set to DX_CALLP. Termination of dialing with call progress analysis is indicated differently depending on whether the function is running asynchronously or synchronously.

If running asynchronously, use Dialogic® Standard Runtime Library (SRL) event management functions to determine when dialing with call progress analysis is complete (TDX_CALLP termination event).

If running synchronously, wait for the function to return a value greater than 0 to indicate successful completion.

- Notes:*
1. **dx_dial**() cannot be used to start an outbound call; instead use the Dialogic® Global Call API.
 2. To issue **dx_dial**() without dialing digits, specify “ ” in the **dialstrp** argument.

7.4.5 Determining the Outcome of a Call

In asynchronous mode, once **dx_dial()** with call progress analysis has terminated, use the extended attribute function **ATDX_CPTERM()** to determine the outcome of the call. (In synchronous mode, **dx_dial()** returns the outcome of the call.) **ATDX_CPTERM()** will return one of the following call progress analysis termination results:

CR_BUSY

Called line was busy.

CR_CEPT

Called line received operator intercept (SIT).

CR_CNCT

Called line was connected. Use **ATDX_CONNTYPE()** to return the connection type for a completed call.

CR_ERROR

Call progress analysis error occurred. Use **ATDX_CPERROR()** to return the type of error.

CR_FAXTONE

Called line was answered by fax machine or modem.

CR_NOANS

Called line did not answer.

CR_NORB

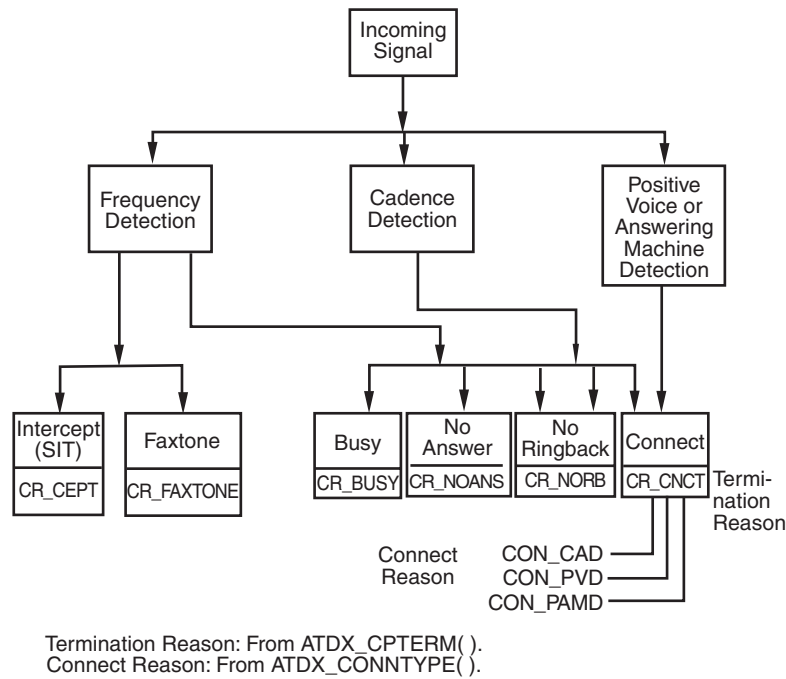
No ringback on called line.

CR_STOPD

Call progress analysis stopped due to **dx_stopch()**.

Figure 2 illustrates the possible outcomes of call progress analysis.

Figure 2. Call Outcomes for Call Progress Analysis



7.4.6 Obtaining Additional Call Outcome Information

To obtain additional call progress analysis information, use the following extended attribute functions:

ATDX_CPERROR()

Returns call analysis error.

ATDX_CPTERM()

Returns last call analysis termination reason.

ATDX_CONNTYPE()

Returns connection type.

See each function reference description in the *Dialogic® Voice API Library Reference* for more information.

7.5 Call Progress Analysis Tone Detection

The following topics discuss tone detection used in call progress analysis:

- [Tone Detection Overview](#)
- [Types of Tones](#)
- [Ringback Detection](#)

Call Progress Analysis

- [Busy Tone Detection](#)
- [Fax or Modem Tone Detection](#)
- [SIT Frequency Detection](#)

7.5.1 Tone Detection Overview

Call progress analysis uses a combination of cadence detection and frequency detection to identify certain signals during the course of an outgoing call. Cadence detection identifies repeating patterns of sound and silence, and frequency detection determines the pitch of the signal. Together, the cadence and frequency of a signal make up its “tone definition”.

7.5.2 Types of Tones

Tone definitions are used to identify several kinds of signals.

The following defined tones and tone identifiers are provided by the voice library. Tone identifiers are returned by the `ATDX_CRTNID()` function.

TID_BUSY1

First signal busy

TID_BUSY2

Second signal busy

TID_DIAL_INTL

International dial tone

TID_DIAL_LCL

Local dial tone

TID_DISCONNECT

Disconnect tone (post-connect)

TID_FAX1

First fax or modem tone

TID_FAX2

Second fax or modem tone

TID_RNGBK1

Ringback (detected as single tone)

TID_RNGBK2

Ringback (detected as dual tone)

TID_SIT_ANY

Catch all (returned for a Special Information Tone sequence or SIT sequence that falls outside the range of known default SIT sequences)

TID_SIT_INEFFECTIVE_OTHER or

TID_SIT_IO

Ineffective other SIT sequence

TID_SIT_NO_CIRCUIT or
 TID_SIT_NC
 No circuit found SIT sequence

TID_SIT_NO_CIRCUIT_INTERLATA or
 TID_SIT_NC_INTERLATA
 InterLATA no circuit found SIT sequence

TID_SIT_OPERATOR_INTERCEPT or
 TID_SIT_IC
 Operator intercept SIT sequence

TID_SIT_REORDER_TONE or
 TID_SIT_RO
 Reorder (system busy) SIT sequence

TID_SIT_REORDER_TONE_INTERLATA or
 TID_SIT_RO_INTERLATA
 InterLATA reorder (system busy) SIT sequence

TID_SIT_VACANT_CIRCUIT or
 TID_SIT_VC
 Vacant circuit SIT sequence

Some of these tone identifiers may be used as input to function calls to change the tone definitions. For more information, see [Section 7.8, “Modifying Default Call Progress Analysis Tone Definitions”](#), on page 46.

7.5.3 Ringback Detection

Call progress analysis uses the tone definition for ringback to identify the first ringback signal of an outgoing call. At the end of the first ringback (that is, normally, at the beginning of the second ringback), a timer goes into effect. The system continues to identify ringback signals (but does not count them). If a break occurs in the ringback cadence, the call is assumed to have been answered, and call progress analysis terminates with the reason CR_CNCT (connect); the connection type returned by the **ATDX_CONNTYPE()** function will be CON_CAD (cadence break).

However, if the timer expires before a connect is detected, then the call is deemed unanswered, and call progress analysis terminates with the reason CR_NOANS.

To enable ringback detection, turn on SIT frequency detection in the DX_CAP ca_intflg field. For details, see [Section 7.4.3, “Setting Up Call Progress Analysis Parameters in DX_CAP”](#), on page 36.

The following DX_CAP fields govern ringback behavior:

ca_cnosig
 Continuous No Signal: the maximum length of silence (no signal) allowed immediately after the ca_stdely period (in 10 msec units). If this duration is exceeded, call progress analysis is terminated with the reason CR_NORB (no ringback detected). Default value: 4000 (40 seconds).

Call Progress Analysis

ca_noanswer

No Answer: the length of time to wait after the first ringback before deciding that the call is not answered (in 10 msec units). If this duration is exceeded, call progress analysis is terminated with the reason CR_NOANS (no answer). Default value: 3000 (30 seconds).

7.5.4 Busy Tone Detection

Call progress analysis specifies two busy tones: TID_BUSY1 and TID_BUSY2. If either of them is detected while frequency detection and cadence detection are active, then call progress is terminated with the reason CR_BUSY. `ATDX_CRTNID()` identifies which busy tone was detected.

To enable busy tone detection, turn on SIT frequency detection in the DX_CAP ca_intflg field. For details, see [Section 7.4.3, “Setting Up Call Progress Analysis Parameters in DX_CAP”](#), on page 36.

7.5.5 Fax or Modem Tone Detection

Call progress analysis specifies two tones: TID_FAX1 and TID_FAX2. If either of these tones is detected while frequency detection and cadence detection are active, then call progress is terminated with the reason CR_FAXTONE. `ATDX_CRTNID()` identifies which fax or modem tone was detected.

To enable fax or modem tone detection, use the ca_intflg field of the DX_CAP structure. For details, see [Section 7.4.3, “Setting Up Call Progress Analysis Parameters in DX_CAP”](#), on page 36.

7.5.6 SIT Frequency Detection

Special Information Tone (SIT) frequency detection is a component of call progress analysis. SIT sequences are defined as standard tone IDs.

To enable SIT frequency detection, use the ca_intflg field of the DX_CAP structure. For more information, see [Section 7.4.3, “Setting Up Call Progress Analysis Parameters in DX_CAP”](#), on page 36.

Table 4 shows default tone definitions for SIT sequences. The values in the “**Freq.**” column represent minimum and maximum values in Hz. “**Time**” refers to minimum and maximum on time in 10 msec units; the maximum off time between each segment is 5 (or 50 msec). The repeat count is 1 for all SIT segments. N/A means “not applicable.”

Table 4. Special Information Tone Sequences

SIT		1st Segment		2nd Segment		3rd Segment	
Tone ID	Description	Freq.	Time	Freq.	Time	Freq.	Time
TID_SIT_NC	No Circuit Found	950/1020	32/45	1400/1450	32/45	1740/1850	N/A
TID_SIT_IC	Operator Intercept	874/955	15/30	1310/1430	15/30	1740/1850	N/A
TID_SIT_VC	Vacant Circuit	950/1020	32/45	1310/1430	15/30	1740/1850	N/A
TID_SIT_RO	Reorder (system busy)	874/955	15/30	1400/1450	32/45	1740/1850	N/A
TID_SIT_NC_INTERLATA	InterLATA No Circuit Found	874/955	32/45	1310/1430	32/45	1740/1850	N/A
TID_SIT_RO_INTERLATA	InterLATA Reorder (system busy)	950/1020	15/30	1310/1430	32/45	1740/1850	N/A
TID_SIT_IO	Ineffective Other	874/955	32/45	1400/1450	15/30	1740/1850	N/A
TID_SIT_ANY	Catch all tone definition	Open	Open	Open	Open	1725/1825	N/A

The following considerations apply to SIT sequences:

- A single tone proxy for the dual tone (also called twin tone) exists for each of the three segments in a SIT sequence. The default definition for the minimum value and maximum value (in Hz) is 0. For more information on this tone, see [Section 7.8.4, “Rules for Using a Single Tone Proxy for a Dual Tone”](#), on page 48.
- These tone IDs have aliases:
 - TID_SIT_NO_CIRCUIT (TID_SIT_NC)
 - TID_SIT_OPERATOR_INTERCEPT (TID_SIT_IC)
 - TID_SIT_VACANT_CIRCUIT (TID_SIT_VC)
 - TID_SIT_REORDER_TONE (TID_SIT_RO)
 - TID_SIT_NO_CIRCUIT_INTERLATA (TID_SIT_NC_INTERLATA)
 - TID_SIT_REORDER_TONE_INTERLATA (TID_SIT_RO_INTERLATA)
 - TID_SIT_INEFFECTIVE_OTHER (TID_SIT_IO)
- Default SIT definitions can be modified, **except for** the following SIT sequences: TID_SIT_ANY, TID_SIT_IO, TID_SIT_NC_INTERLATA, and TID_SIT_RO_INTERLATA. For more information, see [Section 7.8, “Modifying Default Call Progress Analysis Tone Definitions”](#), on page 46.
- For TID_SIT_ANY, the frequency and time of the first and second segments are open; that is, they are ignored. Only the frequency of the third segment is relevant. This catch-all SIT sequence definition is intended to cover SIT sequences that fall outside the range of the defined SIT sequences.

7.6 Media Tone Detection

Media tone detection in call progress analysis is discussed in the following topics:

- [Positive Voice Detection \(PVD\)](#)
- [Positive Answering Machine Detection \(PAMD\)](#)

7.6.1 Positive Voice Detection (PVD)

Positive voice detection (PVD) can detect when a call has been answered by determining whether an audio signal is present that has the characteristics of a live or recorded human voice. This provides a very precise method for identifying when a connect occurs.

The `ca_intflg` field in `DX_CAP` enables/disables PVD. For information on enabling PVD, see [Section 7.4.3, “Setting Up Call Progress Analysis Parameters in DX_CAP”](#), on page 36.

PVD is especially useful in those situations where no other method of answer supervision is available, and where the cadence is not clearly broken for cadence detection to identify a connect (for example, when the nonsilence of the cadence is immediately followed by the nonsilence of speech).

If the `ATDX_CONNTYPE()` function returns `CON_PVD`, the connect was due to positive voice detection.

7.6.2 Positive Answering Machine Detection (PAMD)

Whenever PAMD is enabled, positive voice detection (PVD) is also enabled.

The `ca_intflg` field in `DX_CAP` enables/disables PAMD and PVD. For information on enabling PAMD, see [Section 7.4.3, “Setting Up Call Progress Analysis Parameters in DX_CAP”](#), on page 36.

When enabled, detection of an answering machine will result in the termination of call analysis with the reason `CR_CNCT` (connected); the connection type returned by the `ATDX_CONNTYPE()` function will be `CON_PAMD`.

The following `DX_CAP` fields govern positive answering machine detection:

`ca_pamd_spdval`

PAMD Speed Value: To distinguish between a greeting by a live human and one by an answering machine, use one of the following settings:

- `PAMD_FULLL` – look at the greeting (long method). The long method looks at the full greeting to determine whether it came from a human or a machine. Using `PAMD_FULLL` gives a very accurate determination; however, in situations where a fast decision is more important than accuracy, `PAMD_QUICK` might be preferred.
- `PAMD_QUICK` – look at connect only (quick method). The quick method examines only the events surrounding the connect time and makes a rapid judgment as to whether or not an answering machine is involved.
- `PAMD_ACCU` – look at the greeting (long method) and use the most accuracy for detecting an answering machine. This setting provides the most accurate evaluation. It

detects live voice as accurately as PAMD_FULLL but is more accurate than PAMD_FULLL (although slightly slower) in detecting an answering machine. Use the setting PAMD_ACCU when accuracy is more important than speed.

Default value: PAMD_ACCU

The recommended setting for the call analysis parameter structure (DX_CAP) ca_pamd_spdval field is PAMD_ACCU.

ca_pamd_failtime

maximum time to wait for positive answering machine detection or positive voice detection after a cadence break. Default Value: 400 (in 10 msec units).

7.7 Default Call Progress Analysis Tone Definitions

Table 5 provides the range of values for default tone definitions. These default tone definitions are used in call progress analysis. Amplitudes are given in dBm, frequencies in Hz, and duration in 10 msec units. A dash in a table cell means not applicable.

- Notes:**
1. Voice API functions are provided to manipulate the tone definitions in this table (see [Section 7.8, “Modifying Default Call Progress Analysis Tone Definitions”](#), on page 46). However, not all the functionality provided by these tones is available through the Dialogic® Voice API. You may need to use the Dialogic® Global Call API to access the functionality, for example, in the case of disconnect tone detection.
 2. An On Time maximum value of 0 indicates that this is a continuous tone. For example, TID_DIAL_LCL has an On Time range of 10 to 0. This means that the tone is on for 100 msec. The minimum requirement for detecting a tone is that it must be continuous for at least 100 msec (10 in 10 msec units) after it is detected.
 3. A single tone proxy for a dual tone (twin tone) can help improve the accuracy of dual tone detection in some cases. For more information, see [Section 7.8.4, “Rules for Using a Single Tone Proxy for a Dual Tone”](#), on page 48.

Table 5. Default Call Progress Analysis Tone Definitions

Tone ID	Freq1 (in Hz)	Freq2 (in Hz)	On Time (in 10 msec)	Off Time (in 10 msec)	Reps	Twin Tone Freq (Hz)
TID_BUSY1	450 - 510	590 - 650	30 - 100	30 - 100	2	0
TID_BUSY2	450 - 510	590 - 650	10 - 40	10 - 40	2	0
TID_DIAL_INTL	300 - 380	400 - 480	100 - 0	-	1	300 - 480
TID_DIAL_LCL	300 - 380	400 - 480	10 - 0	-	1	0
TID_FAX2	2000 - 2300	-	10 - 0	-	1	-
TID_RNGBK1	350 - 550	350 - 550	75 - 300	0 - 800	1	350 - 550
TID_RNGBK2 (segment 0)	350 - 550	350 - 550	20 - 100	20 - 100	1	350 - 550
TID_RNGBK2 (segment 1)	350 - 550	350 - 550	20 - 100	100 - 600	1	350 - 550

Table 5. Default Call Progress Analysis Tone Definitions (Continued)

Tone ID	Freq1 (in Hz)	Freq2 (in Hz)	On Time (in 10 msec)	Off Time (in 10 msec)	Reps	Twin Tone Freq (Hz)
TID_DISCONNECT	360 - 410	430 - 440	30 - 60	30 - 60	1	360 - 440
TID_FAX1	1050 - 1150	-	10 - 60	-	1	-
TID_FAX2	2000 - 2300	-	10 - 0	-	1	-
TID_RNGBK1	350 - 550	350 - 550	75 - 300	0 - 800	1	350 - 550
TID_RNGBK2 (segment 0)	350 - 550	350 - 550	20 - 100	20 - 100	1	350 - 550
TID_RNGBK2 (segment 1)	350 - 550	350 - 550	20 - 100	100 - 600	1	350 - 550

7.8 Modifying Default Call Progress Analysis Tone Definitions

Call progress analysis tones are maintained in the firmware. More information on tone definitions is provided in the following topics:

- [API Functions for Manipulating Tone Definitions](#)
- [TONE_DATA Data Structure](#)
- [Rules for Modifying a Tone Definition](#)
- [Rules for Using a Single Tone Proxy for a Dual Tone](#)
- [Steps to Modify a Tone Definition](#)

7.8.1 API Functions for Manipulating Tone Definitions

The following Dialogic® Voice API functions are used to manipulate the default tone definitions shown in Table 5, “Default Call Progress Analysis Tone Definitions”, on page 45 and some, but not all, of the default tone definitions shown in Table 4, “Special Information Tone Sequences”, on page 43.

Note: Default SIT definitions can be modified, **except for** the following SIT sequences: TID_SIT_ANY, TID_SIT_IO, TID_SIT_NC_INTERLATA, and TID_SIT_RO_INTERLATA.

dx_querytone()

gets tone information for a specific call progress tone

dx_deletetone()

deletes a specific call progress tone

dx_createtone()

creates a new tone definition for a specific call progress tone

7.8.2 TONE_DATA Data Structure

The TONE_DATA structure contains tone information for a specific call progress tone. This structure contains a nested array of TONE_SEG substructures. A maximum of six TONE_SEG substructures can be specified. The TONE_DATA structure specifies the following key information:

TONE_SEG.structver

Specifies the version of the TONE_SEG structure. Used to ensure that an application is binary compatible with future changes to this data structure.

TONE_SEG.tn_dflag

Specifies whether the tone is dual tone or single tone. Values are 1 for dual tone and 0 for single tone.

TONE_SEG.tn1_min

Specifies the minimum frequency in Hz for tone 1.

TONE_SEG.tn1_max

Specifies the maximum frequency in Hz for tone 1.

TONE_SEG.tn2_min

Specifies the minimum frequency in Hz for tone 2.

TONE_SEG.tn2_max

Specifies the maximum frequency in Hz for tone 2.

TONE_SEG.tn_twinmin

Specifies the minimum frequency in Hz of the single tone proxy for the dual tone.

TONE_SEG.tn_twinmax

Specifies the maximum frequency in Hz of the single tone proxy for the dual tone.

TONE_SEG.tnon_min

Specifies the debounce minimum ON time in 10 msec units.

TONE_SEG.tnon_max

Specifies the debounce maximum ON time in 10 msec units.

TONE_SEG.tnoff_min

Specifies the debounce minimum OFF time in 10 msec units.

TONE_SEG.tnoff_max

Specifies the debounce maximum OFF time in 10 msec units.

TONE_DATA.structver

Specifies the version of the TONE_DATA structure. Used to ensure that an application is binary compatible with future changes to this data structure.

TONE_DATA.tn_rep_cnt

Specifies the debounce rep count.

TONE_DATA.numofseg

Specifies the number of segments for a multi-segment tone.

7.8.3 Rules for Modifying a Tone Definition

Consider the following rules and guidelines for modifying default tone definitions using the Dialogic® Voice API library:

- You must issue **dx_querytone()**, **dx_deletetone()**, and **dx_createtone()** in this order, one tone at a time, for each tone definition to be modified.
- Attempting to create a new tone definition before deleting the current call progress tone will result in an EDX_TNQUERYDELETE error.
- When **dx_querytone()**, **dx_deletetone()**, or **dx_createtone()** is issued in asynchronous mode and is immediately followed by another similar call prior to completion of the previous call on the same device, the subsequent call will fail with device busy.
- Only default call progress analysis tones and SIT sequences are supported for these three functions. For a list of these tones, see [Table 4, “Special Information Tone Sequences”](#), on page 43 and [Table 5, “Default Call Progress Analysis Tone Definitions”](#), on page 45.
- These three Dialogic® Voice API functions are provided to manipulate the call progress analysis tone definitions. However, not all the functionality provided by these tones is available through the Dialogic® Voice API. You may need to use the Dialogic® Global Call API to access the functionality, for example, in the case of disconnect tone detection.
- If the application deletes all the default call progress analysis tones in a particular set (where a set is defined as busy tones, dial tones, ringback tones, fax tones, disconnect tone, and special information tones), the set itself is deleted from the board and call progress analysis cannot be performed successfully. Therefore, you must have at least one tone defined in each tone set in order for call progress analysis to perform successfully.

7.8.4 Rules for Using a Single Tone Proxy for a Dual Tone

A single tone proxy (also called a twin tone) acts as a proxy for a dual tone. A single tone proxy can be defined when you run into difficulty detecting a dual tone. This situation can arise when the two frequencies of the dual tone are close together, are very short tones, or are even multiples of each other. In these cases, the dual tone might be detected as a single tone. A single tone proxy can help improve the detection of the dual tone by providing an additional tone definition.

The TONE_SEG.tn_twinmin field defines the minimum frequency of the tone and TONE_SEG.tn_twinmax field defines the maximum frequency of the tone.

Consider the following guidelines when creating a single tone proxy:

- It is recommended that you add at least 60 Hz to the top of the dual tone range and subtract at least 60 Hz from the bottom of the dual tone range. For example:
Freq1 (Hz): 400 - 500
Freq 2 (Hz): 600 - 700
Twin tone freq (Hz): 340 - 760
- Before using the TONE_DATA structure in a function call, set any unused fields in the structure to zero to prevent possible corruption of data in the allocated memory space. This guideline is applicable to unused fields in any data structure.

7.8.5 Steps to Modify a Tone Definition

To modify a default tone definition using the voice API library, follow these steps:

Note: This procedure assumes that you have already opened the board device handle in your application. To get the board name in the form *brdBn*, use the **SRLGetPhysicalBoardName()** function. This function and other device mapper functions return information about the structure of the system. For more information, see the *Dialogic® Standard Runtime Library API Library Reference*.

1. Get the tone information for the call progress tone to be modified using **dx_querytone()**. After the function completes successfully, the relevant tone information is contained in the TONE_DATA structure.
2. Delete the current call progress tone using **dx_deletetone()** before creating a new tone definition.
3. Create a new tone definition for the call progress tone using **dx_createtone()**. Specify the new tone information in the TONE_DATA structure.
4. Repeat steps 1-3 in this order **for each tone** to be modified.

7.9 Call Progress Analysis Errors

If **ATDX_CPTERM()** returns CR_ERROR, you can use **ATDX_CPERROR()** to determine the call progress analysis error that occurred. For details on these functions, see the *Dialogic® Voice API Library Reference*.

Call Progress Analysis

This chapter discusses playback and recording features supported by the Dialogic® Voice API library. The following topics are discussed:

- [Overview of Recording and Playback](#) 51
- [Digital Recording and Playback](#) 51
- [Play and Record Functions](#) 52
- [Play and Record Convenience Functions](#) 52
- [Voice Encoding Methods](#) 53
- [G.726 Voice Coder](#) 53
- [Transaction Record](#) 54
- [Silence Compressed Record](#) 55
- [Recording with the Voice Activity Detector](#) 56
- [Streaming to Board](#) 58

8.1 Overview of Recording and Playback

The primary voice processing operations provided by a Dialogic® voice board include:

- recording: digitizing and storing human voice
- playback: retrieving, converting, and playing the stored, digital information to reconstruct the human voice.

The following features related to voice recording and playback operation are documented in other chapters in this document:

- Controlling when a playback or recording terminates using I/O termination conditions is documented in [Section 6.1.2, “Setting Termination Conditions for I/O Functions”](#), on page 27.
- Controlling the volume when messages are played back is documented in [Chapter 9, “Speed and Volume Control”](#).

8.2 Digital Recording and Playback

In digital speech recording, the voice board converts the human voice from a continuous sound wave, or analog signal, into a digital representation. The Dialogic® voice board does this by frequently sampling the amplitude of the sound wave at individual points in the speech signal.

Recording and Playback

The accuracy, and thus the quality, of the digital recording is affected by:

- the sampling rate (number of samples per second), also called digitization rate
- the precision, or resolution, of each sample (the amount of data that is used to represent 1 sample).

If the samples are taken at a greater frequency, the digital representation will be more accurate and the voice quality will be greater. Likewise, if more bits are used to represent the sample (higher resolution), the sample will be more accurate and the voice quality will be greater.

In digital speech playback, the voice board reconstructs the speech signal by converting the digitized voice back into analog voltages. If the voice data is played back at the same rate at which it was recorded, an approximation of the original speech will result.

8.3 Play and Record Functions

The C language function library includes several functions for recording and playing audio data, such as **dx_rec()**, **dx_reciottdata()**, **dx_play()**, and **dx_playiottdata()**. Recording takes audio data from a specified channel and encodes it for storage in memory, in a file on disk, or on a custom device. Playing decodes the stored audio data and plays it on the specified channel. The storage location is one factor in determining which record and play functions should be used. The storage location affects the access speed for retrieving and storing audio data.

One or more of the following data structures are used in conjunction with certain play and record functions: **DV_TPT** to specify a termination condition for the function, **DX_IOTT** to identify a source or destination for the data, and **DX_XPB** to specify the file format, data format, sampling rate, and resolution.

For detailed information about play and record functions, which are also known as I/O functions, see the *Dialogic® Voice API Library Reference*.

8.4 Play and Record Convenience Functions

Several convenience functions are provided to make it easier to implement play and record functionality in an application. Some examples are: **dx_playf()**, **dx_playvox()**, **dx_playwav()**, **dx_recf()**, and **dx_recvox()**. These functions are specific cases of the **dx_play()** and **dx_rec()** functions and run in synchronous mode.

For example, **dx_playf()** performs a playback from a single file by specifying the filename. The same operation can be done using **dx_play()** and specifying a **DX_IOTT** structure with only one entry for that file. Using **dx_playf()** is more convenient for a single file playback because you do not have to set up a **DX_IOTT** structure for the one file and the application does not need to open the file. **dx_recf()** provides the same single file convenience for the **dx_rec()** function.

For a complete list of I/O convenience functions and function reference information, see the *Dialogic® Voice API Library Reference*.

8.5 Voice Encoding Methods

A digitized audio recording is characterized by several parameters as follows:

- the number of samples per second, or sampling rate
- the number of bits used to store a sample, or resolution
- the rate at which data is recorded or played

There are many encoding and storage schemes available for digitized voice. The supported voice encoding methods or data formats are listed in Table 6.

Table 6. Voice Encoding Methods

Digitizing Method	Sampling Rate (kHz)	Resolution (Bits)	Bit Rate (Kbps)	File Format
OKI ADPCM	6	4	24	VOX, WAVE
OKI ADPCM	8	4	32	VOX, WAVE
G.711 PCM A-law and mu-law	6	8	48	VOX, WAVE
G.711 PCM A-law and mu-law	8	8	64	VOX, WAVE
Linear PCM	8	16	128	VOX, WAVE
Linear PCM	11	8	88	VOX, WAVE
GSM 6.10 full rate (Microsoft format)	8	(value ignored)	13	VOX, WAVE
GSM 6.10 full rate (TIPHON format)	8	(value ignored)	13	VOX
G.726 bit exact	8	2	16	VOX, WAVE
G.726 bit exact	8	4	32	VOX, WAVE

8.6 G.726 Voice Coder

G.726 is an ITU-T recommendation that specifies an adaptive differential pulse code modulation (ADPCM) technique for recording and playing back audio files. It is useful for applications that require speech compression, encoding for noise immunity, and uniformity in transmitting voice and data signals.

The voice library provides support for a G.726 bit exact voice coder that is compliant with the ITU-T G.726 recommendation.

Audio encoded in the G.726 bit exact format complies with Voice Profile for Internet Mail (VPIM), a communications protocol that makes it possible to send and receive messages from disparate messaging systems over the Internet. G.726 bit exact is the audio encoding and decoding standard supported by VPIM.

Recording and Playback

VPIM follows the little endian ordering. The 4-bit code words of the G.726 encoding must be packed into octets/bytes as follows:

- The first code word (A) is placed in the four least significant bits of the first octet, with the least significant bit (LSB) of the code word (A0) in the least significant bit of the octet.
- The second code word (B) is placed in the four most significant bits of the first octet, with the most significant bit (MSB) of the code word (B3) in the most significant bit of the octet.
- Subsequent pairs of the code words are packed in the same way into successive octets, with the first code word of each pair placed in the least significant four bits of the octet. It is preferable to extend the voice sample with silence such that the encoded value consists of an even number of code words. However, if the voice sample consists of an odd number of code words, then the last code word will be discarded.

The G.726 encoding for VPIM is illustrated here:

```
+---+---+---+---+---+---+---+---+
|B3|B2|B1|B0|A3|A2|A1|A0|
+---+---+---+---+---+---+---+---+
MSB -> | 7| 6| 5| 4| 3| 2| 1| 0| <- LSB
+---+---+---+---+---+---+---+---+
32K ADPCM / Octet Mapping
```

For more information on G.726 and VPIM, see RFC 3802 on the Internet Engineering Task Force (IETF) website at <http://www.ietf.org>.

To use the G.726 voice coder, specify the coder in the `DX_XPB` structure. Then use `dx_playiottdata()` and `dx_reciottdata()` functions to play and record with this coder. Alternatively, you can also use `dx_playvox()` and `dx_recvox()` convenience functions.

To determine the voice resource handles used with the play and record functions, use SRL device mapper functions to return information about the structure of the system.

See the *Dialogic® Voice API Library Reference* for more information on voice functions and data structures. See the *Dialogic® Standard Runtime Library API Library Reference* for more information on SRL functions.

8.7 Transaction Record

Transaction record enables the recording of a two-party conversation by allowing two time-division multiplexing (TDM) bus time slots from a single channel to be recorded. This feature is useful for call center applications where it is necessary to archive a verbal transaction or record a live conversation. A live conversation requires two time slots on the TDM bus, but Dialogic® voice boards today can only record one time slot at a time. No loss of channel density is realized with this feature. Voice activity on two channels can be summed and stored in a single file, or in a combination of files, devices, and/or memory.

Use the following function for transaction record:

dx_mreciottdata()

records voice data from two channels to a data file, memory, or custom device

See the *Dialogic® Voice API Library Reference* for a full description of this function. For information on running transaction record on a single board, see the technical note posted on the Dialogic support web site at:

<http://www.dialogic.com/support/helpweb/dxall/tnotes/legacy/dlsoft/tn253.htm>.

8.8 Silence Compressed Record

The silence compressed record (SCR) feature is discussed in more detail in the following topics:

- [Overview](#)
- [Enabling](#)
- [Encoding Methods Supported](#)
- [Configuring](#)

8.8.1 Overview

The silence compressed record feature (SCR) enables recording with silent pauses eliminated. This results in smaller recorded files with no loss of intelligibility.

The SCR algorithm operates on one msec blocks of speech and uses a two-fold approach to determine whether a sample is speech or silence. Two probability of speech values are calculated using a zero crossing algorithm and an energy detection algorithm. These values are put together to calculate a combined probability of speech.

The energy detection algorithm allows you to modify the background noise threshold range. Signals above the high threshold are declared speech, and signals below the low threshold are declared silence.

Speech or silence is declared based on the previous sample, the current combined probability of speech in relation to the speech probability threshold and silence probability threshold parameters, and the trailing silence parameter.

8.8.2 Enabling

Use **dx_setparm()** and the define, **DXCH_SCRFEATURE**, to turn SCR on and off. Once enabled, voice record functions automatically record with SCR.

Recording and Playback

For DXCH_SCRFEATURE, the valid values are:

Define	Bytes	Read/Write	Default	Description
DXCH_SCRFEATURE	2	R/W	-	Silence Compressed Record (SCR). Valid values are: <ul style="list-style-type: none">• DXCH_SCRDISABLED – SCR feature disabled• DXCH_SCREENABLED – SCR feature enabled

Note: This will override any board-level settings.

8.8.3 Encoding Methods Supported

The following encoding algorithms and sampling rates are supported in SCR:

- OKI ADPCM, 6 kHz with 4-bit samples (24 kbps) and 8 kHz with 4-bit samples (32 kbps), VOX and WAVE file formats
- linear PCM, 8 kHz sampling 64 Kbps (8 bits), 8 kHz sampling 128 Kbps (16 bits)
- G.711 PCM, 6 kHz with 8-bit samples (48 kbps) and 8 kHz with 8-bit samples (64 kbps) using A-law or mu-law coding, VOX and WAVE file formats
- G.726 bit-exact voice coder at 8 kHz with 2-, 3-, 4-, or 5-bit samples (16, 24, 32, 40 kbps), VOX and WAVE file formats

8.8.4 Configuring

To change the parameter values for SCR, you must add new parameters in the [Encoder] section of the CONFIG file. For more information about configuring SCR, see the Configuration Guide.

8.9 Recording with the Voice Activity Detector

Recording with the voice activity detector is discussed in the following topics:

- [Overview](#)
- [Enabling](#)
- [Encoding Methods Supported](#)

8.9.1 Overview

The `dx_reciottdata()` function, used to record voice data, has two modes that work with the voice activity detector. One mode enables voice activity detection with event notification upon detection. The second mode adds initial silence compression on the line before voice energy is detected; if initial silence is greater than the default allowable amount of silence, the amount in excess is eliminated. This mode uses the same algorithm as the silence compressed record (SCR) feature described in [Section 8.8, “Silence Compressed Record”](#), on page 57.

The voice activity detector (VAD) is a component in the voice software that examines the incoming signal and determines if the signal contains significant energy and is likely to be voice.

8.9.2 Enabling

The modes related to the voice activity detector are specified in the **mode** parameter of the **dx_reciottdata()** function. They are:

RM_VADNOTIFY

generates an event, TDX_VAD, on detection of voice energy during the recording operation

Note: TDX_VAD does not indicate function termination; it is an unsolicited event. Do not confuse this event with the TEC_VAD event which is used in the continuous speech processing (CSP) library.

RM_ISCR

adds initial silence compression to the VAD capability. Initial silence here refers to the amount of silence on the line *before* voice activity is detected. When using RM_ISCR, the default value for the amount of initial silence allowable is 3 seconds. Any initial silence longer than that will be eliminated to the default allowable amount. This default value can be changed by modifying a parameter in the .config file for the board and then generating a new .fcd file. The 0x416 parameter must be added in the [encoder] section of the .config file. For details on using this parameter, see the Configuration Guide.

Note: The RM_ISCR mode can only be used in conjunction with RM_VADNOTIFY.

When these two modes are used together, no data is recorded as output until voice activity is detected on the line. The TDX_VAD event indicates the initiation of voice. The output file will be empty before voice activity is detected, although some initial silence may be included as specified in the .fcd file.

To enable these modes, OR them to the **mode** parameter. For example:

```
t_Return=dx_reciottdata(DevHandle, Iott, Tpt, &t_Xpb, EV_ASYNC|RM_VADNOTIFY);
t_Return=dx_reciottdata(DevHandle, Iott, Tpt, &t_Xpb, EV_ASYNC|RM_VADNOTIFY|RM_ISCR);
```

Note: The **dx_reciottdata()** function does not perform echo-cancelled streaming. For automatic speech recognition applications, use record or streaming functions in the Dialogic® Continuous Speech Processing (CSP) API library. For more information, see the *Dialogic® Continuous Speech Processing API Programming Guide* and *Dialogic® Continuous Speech Processing API Programming Guide*.

8.9.3 Encoding Methods Supported

The following encoding algorithms and sampling rates are supported for recording with the voice activity detector:

- OKI ADPCM, 6 kHz with 4-bit samples (24 kbps) and 8 kHz with 4-bit samples (32 kbps), VOX and WAVE file formats
- linear PCM, 8 kHz sampling 64 Kbps (8 bits), 8 kHz sampling 128 Kbps (16 bits), VOX and WAVE file formats

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- G.711 PCM, 6 kHz with 8-bit samples (48 kbps) and 8 kHz with 8-bit samples (64 kbps) using A-law or mu-law coding, VOX and WAVE file formats
- G.726 bit-exact voice coder at 8 kHz with 2-, 3-, 4-, or 5-bit samples (16, 24, 32, 40 kbps), VOX and WAVE file formats

8.10 Streaming to Board

The streaming to board feature is discussed in the following topics:

- [Streaming to Board Overview](#)
- [Streaming to Board Functions](#)
- [Implementing Streaming to Board](#)
- [Streaming to Board Guidelines](#)

8.10.1 Streaming to Board Overview

The streaming to board feature provides a way to stream data in real time to a network interface. Unlike the standard voice play feature (store and forward method), data can be streamed with little delay as the amount of initial data required to start the stream is configurable. The streaming to board feature is essential for applications such as text-to-speech, distributed prompt servers, and IP gateways.

The streaming to board feature uses a circular stream buffer to hold data, provides configurable high and low water mark parameters, and generates events when those water marks are reached.

8.10.2 Streaming to Board Functions

The following functions are used by the streaming to board feature:

- dx_OpenStreamBuffer()**
creates and initializes a circular stream buffer
- dx_SetWaterMark()**
sets high and low water marks for the circular stream buffer
- dx_PutStreamData()**
places data into the circular stream buffer
- dx_GetStreamInfo()**
retrieves information about the circular stream buffer
- dx_ResetStreamBuffer()**
resets internal data for a circular stream buffer
- dx_CloseStreamBuffer()**
deletes a circular stream buffer

For details on these functions, see the *Dialogic® Voice API Library Reference*.

8.10.3 Implementing Streaming to Board

Perform the following steps to implement streaming to board in your application:

Note: These steps do not represent every task that must be performed to create a working application but are intended as general guidelines for implementing streaming to board.

1. Decide on the size of the circular stream buffer. This value is used as input to the **dx_OpenStreamBuffer()** function. To determine the circular stream buffer size, see [Section 8.10.4, “Streaming to Board Guidelines”](#), on page 59.
2. Based on the circular stream buffer and the bulk queue buffer size, decide on values for the high and low water marks for the circular stream buffer. To determine high and low water mark values, see [Section 8.10.4, “Streaming to Board Guidelines”](#), on page 59.
3. Initialize and create a circular stream buffer using **dx_OpenStreamBuffer()**.
4. Set the high and low water marks using **dx_SetWaterMark()**.
5. Start the play using **dx_playiottdata()** or **dx_play()** in asynchronous mode with the `io_type` field in `DX_IOTT` data structure set to `IO_STREAM`.
6. Put data in the circular stream buffer using **dx_PutStreamData()**.
7. Wait for events.

The `TDX_LOW WATER` event is generated every time data in the buffer falls below the low water mark. The `TDX_HIGH WATER` event is generated every time data in the buffer is above the high water mark. The application receives `TDX_LOW WATER` and `TDX_HIGH WATER` events regardless of whether or not **dx_SetWaterMark()** is used in your application. These events are generated when there is a play operation with this buffer and are reported on the device that is performing the play. If there is no active play, the application will **not** receive any of these events.

`TDX_PLAY` indicates that play has completed.

8. When all files are played, issue **dx_CloseStreamBuffer()**.

8.10.4 Streaming to Board Guidelines

On Windows®

Consider the following usage guidelines when implementing streaming to board in your application on a Windows® system:

- You can create as many circular stream buffers as needed on a channel. You can use more than one circular stream buffer per play via the `DX_IOTT` structure. In this case, specify that the data ends in one buffer using the `STREAM_EOD` flag so that the play can process the next `DX_IOTT` structure in the chain.
- In general, the larger you define the circular stream buffer size, the better. Factors to take into consideration include the average input file size, the amount of memory on your system, the total number of channels in your system, and so on. Having an optimal circular stream buffer size results in the high and low water marks being reached less often. In a well-tuned system, the high and low water marks should rarely be reached.
- When adjusting circular stream buffer sizes, be aware that you must also adjust the high and low water marks accordingly.

Recording and Playback

- Recommendation for the high water mark: it should be based on the following:
size of the circular stream buffer minus two times the size of the bulk queue buffer
For example, if the circular stream buffer is 100 kbytes, and the bulk queue buffer size is 8 kbytes, set the high water mark to 84 kbytes. The bulk queue buffer size is set through the **dx_setchxfercnt()** function.
- Recommendation for the low water mark:
 - If the bulk queue buffer size is less than 8 kbytes, the low water mark should be four times the size of the bulk queue buffer size.
 - If the bulk queue buffer size is greater than 8 kbytes and less than 16 kbytes, the low water mark should be three times the size of the bulk queue buffer size.
 - If the bulk queue buffer size is greater than 16 kbytes, the low water mark should be two times the size of the bulk queue buffer size.
- When a TDX_LOWWATER event is received, continue putting data in the circular stream buffer. Remember to set STREAM_EOD flag to EOD on the last piece of data.
- When a TDX_HIGHWATER event is received, stop putting data in the circular stream buffer. If using a text-to-speech (TTS) engine, you will have to stop the engine from sending more data. If you cannot control the output of the TTS engine, you will need to control the input to the engine.
- It is recommended that you enable the TDX_UNDERRUN event to notify the application of firmware underrun conditions on the board. Specify DM_UNDERRUN in **dx_setevtmsk()**.

On Linux

Consider the following usage guidelines when implementing streaming to board in your application on a Linux system:

- You can create as many circular stream buffers as needed on a channel. You can use more than one circular stream buffer per play via the DX_IOTT structure. In this case, specify that the data ends in one buffer using the STREAM_EOD flag so that the play can process the next DX_IOTT structure in the chain.
- The bulk queue buffer specifies the size of the buffer used to transfer voice data between the application and the driver. This buffer is set to 16 kbytes and cannot be modified; the **dx_setchxfercnt()** function, which is used to modify the bulk queue buffer size, is not currently supported.
- In general, the larger you define the circular stream buffer size, the better. Factors to take into consideration include the average input file size, the amount of memory on your system, the total number of channels in your system, and so on. Having an optimal circular stream buffer size results in the high and low water marks being reached less often. In a well-tuned system, the high and low water marks should rarely be reached.
- When adjusting circular stream buffer sizes, be aware that you must also adjust the high and low water marks accordingly.
- Recommendation for the high water mark: it should be based on the following:
size of the circular stream buffer minus two times the size of the bulk queue buffer
For example, if the circular stream buffer is 500 kbytes, and the bulk queue buffer size is 16 kbytes, set the high water mark to 468 kbytes ($500-32=468$).

- Recommendation for the low water mark: it should be two times the size of the bulk queue buffer size.

Based on the previous example, since the bulk queue buffer size is 16 kbytes, set the low water mark to 32 kbytes.

- When a TDX_LOWWATER event is received, continue putting data in the circular stream buffer. Remember to set STREAM_EOD flag to EOD on the last piece of data.
- When a TDX_HIGHWATER event is received, stop putting data in the circular stream buffer. If using a text-to-speech (TTS) engine, you will have to stop the engine from sending more data. If you cannot control the output of the TTS engine, you will need to control the input to the engine.
- It is recommended that you enable the TDX_UNDERRUN event to notify the application of firmware underrun conditions on the board. Specify DM_UNDERRUN in **dx_setevtmsk()**.

Recording and Playback

This chapter describes how to control the speed and volume of play on a channel. The following topics are discussed:

- [Speed and Volume Control Overview](#) 63
- [Speed and Volume Convenience Functions](#) 64
- [Speed and Volume Adjustment Functions](#) 64
- [Speed and Volume Modification Tables](#) 64
- [Play Adjustment Digits](#) 68
- [Setting Play Adjustment Conditions](#) 68
- [Explicitly Adjusting Speed and Volume](#) 68

9.1 Speed and Volume Control Overview

The voice software contains functions and data structures to control the speed and volume of play on a channel. This allows an end user to control the speed or volume of a message by entering a DTMF tone, for example.

Note: Before using the speed control feature, you must enable this feature in the [decoder] section of the CONFIG file. The speed control feature is disabled by default to preserve MIPS usage. For more information on enabling speed control, see the Configuration Guide.

Speed can be controlled on playbacks using the following encoding methods:

- OKI ADPCM 24 kbps and 32 kbps
- G.711 PCM A-law or mu-law encoding 48 kbps and 64 kbps
- linear PCM 128 kbps

If an attempt is made to adjust speed on an unsupported board, the firmware will ignore the request and play will continue at normal speed.

Volume can be controlled on all playbacks regardless of the encoding algorithm. For a list of supported encoding methods, see [Section 8.5, “Voice Encoding Methods”](#), on page 53.

9.2 Speed and Volume Convenience Functions

The convenience functions set a digit that will adjust speed or volume, but do not use any data structures. These convenience functions will only function properly if you use the default settings of the speed or volume modification tables. These functions assume that the modification tables have not been modified. The convenience functions are:

dx_addspddig()

adds a digit that will modify speed by a specified amount

dx_addvoldig()

adds a digit that will modify volume by a specified amount

See the *Dialogic® Voice API Library Reference* for detailed information about these functions.

9.3 Speed and Volume Adjustment Functions

Speed or volume can be adjusted explicitly or can be set to adjust in response to a preset condition, such as a specific digit. For example, speed could be set to increase a certain amount when “1” is pressed on the telephone keypad. The functions used for speed and volume adjustment are:

dx_setsvcond()

Sets conditions that adjust speed or volume. Use this function to adjust speed or volume in response to a DTMF digit or start of play.

dx_adjsv()

Adjusts speed or volume explicitly. Use this function if your adjustment condition is not a digit or start of play. For example, the application could call this function after detecting a spoken word (voice recognition) or a certain key on the keyboard.

See the *Dialogic® Voice API Library Reference* for detailed information about these functions.

9.4 Speed and Volume Modification Tables

Each channel has a speed or volume modification table for play speed or play volume adjustments. Except for the value of the settings, the table is the same for speed and volume.

Each speed or volume modification table (SVMT) table has 21 entries, 20 that allow for a maximum of 10 increases and decreases in speed or volume. The entry in the middle of the table is referred to as the “origin” entry that represents normal speed or volume. The normal speed or volume is how playback occurs when the speed and volume control feature is not used. See [Table 7, “Default Speed Modification Table”](#), on page 66 and [Table 8, “Default Volume Modification Table”](#), on page 67.

The origin, or normal speed or volume, is the basis for all settings in the table. Typically, the origin is set to 0. Speed and volume increases or decreases by moving up or down the tables. Other entries in the table specify a speed or volume setting in terms of a deviation from normal. For example, if a

speed modification table (SMT) entry is -10, this value represents a 10% decrease from the normal speed.

Although the origin is typically set to normal speed/volume, changing the setting of the origin does not affect the other settings, because all values in the SVMT are based on a deviation from normal speed/volume.

Speed and volume control adjustments are specified by moving the current speed/volume pointer in the table to another SVMT table entry; this translates to increasing or decreasing the current speed/volume to the value specified in the table entry.

A speed/volume adjustment stays in effect until the next adjustment on that channel or until a system reset.

The SVMT is like a 21-speed bicycle. You can select the gear ratio for each of the 21 speeds before you go for a ride (by changing the values in the SVMT). And you can select any gear once you are on the bike, like adjusting the current speed/volume to any setting in the SVMT.

To change the default values of the speed or volume modification table, use the **dx_setsvmt()** function, which in turn uses the DX_SVMT data structure. To return the current values of a table to the DX_SVMT structure, use **dx_getsvmt()**. The DX_SVCB data structure uses this table when setting adjustment conditions.

See the *Dialogic® Voice API Library Reference* for detailed information about these functions and data structures.

Adjustments to speed or volume are made by **dx_adjsv()** and **dx_setsvcond()** according to the speed or volume modification table settings. These functions adjust speed or volume to one of the following:

- a specified level (that is, to a specified absolute position in the speed table or volume table)
- a change in level (that is, by a specified number of steps up or down in the speed table or volume table)

For example, by default, each entry in the volume modification table is equivalent to 2 dB from the origin. Volume could be decreased by 2 dB by specifying position 1 in the table, or by moving one step down from the origin.

Speed and Volume Control

The default speed modification table is shown in Table 7.

Table 7. Default Speed Modification Table

Table Entry	Default Value (%)	Absolute Position
decrease[0]	-128 (80h)	-10
decrease[1]	-128 (80h)	-9
decrease[2]	-128 (80h)	-8
decrease[3]	-128 (80h)	-7
decrease[4]	-128 (80h)	-6
decrease[5]	-50	-5
decrease[6]	-40	-4
decrease[7]	-30	-3
decrease[8]	-20	-2
decrease[9]	-10	-1
origin	0	0
increase[0]	+10	1
increase[1]	+20	2
increase[2]	+30	3
increase[3]	+40	4
increase[4]	+50	5
increase[5]	-128 (80h)	6
increase[6]	-128 (80h)	7
increase[7]	-128 (80h)	8
increase[8]	-128 (80h)	9
increase[9]	-128 (80h)	10

Consider the following usage information on the speed modification table:

- Each entry in the table is a percentage deviation from the default play speed (“origin”). For example, the decrease[6] position reduces speed by 40%. This is four steps from the origin.
- The total speed modification range is from -50% to +50%. In this table, the lowest position used is the decrease[5] position. The remaining decrease fields are set to -128 (80h). If these “nonadjustment” positions are selected, the default action is to play at the decrease[5] speed.
- These fields can be reset, as long as no values lower than -50 are used. For example, you could spread the 50% speed decrease over 10 steps rather than 5. Similarly, you could spread the 50% speed increase over 10 steps rather than 5.
- The default entries for index values -10 to -6 and +6 to +10 are -128 which represent a null-entry. To customize the table entries, you must use the `dx_setsvmt()` function.
- When adjustment is associated with a DTMF digit, speed can be increased or decreased in increments of 1 (10%) only. To achieve an increase in speed of 30% for example, the user would press the DTMF digit three times.

The default volume modification table is shown in Table 8.

Table 8. Default Volume Modification Table

Table Entry	Default Value (dB)	Absolute Position
decrease[0]	-20	-10
decrease[1]	-18	-9
decrease[2]	-16	-8
decrease[3]	-14	-7
decrease[4]	-12	-6
decrease[5]	-10	-5
decrease[6]	-08	-4
decrease[7]	-06	-3
decrease[8]	-04	-2
decrease[9]	-02	-1
origin	0	0
increase[0]	+02	1
increase[1]	+04	2
increase[2]	+06	3
increase[3]	+08	4
increase[4]	+10	5
increase[5]	-128 (80h)	6
increase[6]	-128 (80h)	7
increase[7]	-128 (80h)	8
increase[8]	-128 (80h)	9
increase[9]	-128 (80h)	10

Consider the following usage information on the volume modification table:

- Each entry in the table is a deviation in decibels from the starting point or volume (“origin”). Each entry in the table is equivalent to 2 dB from the origin. Volume can be decreased 2 dB by specifying position 1 in the table, or by moving one step down. For example, the increase[1] position (two steps from the origin) increases volume by 4 dB.
- The total volume modification range is from -20 dB to +10 dB. In this table, the highest position utilized is the increase[4] position. The remaining increase fields are set to -128 (80h). If these “non-adjustment” positions are selected, the default action is to play at the increase[4] volume. These fields can be reset, as long as no values higher than +10 are used; for example, you could spread the 10 dB volume increase over 10 steps rather than 5.
- In the volume modification table, the default entries for index values +6 to +10 are -128 which represent a null-entry. To customize the table entries, you must use the `dx_setsvmt()` function.
- When adjustment is associated with a DTMF digit, volume can be increased or decreased in increments of 1 (2 dB) only. To achieve an increase in volume of 6 dB for example, the user would press the DTMF digit three times.

9.5 Play Adjustment Digits

The voice software processes play adjustment digits differently from normal digits:

- If a play adjustment digit is entered during playback, it causes a play adjustment only and has no other effect. This means that the digit is not added to the digit queue; it cannot be retrieved with the **dx_getdig()** function.
- Digits that are used for play adjustment may also be used as a terminating condition. If a digit is defined as both, then both actions are applied upon detection of that digit.
- If the digit queue contains adjustment digits when a play begins and play adjustment is set to be level sensitive, the digits will affect the speed or volume and then be removed from the queue.

9.6 Setting Play Adjustment Conditions

Adjustment conditions are set in the same way for speed or volume. The following steps describe how to set conditions upon which volume should be adjusted:

1. Set up the volume modification table (if you do not want to use the defaults):
 - Set up the **DX_SVMT** structure to specify the size and number of the steps in the table.
 - Call the **dx_setsvmt()** function, which points to the **DX_SVMT** structure, to modify the volume modification table (**dx_setsvmt()** can also be used to reset the table to its default values).
2. Set up the **DX_SVCB** structure to specify the condition, the size, and the type of adjustment.
3. Call **dx_setsvcond()**, which points to an array of **DX_SVCB** structures. All subsequent plays will adjust volume as required whenever one of the conditions specified in the array occurs.

See the *Dialogic® Voice API Library Reference* for more information about these functions and data structures.

9.7 Explicitly Adjusting Speed and Volume

Speed and volume adjustments are made in the same way. The following steps describe how to adjust speed, but you can use exactly the same procedure for volume:

1. Set up the speed modification table (if you do not want to use the defaults):
 - Set up the **DX_SVMT** structure to specify the size and number of the steps in the table.
 - Call the **dx_setsvmt()** function, which points to the **DX_SVMT** structure, to modify the speed modification table (**dx_setsvmt()** can also be used to reset the table to its default values).
2. When required, call **dx_adjsv()** to adjust the speed modification table by specifying the size and type of the adjustment.

See the *Dialogic® Voice API Library Reference* for more information about these functions and data structures.

Global Tone Detection and Generation, and Cadenced Tone Generation

This chapter discusses global tone detection (GTD), global tone generation (GTG), and cadenced tone generation:

- Global Tone Detection (GTD)..... 69
- Global Tone Generation (GTG)..... 75
- Cadenced Tone Generation 76

10.1 Global Tone Detection (GTD)

Global tone detection (GTD) is described in the following sections:

- Overview of Global Tone Detection
- Global Tone Detection on Dialogic® Host Media Processing (HMP) Software versus Dialogic® Springware Boards
- Defining Global Tone Detection Tones
- Building Tone Templates
- Working with Tone Templates
- Retrieving Tone Events
- Setting GTD Tones as Termination Conditions
- Guidelines for Creating User-Defined Tones
- Global Tone Detection Application

10.1.1 Overview of Global Tone Detection

Global tone detection (GTD) allows you to define single or dual frequency tones for detection. The characteristics of a tone are defined in a GTD tone template. A tone template contains parameters that allow you to assign frequency bounds and cadence components. GTD can detect single and dual frequency tones by comparing all incoming sounds to the GTD tone templates. Global tone detection and GTD tones are also known as **user-defined tone detection** and **user-defined tones**.

The typical use of global tone detection is to detect single and dual frequency tones other than those automatically provided with the voice software. This includes tones outside the standard DTMF set (0-9, a-d, *, and #), and the standard MF set (0-9, a-c, and *). GTD works simultaneously with DTMF and MF tone detection.

Global Tone Detection and Generation, and Cadenced Tone Generation

When GTD detects a tone, it responds by producing either a **tone event** on the event queue or a **digit** on the digit queue. The particular response depends on the GTD tone configuration.

10.1.2 Global Tone Detection on Dialogic® Host Media Processing (HMP) Software versus Dialogic® Springware Boards

On Dialogic® HMP Software, tone templates are managed internally on a board basis, while on Springware boards, tone templates are managed internally on a channel basis.

On Dialogic® HMP Software, once a tone template is defined, it can be added to any number of channels for global tone detection on a board. The tone template is stored only once on the board. A counter in the tone template tracks the number of channels that are using this template; the template remains active until the very last channel of the board deletes the template.

On Dialogic® HMP Software, the memory available for all tone templates (including call progress analysis tones and user-defined tones) is pre-allocated and fixed. Each tone template takes up the same amount of memory. However, a limitation exists on the number of tone templates that can be active at one time on a channel, due to the number of events that are generated. See [Section 10.1.8, “Guidelines for Creating User-Defined Tones”](#), on page 74 for more information.

On Dialogic® Springware boards, once a tone template is defined, it can be added to any number of channels for global tone detection as well. However, the tone template is stored for each channel that uses it.

10.1.3 Defining Global Tone Detection Tones

GTD tones can have an associated ASCII digit (and digit type) specified using the **digit** and **digtype** parameters in the `dx_addtone()` function. When the tone is detected, the digit is placed in the DV_DIGIT buffer and can be retrieved using the `dx_getdig()` function. When the tone is detected, either the tone event or the digit associated with the tone can be used as a termination condition to terminate I/O functions.

Termination conditions are set using the DV_TPT data structure. To terminate on multiple tones (or digits), you must specify the terminating conditions for each tone in a separate DV_TPT data structure.

For details on voice functions and data structures, see the *Dialogic® Voice API Library Reference*.

10.1.4 Building Tone Templates

When creating a tone template, you can define the following:

- single frequency or dual frequency (300 - 3500 Hz)
- optional ASCII digit associated with the tone template
- cadence components

Adding a tone template to a channel enables detection of a tone on that channel. Although only one tone template can be created at a time, multiple tone templates can be added to a channel. Each channel can have a different set of tone templates. Once created, tone templates can be selectively enabled or disabled.

API Library Functions

The following functions are used to build and define tone templates:

dx_bldst()

Defines a single frequency tone. Subsequent calls to **dx_addtone()** will use this tone until another tone is defined.

dx_blddt()

Defines a simple dual frequency tone. Subsequent calls to **dx_addtone()** will use this tone until another tone is defined.

Note that the boards cannot detect dual tones with frequency components closer than approximately 63 Hz. Use a single tone description to detect dual tones that are closer together than the ranges specified above.

dx_bldstcad()

Defines a simple single frequency cadence tone. Subsequent calls to **dx_addtone()** will use this tone until another tone is defined. A single frequency cadence tone has single frequency signals with specific on/off characteristics.

dx_blddtcad()

Defines a simple dual frequency cadence tone. Subsequent calls to **dx_addtone()** will use this tone until another tone is defined. A dual frequency cadence tone has dual frequency signals with specific on/off characteristics. The minimum on- and off-time for cadence detection is 40 msec on and 40 msec off.

dx_setgtdamp()

Sets the amplitudes used by GTD. The amplitudes set using **dx_setgtdamp()** are the default amplitudes that apply to all tones built using the build-tone functions. The amplitudes remain valid for all tones built until **dx_setgtdamp()** is called again and amplitudes are changed.

Instructions for Building Tone Templates

Follow these steps to build a tone template for global tone detection:

1. Determine the characteristics of the tone you wish to detect: single frequency or dual frequency tone, frequency range, cadence components, and so on.
2. Use the appropriate build-tone function, such as **dx_bldst()**, **dx_blddt()** and so on, to build and define the tone template. The functions require that you specify a unique tone ID for each tone template.
3. Use the **dx_addtone()** function to add the tone template to a channel. Subsequent calls to **dx_addtone()** will add this tone template until another tone is defined. Adding a tone template to a channel enables detection of a tone on that channel.
4. Repeat steps 1 - 3 as needed.

Tips and Hints for Building Tone Templates

The following are tips and hints when building a tone template for global tone detection:

- After using a build-tone function to define a new tone template, you must call **dx_addtone()** to add the tone template to a channel and enable detection of that tone on a channel.
- After using a build-tone function to define a tone template, if the template is not added to a channel, the next call to a build-tone function will overwrite the tone definition contained in the previous template.
- Only one tone template can be created at a time; however, multiple tone templates can be added to a channel. Each channel can have a different set of tone templates. Once created, tone templates can be selectively disabled or enabled by using **dx_distone()** and **dx_enbtone()**. See [Section 10.1.5, “Working with Tone Templates”](#), on page 72 for more information.
- Each tone template must have a unique tone ID.
- On Windows®, do not use tone IDs 261, 262 and 263; they are reserved for library use.
- A particular tone template that has been added to a channel cannot be changed or deleted. A tone template can be disabled on a channel, but to delete a tone template, all tone templates on that channel must be deleted. See [Section 10.1.5, “Working with Tone Templates”](#), on page 72 for more information.

Table 9 lists some standard Bell System Network call progress tones. The frequencies are useful to know when creating tone templates.

Table 9. Standard Bell System Network Call Progress Tones

Tone	Frequency (Hz)	On Time (msec)	Off Time (msec)
Dial	350 + 440	Continuous	
Busy	480 + 620	500	500
Congestion (Toll)	480 + 620	200	300
Reorder (Local)	480 + 620	300	200
Ringback	440 + 480	2000	4000

10.1.5 Working with Tone Templates

After building a tone template, use the following functions to add/delete tone templates or to enable/disable tone detection:

dx_addtone()

Adds a tone template that was defined by the most recent build-tone function call to the specified channel. Adding a tone template to a channel downloads it to the board and enables detection of tone-on and tone-off events for that tone template.

dx_deltone()

Removes all tone templates previously added to a channel with **dx_addtone()**. If no tone templates were previously enabled for this channel, the function has no effect.

dx_deltone() does not affect tones defined by build-tone template functions and tone templates not yet defined. If you have added tones for call progress analysis, these tones are also deleted.

dx_distone()

Disables detection of a user-defined tone on a channel as well as the DE_TONEON and/or DE_TONEOFF events for that tone. Detection capability for user-defined tones is enabled on a channel by default when **dx_addtone()** is called.

dx_enbtone()

Enables detection of a user-defined tone that was previously disabled by **dx_distone()**. Also enables detection of DE_TONEON and/or DE_TONEOFF events for that tone. Detection capability for user-defined tones is enabled on a channel by default when **dx_addtone()** is called.

10.1.6 Retrieving Tone Events

Tone-on events (DE_TONEON) and tone-off events (DE_TONEOFF) are call status transition (CST) events. Retrieval of these events is handled differently for asynchronous and synchronous applications. Table 10 outlines the different processes for retrieving tone events.

Table 10. Asynchronous/Synchronous Tone Event Handling

Synchronous	Asynchronous
Call dx_addtone() or dx_enbtone() to enable tone-on/off detection.	Call dx_addtone() or dx_enbtone() to enable tone-on/off detection.
Call dx_getevt() to wait for CST event(s). Events are returned in the DX_EBLK data structure.	Use Dialogic® Standard Runtime Library (SRL) to asynchronously wait for TDX_CST event(s).
N/A	Use sr_getevtdatap() to retrieve DX_CST data structure.
Note: These procedures are the same as the retrieval of any other CST event, except that dx_addtone() or dx_enbtone() are used to enable event detection instead of dx_setevtmask() .	

You can optionally specify an associated ASCII digit (and digit type) with the tone template. In this case, the tone template is treated like DTMF tones. When the digit is detected, it is placed in the digit buffer and can be used for termination. When an associated ASCII digit is specified, tone events will not be generated for that tone.

Note: Cadence tone on events are reported differently on Dialogic® HMP software versus Springware boards. On Dialogic® HMP software, if a cadence tone occurs continuously, a DE_TONEON event is reported for each on/off cycle. On Springware boards, a DE_TONEON event is reported for the first on/off cycle only. On Dialogic® HMP software and on Dialogic® Springware boards, a DE_TONEOFF event is reported when the tone is no longer present.

10.1.7 Setting GTD Tones as Termination Conditions

To detect a GTD (user-defined) tone, you can specify it as a termination condition for I/O functions. Set the `tp_termno` field in the `DV_TPT` structure to `DX_TONE`, and specify `DX_TONEON` or `DX_TONEOFF` in the `tp_data` field.

10.1.8 Guidelines for Creating User-Defined Tones

The following guidelines apply when creating user-defined tones:

Note: The terms “user-defined tones” and “tone templates” are used interchangeably. Each tone template specifies the characteristics of one user-defined tone.

- The maximum number of user-defined tone templates is based on tone templates that define a *dual* tone with a frequency range (bandwidth) of 63 Hz. (The detection range is the difference between the minimum and maximum defined frequencies for the tone.)
- The maximum number of user-defined tone templates is 40.

Note: In the case where the number of of tone templates is exhausted, no error is returned on `dx_addtone()` and subsequent tone detection may fail.

- The number of user-defined tone templates that are active and enabled on a channel is limited due to the number of events that a channel can report. The default maximum number of events for a channel is 20. (A tone-on event and a tone-off event in the same tone template count as two events.)
- The default call progress analysis tones (which include tri-tone special information tone sequences or SIT sequences) are created at board initialization time and are available for use. If you create new call progress analysis tone templates using `dx_querytone()`, `dx_deletetone()` and `dx_createtone()`, each tone template counts as a user-defined tone template, which reduces the number of user-defined tones you can create.
- Building and adding tones of zero frequency values to a tone template can cause firmware failures.
- The `dx_deltone()` function deletes all user-defined tones from a specified channel and releases the memory that was used for those user-defined tones. When an associated ASCII digit is specified, tone events will not be generated for that tone.

10.1.9 Global Tone Detection Application

A sample application for global tone detection (GTD) is detecting leading edge debounce time.

Rather than detecting a signal immediately, an application may want to wait for a period of time (debounce time) before the `DE_TONEON` event is generated indicating the detection of the signal. The `dx_bldstcad()` and `dx_blddtcad()` functions can detect leading edge debounce on-time. A tone must be present at a given frequency and for a period of time (debounce time) before a `DE_TONEON` event is generated. The debounce time is specified using the tone-on time, tone-on time deviation, tone-off time, tone-off time deviation, and repetition count parameters in the `dx_bldstcad()` or `dx_blddtcad()` functions.

To detect leading edge debounce time, specify the following values for the **dx_bldstcad()** or **dx_blddtcad()** function parameters listed below:

- For **ontime**, specify 1/2 of the desired debounce time
- For **ontdev**, specify -1/2 of the desired debounce time
- For **offtime**, specify 0
- For **offtdev**, specify 0
- For **repcnt**, specify 0

Note: The **dx_blddt()** and **dx_bldst()** functions cannot be used to detect leading edge debounce time because they do not have timing field parameters.

10.2 Global Tone Generation (GTG)

The following topics provide information on using global tone generation:

- [Using GTG](#)
- [GTG Functions](#)
- [Building and Implementing a Tone Generation Template](#)

10.2.1 Using GTG

Global tone generation enables the creation of user-defined tones. The tone generation template, TN_GEN, is used to define the tones with the following information:

- Single or dual tone
- Frequency fields
- Amplitude for each frequency
- Duration of tone

The functions and data structures associated with global tone generation are described in the *Dialogic® Voice API Library Reference*.

10.2.2 GTG Functions

The following functions are used to generate tones:

dx_bldtngen()

Builds a tone generation template. This convenience function sets up the tone generation template data structure (TN_GEN) by allowing the assignment of specified values to the appropriate fields. The tone generation template is placed in the user's return buffer and can then be used by the **dx_playtone()** function to generate the tone.

dx_playtone()

Plays a tone specified by the tone generation template (pointed to by **tngenp**). Termination conditions are set using the DV_TPT structure. The reason for termination is returned by the

Global Tone Detection and Generation, and Cadenced Tone Generation

ATDX_TERMMSK() function. **dx_playtone()** returns a 0 to indicate that it has completed successfully.

10.2.3 Building and Implementing a Tone Generation Template

The tone generation template defines the frequency, amplitude, and duration of a single or dual frequency tone to be played. You can use the convenience function **dx_bldtngen()** to set up the structure. Use **dx_playtone()** to play the tone.

The TN_GEN data structure is defined as:

```
typedef struct {
    unsigned short tg_dflag;          /* dual tone = 1, single tone = 0 */
    unsigned short tg_freq1;         /* frequency of tone 1 (in Hz) */
    unsigned short tg_freq2;         /* frequency of tone 2 (in Hz) */
    short int      tg_ampl1;         /* amplitude of tone 1 (in dB) */
    short int      tg_ampl2;         /* amplitude of tone 2 (in dB) */
    short int      tg_dur;           /* duration (in 10 msec units) */
} TN_GEN;
```

After you build the TN_GEN data structure, there are two ways to define each tone template. You may either:

- Include the values in the structure
- Pass the values to TN_GEN using the **dx_bldtngen()** function

If you include the values in the structure, you must create a structure for each tone template. If you pass the values using the **dx_playtone()** function, then you can reuse the structure. If you are only changing one value in a template with many variables, it may be more convenient to use several structures in the code instead of reusing just one.

After defining the template by either of these methods, pass TN_GEN to **dx_playtone()** to play the tone.

10.3 Cadenced Tone Generation

The following topics provide information on enabling and using cadenced tone generation:

- [Using Cadenced Tone Generation](#)
- [How To Generate a Custom Cadenced Tone](#)
- [How To Generate a Non-Cadenced Tone](#)
- [TN_GENCAD Data Structure - Cadenced Tone Generation](#)
- [How To Generate a Standard PBX Call Progress Signal](#)
- [Predefined Set of Standard PBX Call Progress Signals](#)
- [Important Considerations for Using Predefined Call Progress Signals](#)

10.3.1 Using Cadenced Tone Generation

Cadenced tone generation is an enhancement to global tone generation that enables you to generate a signal with up to four single or dual tone elements, each with its own on/off duration creating the signal pattern or cadence.

Cadenced tone generation is accomplished with the **dx_playtoneEx()** function and the TN_GENCAD data structure.

You can define your own custom cadenced tone or take advantage of the built-in set of standard PBX call progress signals.

The functions and data structures associated with cadenced tone generation are described in the *Dialogic® Voice API Library Reference*.

10.3.2 How To Generate a Custom Cadenced Tone

A custom cadenced tone is defined by specifying in a TN_GENCAD data structure the repeating elements of the signal (the cycle) and the number of desired repetitions.

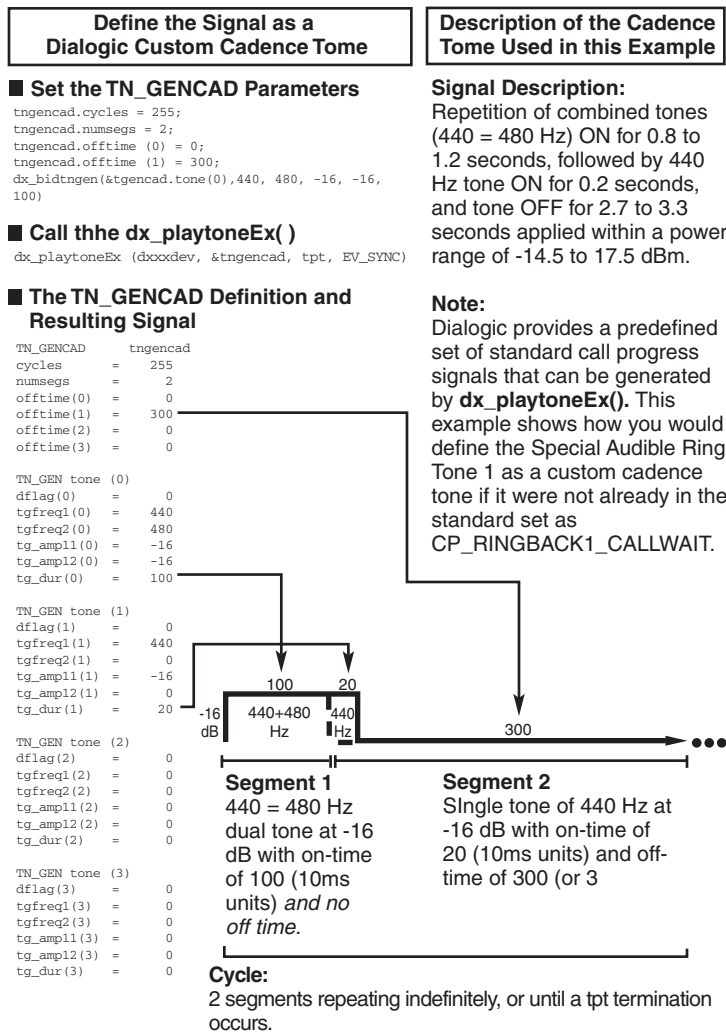
The cycle can consist of up to 4 segments, each with its own tone definition and cadence. A segment consists of a TN_GEN single or dual tone definition (frequency, amplitude, & duration) followed by a corresponding off-time (silence duration) that is optional. The **dx_bldtngen()** function can be used to set up the TN_GEN components of the TN_GENCAD structure. The tone duration, or on-time, from TN_GEN (tg_dur) and the offtime from TN_GENCAD are combined to produce the cadence for the segment. The segments are seamlessly concatenated in ascending order to generate the signal cycle.

Use the following procedure to generate a custom cadenced tone:

1. Identify the repeating elements of the signal (the cycle).
2. Use a TN_GENCAD structure to define the segments in the cycle:
 - a. Start with the first tone element in the cycle and identify the single or dual tone frequencies, amplitudes, and duration (on-time).
 - b. Use the **dx_bldtngen()** function to specify this tone definition in tone[0] (the first TN_GEN tone array element) of the TN_GENCAD structure.
 - c. Identify the off-time for the first tone element and specify it in offtime[0]. If the first tone element is followed immediately by a second tone element, set offtime[0] = 0.
 - d. Define the next segment of the cycle in tone[1] and offtime[1] the same way as above, and so on, up to the maximum of 4 segments or until you reach the end of the cycle.
3. Use the TN_GENCAD to define the parameters of the cycle:
 - a. Specify the number of segments in the cycle (numsegs).
 - b. Specify the number of cycle repetitions (cycles).
4. Set the termination conditions in a DV_TPT structure.
5. Call the **dx_playtoneEx()** function and use the **tngetcadv** parameter to specify the custom cadenced tone that you defined in the TN_GENCAD.

For an illustration of this procedure, see Figure 3.

Figure 3. Example of Custom Cadenced Tone Generation



10.3.3 How To Generate a Non-Cadenced Tone

Both `dx_playtoneEx()` and `dx_playtone()` can generate a non-cadenced tone.

Non-cadenced call progress signals can be generated by the `dx_playtone()` function if you define them in a TN_GEN: Dial Tone, Executive Override Tone, and Busy Verification Tone Part A.

The `dx_playtoneEx()` function can also generate a non-cadenced tone by using a TN_GENCAD data structure that defines a single segment.

If you want to generate a continuous, non-cadenced signal, use a single segment and zero off-time, and specify 1) an infinite number of cycles, 2) an infinite on-time, or 3) both. (You must also specify the appropriate termination conditions in a DV_TPT structure or else the tone will never end). For example:

```
cycles = 255;
numsegs = 1;
offtime[0] = 0;
tone[0].tg_dur = -1
```

10.3.4 TN_GENCAD Data Structure - Cadenced Tone Generation

TN_GENCAD is a voice library data structure (*dxxxlib.h*) that defines a cadenced tone that can be generated by using the `dx_playtoneEx()` function.

The TN_GENCAD data structure contains a tone array with four elements that are TN_GEN data structures (Tone Generation Templates). For details on TN_GEN and TN_GENCAD, see the *Voice API Library Reference*.

For examples of TN_GENCAD, see the definitions of standard call progress signals in [Table 12, “TN_GENCAD Definitions for Standard PBX Call Progress Signals”](#), on page 84.

10.3.5 How To Generate a Standard PBX Call Progress Signal

Use the following procedure to generate a standard PBX call progress signal from the predefined set of standard PBX call progress signals:

1. Select a call progress signal from [Table 11, “Standard PBX Call Progress Signals”](#), on page 81 and note the signal ID (see also [Figure 4, “Standard PBX Call Progress Signals \(Part 1\)”](#), on page 82).
2. Set the termination conditions in a DV_TPT structure.
3. Call the `dx_playtoneEx()` function and specify the signal ID for the `tngetcadv` parameter.

For example:

```
dx_playtoneEx(dxxxdev, CP_BUSY, tpt, EV_SYNC)
```

10.3.6 Predefined Set of Standard PBX Call Progress Signals

The following information describes the predefined set of standard PBX call progress signals that are provided by Dialogic:

- [Table 11, “Standard PBX Call Progress Signals”](#), on page 81 lists the predefined, standard, call progress signals and their signal IDs. The signal IDs can be used with the `dx_playtoneEx()` function to generate the signal. (The #defines for the signal IDs are located in the `dxxlib.h` file.)
- [Figure 4, “Standard PBX Call Progress Signals \(Part 1\)”](#), on page 82 illustrates the signals along with their tone specifications and cadences. The signals were divided into two parts so they could be illustrated to scale while providing sufficient detail. Part 1 uses a smaller scale than Part 2. (For this reason, the order of the signals is different than in the tables.)
- [Table 12, “TN_GENCAD Definitions for Standard PBX Call Progress Signals”](#), on page 84 lists the TN_GENCAD definitions of the signal cycle and segment definitions for each predefined call progress signal. These definitions are located in the `dxgtd.c` file.
- [Section 10.3.7, “Important Considerations for Using Predefined Call Progress Signals”](#), on page 85 describes what standard was used, how the standard was implemented, information regarding the signal power levels, usage and other considerations.

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Table 11. Standard PBX Call Progress Signals

Name	Meaning	Signal ID (tngencadp)
Dial Tone	Ready for dialing	CP_DIAL
Reorder Tone (Paths-Busy, All-Trunks-Busy, Fast Busy)	Call blocked: resources unavailable	CP_REORDER
Busy Tone (Slow Busy)	Called line is busy	CP_BUSY
Audible Ring Tone 1 (Ringback Tone)	Called party is being alerted	CP_RINGBACK1
Audible Ring Tone 2 ¹ (Slow Ringback Tone)	Called party is being alerted	CP_RINGBACK2
Special Audible Ring Tone 1 ¹	Called party has Call Waiting feature and is being alerted	CP_RINGBACK1_CALLWAIT
Special Audible Ring Tone 2 ¹	Called party has Call Waiting feature and is being alerted	CP_RINGBACK2_CALLWAIT
Recall Dial Tone	Ready for additional dialing on established connection	CP_RECALL_DIAL
Intercept Tone	Call blocked: invalid	CP_INTERCEPT
Call Waiting Tone 1 ²	Call is waiting: single burst	C_CALLWAIT1
Call Waiting Tone 2 ²	Call is waiting: double burst	CP_CALLWAIT2
Busy Verification Tone (Part A)	Alerts parties that attendant is about to enter connection	CP_BUSY_VERIFY_A
Busy Verification Tone (Part B)	Attendant remains connected	CP_BUSY_VERIFY_B
Executive Override Tone	Overriding party about to be bridged onto connection	CP_EXEC_OVERRIDE
Confirmation Tone	Feature has been activated or deactivated	CP_FEATURE_CONFIRM
Stutter Dial Tone (Message Waiting Dial Tone)	Message waiting; ready for dialing	CP_STUTTER_DIAL or CP_MSG_WAIT_DIAL
¹ Either of the two Audible Ring Tones can be used but are not intended to be intermixed in a system. When using the <i>Special</i> Audible Ring Tone (1 or 2), it should correspond to the Audible Ring Tone (1 or 2) that is used. ² The two Call Waiting Tones (1 & 2) can be used to differentiate between internally and externally originated calls. Call Waiting Tone 2 is defined as a double burst in this implementation, although "multiple" bursts are permissible.		

Figure 4. Standard PBX Call Progress Signals (Part 1)

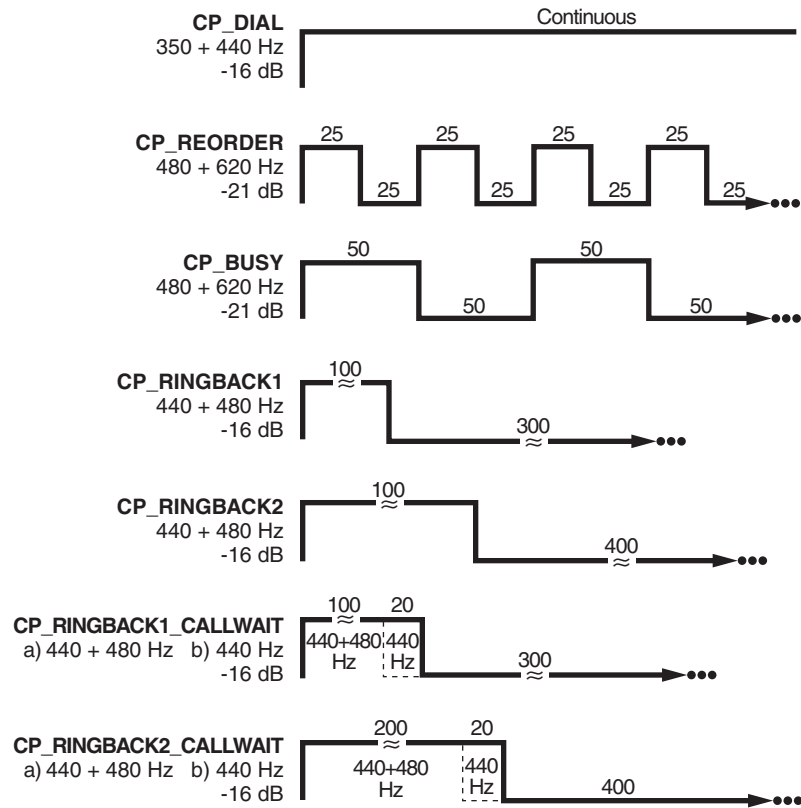
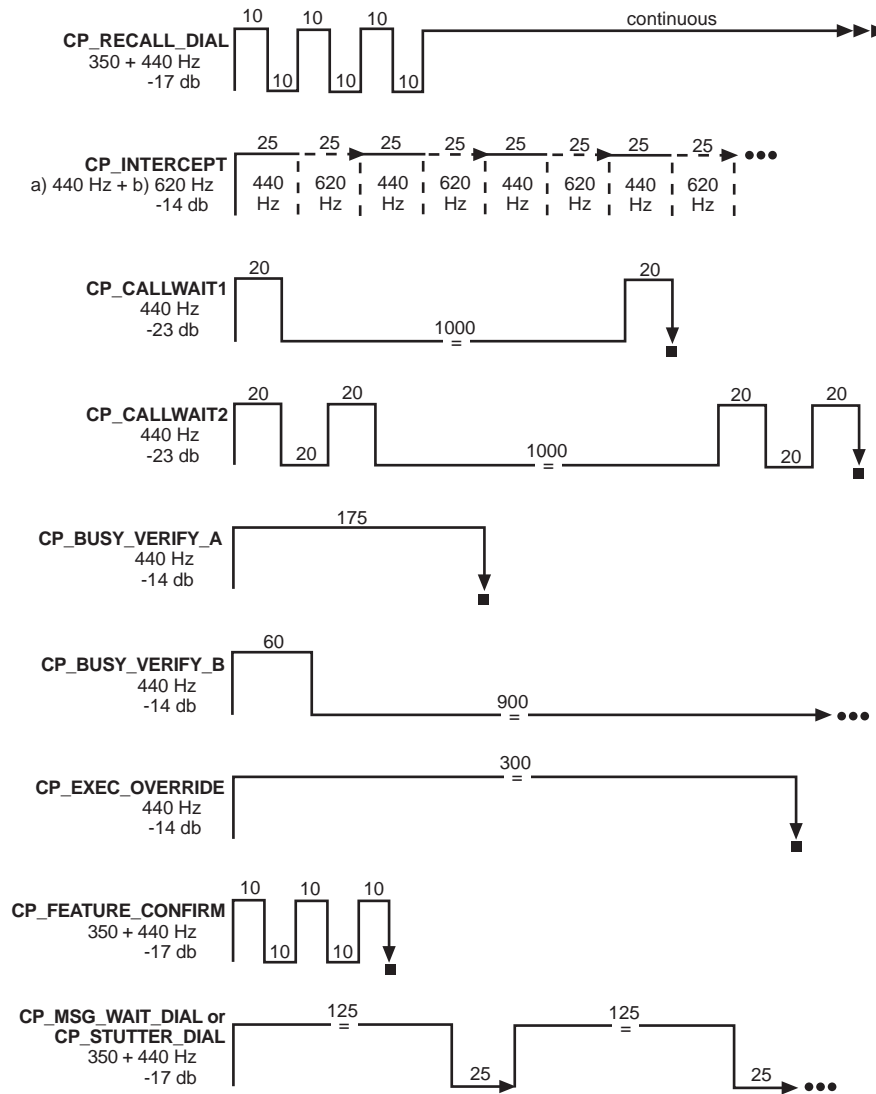


Figure 5. Standard PBX Call Progress Signals (Part 2)



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Table 12. TN_GENCAD Definitions for Standard PBX Call Progress Signals

SIGNAL_ID		Segment Definitions						On-Time ² (10 msec)	Off-Time (10 msec)
Cycle Definition		Frequency #1 (Hz)	Frequency #2 (Hz)	Amplitude #1 (dB)	Amplitude #2 (dB)	On-Time ² (10 msec)	Off-Time (10 msec)		
Number of Cycles ¹	Number of Segments in Cycle	Frequency #1 (Hz)	Frequency #2 (Hz)	Amplitude #1 (dB)	Amplitude #2 (dB)	On-Time ² (10 msec)	Off-Time (10 msec)		
cycles	numsegs	tg_freq1	tg_freq2	tg_ampl1	tg_ampl2	tg_dur	offtime		
CP_DIAL									
1	1	350	440	-17	-17	-1	0		
CP_REORDER									
255	1	480	620	-21	-21	25	25		
CP_BUSY									
255	1	480	620	-21	-21	50	50		
CP_RINGBACK1									
255	1	440	480	-16	-16	100	300		
CP_RINGBACK2									
255	1	440	480	-16	-16	200	400		
CP_RINGBACK1_CALLWAIT									
255	2	440 440	480 0	-16 -16	-16 0	100 20	0 300		
CP_RINGBACK2_CALLWAIT									
255	2	440 440	480 0	-16 -16	-16 0	200 20	0 400		
CP_RECALL_DIAL									
1	4	350 350 350 350	440 440 440 440	-17 -17 -17 -17	-17 -17 -17 -17	10 10 10 -1	10 10 10 0		
CP_INTERCEPT									
255	2	440 620	0 0	-14 -14	0 0	25 25	0 0		
CP_CALLWAIT1									
1	2	440 440	0 0	-23 -23	0 0	20 20	1000 0		
CP_CALLWAIT2									
1	4	440 440 440 440	0 0 0 0	-23 -23 -23 -23	0 0 0 0	20 20 20 20	20 1000 20 0		
¹ 255 specifies an infinite number of cycles (cycles) ² -1 specifies an infinite tone duration (tg_dur)									

Table 12. TN_GENCAD Definitions for Standard PBX Call Progress Signals (Continued)

SIGNAL_ID							
Cycle Definition		Segment Definitions					
Number of Cycles ¹	Number of Segments in Cycle	Frequency #1 (Hz)	Frequency #2 (Hz)	Amplitude #1 (dB)	Amplitude #2 (dB)	On-Time ² (10 msec)	Off-Time (10 msec)
cycles	numsegs	tg_freq1	tg_freq2	tg_ampl1	tg_ampl2	tg_dur	offtime
CP_BUSY_VERIFY_A							
1	1	440	0	-14	0	175	0
CP_BUSY_VERIFY_B							
255	1	440	0	-14	0	60	900
CP_EXEC_OVERRIDE							
1	1	440	0	-14	0	300	0
CP_FEATURE_CONFIRM							
1	3	350	440	-17	-17	10	10
		350	440	-17	-17	10	10
		350	440	-17	-17	10	0
CP_STUTTER_DIAL or CP_MSG_WAIT_DIAL							
255	1	350	440	-17	-17	125	25
¹ 255 specifies an infinite number of cycles (cycles) ² -1 specifies an infinite tone duration (tg_dur)							

10.3.7 Important Considerations for Using Predefined Call Progress Signals

Take into account the following considerations when using the predefined call progress signals:

- Signal definitions are based on the TIA/EIA Standard: Requirements for Private Branch Exchange (PBX) Switching Equipment, TIA/EIA-464-B, April 1996 (Telecommunications Industry Association in association with the Electronic Industries Association, Standards and Technology Department, 2500 Wilson Blvd., Arlington, VA 22201). To order copies, contact Global Engineering Documents in the USA at 1-800-854-7179 or 1-303-397-7956.
- A separate Line Lockout Warning Tone, which indicates that the station line has been locked out because dialing took too long or the station failed to disconnect at the end of a call, is not necessary and is not recommended. You can use the Reorder tone over trunks; or the Intercept, Reorder, or Busy tone over stations.
- For signals that specify an infinite repetition of the signal cycle (cycles = 255 on Dialogic® Springware board or 40 on Dialogic® DM3 board) or an infinite duration of a tone (tg_dur = -1), you must specify the appropriate termination conditions in the DV_TPT structure used by **dx_playtoneEx()**.
- There may be more than one way to use TN_GENCAD to generate a given signal. For example, the three bursts of the Confirmation Tone can be created through one cycle

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containing three segments (as in the Dialogic implementation) or through a single segment that is repeated in three cycles.

- To generate a continuous, non-cadenced signal, you can use **dx_playtoneEx()** and **TN_GENCAD** to specify a single segment with zero off-time and with an infinite number of cycles and/or an infinite on-time.
Alternatively, you could use **dx_playtone()** and **TN_GEN** to generate a non-cadenced signal. The following non-cadenced call progress signals could be generated by the **dx_playtone()** function if you defined them in a **TN_GEN**: 1) Dial Tone, 2) Executive Override Tone, and 3) Busy Verification Tone Part A.
- Note that the Intercept Tone consists of alternating single tones.
- Although the TIA/EIA Standard describes the Busy Verification Tone as one signal, the two segments are separate tones/events: Part A is a single burst almost three times longer than Part B and it alerts the parties before the attendant intrudes; Part B is a short burst every 9 seconds continuing as long as the interruption lasts. The TIA/EIA Standard does not define an off-time between Part A and B. Therefore, the application developer is responsible for implementing the timing between the two parts of this signal.
- The TIA/EIA Standard specifies the range of permissible power levels per frequency for 1) the Central Office trunk interface and 2) all other interfaces (including off-premise stations and tie trunks). The Dialogic implementation adopted the approximate middle value in the acceptable range of power levels for applying the signals to the CO trunk interface. These power levels were more restrictive than those for the other interfaces. According to the following statement in the TIA/EIA Standard, additional requirements and considerations may apply:
“Studies have shown that the lower level tones that are transmitted over trunks should be 6 dB hotter at the trunk interface (than at the line interface) to compensate for increased loss on the end-to-end connection. In the case of tones used at higher levels, the 6 dB difference is not used since power at trunk interfaces must be limited to -13 dBm0 total when averaged over any 3-second interval to prevent carrier overload. Maximum permissible powers listed are consistent with this requirement taking into account the allowable interruption rates for the various tones. Uninterrupted tones, such as Dial Tone and Intercept Tone, shall be continuously limited to -13 dBm.”
For related power level information, see also Note 1 for Tables 29 and 30, Section 5.9, and Section 6.3.5.

This chapter provides information on building applications using the Dialogic® Voice API library. The following topics are discussed:

- Dialogic® Voice and SRL API Libraries 87
- Compiling and Linking 88

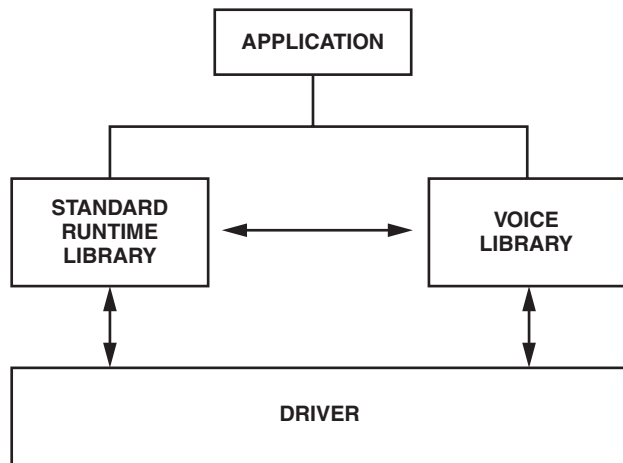
11.1 Dialogic® Voice and SRL API Libraries

The C-language application programming interface (API) included with the voice software provides a library of functions used to create voice processing applications.

The Dialogic® Voice API library and Dialogic® Standard Runtime Library (SRL) files are part of the voice software. These libraries provide the interface to the voice driver. For detailed information on the SRL, see the *Dialogic® Standard Runtime Library API Programming Guide* and *Dialogic® Standard Runtime Library API Programming Guide*.

Figure 6 illustrates how the Dialogic® Voice and SRL API libraries interface with the driver.

Figure 6. Dialogic® Voice and SRL API Libraries



11.2 Compiling and Linking

The following topics discuss compiling and linking requirements:

- [Include Files](#)
- [Required Libraries for Linux](#)
- [Required Libraries for Windows®](#)
- [Variables for Compiling and Linking](#)

11.2.1 Include Files

Function prototypes and equates are defined in include files, also known as header files. Applications that use the Dialogic® Voice API library functions must contain statements for include files in this form, where filename represents the include file name:

```
#include <filename.h>
```

The following header files must be included in application code **in the order shown** prior to calling Dialogic® Voice API library functions:

srlib.h

Contains function prototypes and equates for the Dialogic® Standard Runtime Library (SRL). Used for all application development.

dxxlib.h

Contains function prototypes and equates for the Dialogic® Voice API library. Used for voice processing applications.

Note: *srlib.h* must be included in code before all other Dialogic header files.

11.2.2 Required Libraries for Linux

Simple C language interfaces in source-code format are provided to each individual technology DLL (such as standard runtime, voice, fax, and network interfaces). These C language interfaces allow an application to perform run-time linking instead of compile-time linking.

Note: Compile-time linking requires that all functions called in an application be contained in the DLL that resides on the system.

By default, the library files are located in the directory given by the INTEL_DIALOGIC_LIB environment variable.

You must link the following shared object library files **in the order shown** when compiling your voice processing application:

libdxxx.so

Main voice library file. Specify **-ldxxx** in makefile.

libsrl.so

Standard Runtime Library file. Specify **-lsrl** in makefile.

If you use **courses**, you must ensure that it is the last library to be linked.

Note: When compiling an application, you must list Dialogic® libraries before all other libraries such as operating system libraries.

11.2.3 Required Libraries for Windows®

Simple C language interfaces in source-code format are provided to each individual technology DLL (such as standard runtime, voice, fax, and network interfaces). These C language interfaces allow an application to perform run-time linking instead of compile-time linking.

Note: Compile-time linking requires that all functions called in an application be contained in the DLL that resides on the system.

By default, the library files are located in the directory given by the INTEL_DIALOGIC_LIB environment variable.

You must link the following library files **in the order shown** when compiling your voice processing application:

libdxxmt.lib

Main voice library file.

libsrlmt.lib

Standard Runtime Library file.

11.2.4 Variables for Compiling and Linking

The following variables provide a standardized way of referencing the directories that contain header files and shared objects:

INTEL_DIALOGIC_DIR

Variable that points to the directory in which the release software is installed.

INTEL_DIALOGIC_INC

Variable that points to the directory in which header files are stored.

INTEL_DIALOGIC_LIB

Variable that points to the directory in which shared library files are stored.

These variables are automatically set at login and should be used in compiling and linking commands.

Note: It is strongly recommended that you use these variables when compiling and linking applications. The name of the variables will remain constant, but the values may change in future releases.

Building Applications

Glossary

A-law: Pulse Code Modulation (PCM) algorithm used in digitizing telephone audio signals in E1 areas. Contrast with [mu-law](#).

ADPCM (Adaptive Differential Pulse Code Modulation): A sophisticated compression algorithm for digitizing audio that stores the differences between successive samples rather than the absolute value of each sample. This method of digitization reduces storage requirements from 64 kilobits/second to as low as 24 kilobits/second.

AGC (Automatic Gain Control): An electronic circuit used to maintain the audio signal volume at a constant level. AGC maintains nearly constant gain during voice signals, thereby avoiding distortion, and optimizes the perceptual quality of voice signals by using a new method to process silence intervals (background noise).

analog: 1. A method of telephony transmission in which the signals from the source (for example, speech in a human conversation) are converted into an electrical signal that varies continuously over a range of amplitude values analogous to the original signals. 2. Not digital signaling. 3. Used to refer to applications that use loop start signaling.

ANI (Automatic Number Identification): Identifies the phone number that is calling. Digits may arrive in analog or digital form.

API (Application Programming Interface): A set of standard software interrupts, calls, and data formats that application programs use to initiate contact with network services, mainframe communications programs, or other program-to-program communications.

ASCIIZ string: A null-terminated string of ASCII characters.

asynchronous function: A function that allows program execution to continue without waiting for a task to complete. To implement an asynchronous function, an application-defined event handler must be enabled to trap and process the completed event. Contrast with [synchronous function](#).

bit mask: A pattern which selects or ignores specific bits in a bit-mapped control or status field.

bitmap: An entity of data (byte or word) in which individual bits contain independent control or status information.

board device: On Dialogic® Host Media Processing (HMP) Software, a board-level object that can be manipulated by a physical library. Dialogic® HMP Software performs like a virtual Dialogic® DM3 board.

buffer: A block of memory or temporary storage device that holds data until it can be processed. It is used to compensate for the difference in the rate of the flow of information (or time occurrence of events) when transmitting data from one device to another.

bus: An electronic path that allows communication between multiple points or devices in a system.

busy device: A device that has one of the following characteristics: is stopped, being configured, has a multitasking or non-multitasking function active on it, or I/O function active on it.

cadence: A pattern of tones and silence intervals generated by a given audio signal. The pattern can be classified as a single ring, a double ring, or a busy signal.

cadence detection: A voice driver feature that analyzes the audio signal on the line to detect a repeating pattern of sound and silence.

call progress analysis: A process used to automatically determine what happens after an outgoing call is dialed. A further distinction is made. Call progress refers to activity that occurs before a call is connected (pre-connect), such as busy or ringback. Call analysis refers to activity that occurs after a call is connected (post-connect), such as voice detection and answering machine detection. The term call progress analysis is used to encompass both call progress and call analysis.

call status transition event functions: A class of functions that set and monitor events on devices.

caller ID: calling party identification information.

CCITT (Comite Consultatif Internationale de Telegraphique et Telephonique): One of the four permanent parts of the International Telecommunications Union, a United Nations agency based in Geneva. The CCITT is divided into three sections: 1. Study Groups set up standards for telecommunications equipment, systems, networks, and services. 2. Plan Committees develop general plans for the evolution of networks and services. 3. Specialized Autonomous Groups produce handbooks, strategies, and case studies to support developing countries.

channel: 1. When used in reference to a Dialogic® analog expansion board, an audio path, or the activity happening on that audio path (for example, when you say the channel goes off-hook). 2. When used in reference to an Dialogic® digital expansion board, a data path, or the activity happening on that data path. 3. When used in reference to a bus, an electrical circuit carrying control information and data.

channel device: A channel-level object that can be manipulated by a physical library, such as an individual telephone line connection. A channel is also a subdevice of a board. See also [subdevice](#).

CO (Central Office): A local phone network exchange, the telephone company facility where subscriber lines are linked, through switches, to other subscriber lines (including local and long distance lines). The term “Central Office” is used in North America. The rest of the world calls it “PTT”, for Post, Telephone, and Telegraph.

computer telephony (CT): The extension of computer-based intelligence and processing over the telephone network to a telephone. Sometimes called computer-telephony integration (CTI), it lets you interact with computer databases or applications from a telephone, and enables computer-based applications to access the telephone network. Computer telephony technology supports applications such as: automatic call processing; automatic speech recognition; text-to-speech conversion for information-on-demand; call switching and conferencing; unified messaging, which lets you access or transmit voice, fax, and e-mail messages from a single point; voice mail and voice messaging; fax systems, including fax broadcasting, fax mailboxes, fax-on-demand, and fax gateways; transaction processing, such as Audiotex and Pay-Per-Call information systems; and call centers handling a large number of agents or telephone operators for processing requests for products, services, or information.

configuration file: An unformatted ASCII file that stores device initialization information for an application.

convenience function: A class of functions that simplify application writing, sometimes by calling other, lower-level API functions.

CPE: customer premise equipment.

CT Bus: Computer Telephony bus. A time division multiplexing communications bus that provides 4096 time slots for transmission of digital information between CT Bus products. See [TDM bus](#).

data structure: Programming term for a data element consisting of fields, where each field may have a different type definition and length. A group of data structure elements usually share a common purpose or functionality.

DCM: configuration manager. On Windows® only, a utility with a graphical user interface (GUI) that enables you to add new boards to your system, start and stop system service, and work with board configuration data.

debouncing: Eliminating false signal detection by filtering out rapid signal changes. Any detected signal change must last for the minimum duration as specified by the debounce parameters before the signal is considered valid. Also known as deglitching.

deglitching: See [debouncing](#).

device: A computer peripheral or component controlled through a software device driver. A Dialogic® voice and/or network interface expansion board is considered a physical board containing one or more logical board devices, and each channel or time slot on the board is a device.

device channel: A Dialogic® voice data path that processes one incoming or outgoing call at a time (equivalent to the terminal equipment terminating a phone line).

device driver: Software that acts as an interface between an application and hardware devices.

device handle: Numerical reference to a device, obtained when a device is opened using **xx_open()**, where *xx* is the prefix defining the device to be opened. The device handle is used for all operations on that device.

device name: Literal reference to a device, used to gain access to the device via an **xx_open()** function, where *xx* is the prefix defining the device to be opened.

digitize: The process of converting an analog waveform into a digital data set.

DM3: Refers to Dialogic® mediastream processing architecture, which is open, layered, and flexible, encompassing hardware as well as software components. A whole set of products from Dialogic are built on the Dialogic® DM3 architecture. Contrast with [Springware](#), which is earlier-generation architecture.

download: The process where board level program instructions and routines are loaded during board initialization to a reserved section of shared RAM.

driver: A software module which provides a defined interface between an application program and the firmware interface.

DTMF (Dual-Tone Multi-Frequency): Push-button or touch-tone dialing based on transmitting a high- and a low-frequency tone to identify each digit on a telephone keypad.

echo: The component of an analog device's receive signal reflected into the analog device's transmit signal.

echo cancellation: Removal of echo from an echo-carrying signal.

event: An unsolicited or asynchronous message from a hardware device to an operating system, application, or driver. Events are generally attention-getting messages, allowing a process to know when a task is complete or when an external event occurs.

event handler: A portion of an application program designed to trap and control processing of device-specific events.

extended attribute functions: A class of functions that take one input parameter (a valid Dialogic® device handle) and return device-specific information. For instance, a voice device's extended attribute function returns information specific to the voice devices. Extended attribute function names are case-sensitive and must be in capital letters. See also [standard runtime library \(SRL\)](#).

firmware: A set of program instructions that reside on an expansion board.

firmware load file: The firmware file that is downloaded to a voice board.

flash: A signal generated by a momentary on-hook condition. This signal is used by the voice hardware to alert a telephone switch that special instructions will follow. It usually initiates a call transfer. See also [I/O](#).

G.726: An international standard for encoding 8 kHz sampled audio signals for transmission over 16, 24, 32 and 40 kbps channels. The G.726 standard specifies an adaptive differential pulse code modulation (ADPCM) system for coding and decoding samples.

GSM (Global System for Mobile Communications): A digital cellular phone technology based on time division multiple access (TDMA) used in Europe, Japan, Australia and elsewhere around the world.

I/O: Input-Output

idle device: A device that has no functions active on it.

in-band: The use of robbed-bit signaling (T1 systems only) on the network. The signaling for a particular channel or time slot is carried within the voice samples for that time slot, thus within the 64 kbps (kilobits per second) voice bandwidth.

kernel: A set of programs in an operating system that implement the system's functions.

mu-law: (1) Pulse Code Modulation (PCM) algorithm used in digitizing telephone audio signals in T1 areas. (2) The PCM coding and companding standard used in Japan and North America. See also [A-law](#).

PBX: Private Branch Exchange. A small version of the phone company's larger central switching office. A local premises or campus switch.

PCM (Pulse Code Modulation): A technique used in DSP voice boards for reducing voice data storage requirements. Dialogic supports either mu-law PCM, which is used in North America and Japan, or A-law PCM, which is used in the rest of the world.

polling: The process of repeatedly checking the status of a resource to determine when state changes occur.

PSTN (or STN): Public (or Private) Switched Telephony Network

resource: Functionality (for example, voice-store-and-forward) that can be assigned to a call. Resources are *shared* when functionality is selectively assigned to a call and may be shared among multiple calls. Resources are *dedicated* when functionality is fixed to the one call.

resource board: A Dialogic® expansion board that needs a network or switching interface to provide a technology for processing telecommunications data in different forms, such as voice store-and-forward, speech recognition, fax, and text-to-speech.

RFU: reserved for future use

ring detect: The act of sensing that an incoming call is present by determining that the telephone switch is providing a ringing signal to the voice board.

route: Assign a resource to a time slot.

sampling rate: Frequency at which a digitizer quantizes the analog voice signal.

SCbus (Signal Computing Bus): A hardwired connection between Switch Handlers on SCbus-based products. SCbus is a third generation TDM (Time Division Multiplexed) resource sharing bus that allows information to be transmitted and received among resources over 1024 time slots.

signaling insertion: The signaling information (on hook/off hook) associated with each channel is digitized, inserted into the bit stream of each time slot by the device driver, and transmitted across the bus to another resource device. The network interface device generates the outgoing signaling information.

silence threshold: The level that sets whether incoming data to the voice board is recognized as silence or non-silence.

SIT: (1) Standard Information Tones: tones sent out by a central office to indicate that the dialed call has been answered by the distant phone. (2) Special Information Tones: detection of a SIT sequence indicates an operator intercept or other problem in completing the call.

solicited event: An expected event. It is specified using one of the device library's asynchronous functions.

Springware: Software algorithms built into the downloadable firmware that provide the voice processing features available on older-generation Dialogic® voice boards. The term Springware is also used to refer to a whole set of boards from Dialogic built using this architecture. Contrast with [DM3](#), which is a newer-generation architecture.

SRL: See **Standard Runtime Library**.

standard attribute functions: Class of functions that take one input parameter (a valid device handle) and return generic information about the device. For instance, standard attribute functions return IRQ and error information for all device types. Standard attribute function names are case-sensitive and must be in capital letters. Standard attribute functions for Dialogic® devices are contained in the SRL. See [standard runtime library \(SRL\)](#).

standard runtime library (SRL): A Dialogic® software resource containing event management and standard attribute functions and data structures used by Dialogic® devices.

station device: Any analog telephone or telephony device (such as a telephone or headset) that uses a loop-start interface and connects to a station interface board.

string: An array of ASCII characters.

subdevice: Any device that is a direct child of another device. Since “subdevice” describes a relationship between devices, a subdevice can be a device that is a direct child of another subdevice, as a channel is a child of a board.

synchronous function: Blocks program execution until a value is returned by the device. Also called a blocking function. Contrast with [asynchronous function](#).

system release: The software and user documentation provided by Dialogic that is required to develop applications.

TDM (Time Division Multiplexing): A technique for transmitting multiple voice, data, or video signals simultaneously over the same transmission medium. TDM is a digital technique that interleaves groups of bits from each signal, one after another. Each group is assigned its own time slot and can be identified and extracted at the receiving end. See also [time slot](#).

TDMA (Time Division Multiple Access): A method of digital wireless communication using time division multiplexing.

TDM bus: Time division multiplexing bus. A resource sharing bus such as the SCbus or CT Bus that allows information to be transmitted and received among resources over multiple data lines.

termination condition: An event or condition which, when present, causes a process to stop.

termination event: An event that is generated when an asynchronous function terminates. See also [asynchronous function](#).

time division multiplexing (TDM): See [TDM \(Time Division Multiplexing\)](#).

time slot: The smallest, switchable data unit on a TDM bus. A time slot consists of 8 consecutive bits of data. One time slot is equivalent to a data path with a bandwidth of 64 kbps. In a digital telephony environment, a normally continuous and individual communication (for example, someone speaking on a telephone) is (1) digitized, (2) broken up into pieces consisting of a fixed number of bits, (3) combined with pieces of other individual communications in a regularly repeating, timed sequence (multiplexed), and (4) transmitted serially over a single telephone line. The process happens at such a fast rate that, once the pieces are sorted out and put back together again at the receiving end, the speech is normal and continuous. Each individual, pieced-together communication is called a time slot.

time slot assignment: The ability to route the digital information contained in a time slot to a specific analog or digital channel on an expansion board. See also [device channel](#).

underrun: data is not being delivered to the board quickly enough which can result in loss of data and gaps in the audio

virtual board: In the traditional voice processing board environment, the device driver views a single physical voice board with more than four channels as multiple emulated D/4x boards. These emulated boards are called virtual boards. This concept extends to the Dialogic[®] Host Media Processing (HMP) Software environment. A system with 44 channels consists of 11 virtual boards.

voice processing: The science of converting human voice into data that can be reconstructed and played back at a later time.

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