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1. Voice Software Reference
Overview

1.1. Voice Product Terminology

The following product naming conventions are used throughout this guide:

**D/2x** refers to any model of the Dialogic DIALOG series of 2-channel voice-
store-and-forward expansion boards. This series includes **D/21D, D41/E, and D41ESC** boards.

**D/4x** refers to any model of the Dialogic DIALOG series of 4-channel voice-
store-and-forward expansion boards. This series includes the **D/41D, D/41E, and D/41ESC** boards.

**D/12x** refers to any model of the Dialogic series of 12-channel voice-
store-and-forward expansion boards. **D/120, D/121, D/121A, and D/121B** are specific models of this board.

**D/81A** refers to the Dialogic 8-channel voice-store-and-forward expansion board.

**D/160SC** refers to the Dialogic 16-channel voice board with onboard analog loop
start interface.

**D/240SC** refers to the Dialogic 24-channel voice board for use with a network
interface board.

**D/240SC-T1** refers to the Dialogic 24-channel voice board with onboard T-1
digital interface.

**D/300SC-E1** refers to the Dialogic 30-channel voice board with onboard E-1
digital interface.

**D/320SC** refers to the Dialogic 32-channel voice board for use with a network
interface board.
D/xxx refers to D/2x, D/4x, D/81A and D/12x expansion boards.

D/xxxSC refers to voice and telephone network interface resource boards that communicate via the SCbus. These boards include D/41ESC, D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1, and D/320SC.

DIALOG/HD or Spancard refers to voice and telephone network interface resource boards that communicate via the SCbus. These boards include D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1, and D/320SC.

DTI/xxx refers to any of Dialogic’s digital telephony interface expansion boards for the AT-bus architecture. These boards include: DTI/101, DTI/211, DTI/212, DTI/240SC, and DTI/300SC boards.

FAX/xxx refers to Dialogic’s FAX resource expansion boards. FAX/120 is a 12-channel model that connects to a D/121A or D/121B board.

Firmware Load File refers to the firmware file that is downloaded to a Voice board. This file has a .fwl extension.

LSI refers to Dialogic’s PEB-based loop start interface expansion boards. The LSI/120 is a specific model of this board. LSI/80-int refers to the international versions of Dialogic’s loop start interface expansion boards.

PEB is the PCM expansion bus connecting the D/81A or D/12x voice boards to the network interface boards.

SCbus is the TDM (Time Division Multiplexed) bus connecting SCSA (Signal Computing System Architecture) voice, telephone network interface and other technology resource boards together.

Spancard same as DIALOG/HD.

SpringBoard refers to the hardware platform used with the D/21D, D/41D, D/21E, D/41E, D/81A, D/121, D/121A, and D/121B board.

SpringWare refers to the software algorithms built into the downloadable firmware that provides the voice processing features available on all Dialogic voice boards.

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1. Voice Software Reference Overview

**VFX/40ESC** is a Dialogic SCbus voice and FAX resource board with on-board loop-start interfaces. The **VFX/40ESC** board provides 4-channels of enhanced voice and FAX services in a single slot.

**Voice** hardware and software refers to D/2x, D/4x, D/81A, D/12x, and D/xxxSC expansion boards and associated software.

1.2. Organization of This Voice Reference Guide


**Chapter 1. Voice Software Reference Overview** provides an overview of the voice software. It lists each of the components and supported Dialogic boards and describes the Voice Driver and Voice Libraries.

**Chapter 2. Using the Voice Reference Library** provides general information about the Voice Library *libdxxmt.lib*. It provides an overview of the function categories, and describes the programming requirements when using the library.

**Chapter 3. Voice Function Reference** provides a complete function reference (in alphabetical order) for all of the functions in the Voice Library.

**Chapter 4. Voice Data Structures and Device Parameters** describes the following topics:

- data structures and tables contained in the Voice Library.
- parameter defines for Voice Devices that can be set or retrieved using `dx_getparm()` and `dx_setparm()`.

**Chapter 5. Voice Programming Conventions** lists programming techniques that simplify programming with the Dialogic Voice Library.

*Appendix A* lists the voice device entries and returns for the Standard Runtime Library.

*Appendix B* list the Voice Library error defines.
Appendix C describes differences and similarities between DTMF and MF tones.

Appendix D provides a list of related Dialogic publications.

A Glossary and an Index are also provided.

This chapter provides an overview of the voice software for Windows NT. The voice software consists of the following:

- Voice Driver
- Voice Library of C functions
- Standard Runtime Library

The Voice Driver and Voice Libraries are described in this chapter. For information about the remaining voice software see the following:

- For installation of the voice software and demonstration program, see the System Release Software Installation Reference for Windows NT
- For the Standard Runtime Library, see Appendix A and the Standard Runtime Library Programmer’s Guide for Windows NT
- For Voice Driver Features and a description of the Demonstration program, see the Voice Features Guide for Windows NT

1.3. Voice Driver

The Voice Driver communicates with and controls the voice hardware. Voice hardware consists of voice store-and-forward boards which include the following boards:

For SCbus-based applications:

- D/41ESC
- D/160SC-LS
- D/240SC, D/320SC
- D/240SC-T1, D/300SC-E1
- DTI/241SC, DTI/301SC
- LSI/81SC, LSI/161SC
The D/41ESC, D/160SC-LS, D/240SC, D/320SC, D/240SC-T1, and D300SC-E1 boards support a range of Voice Processing features such as:

- Record and playback of voice data
- Speed and volume control of play
- Call handling
- Call Analysis - Basic and Enhanced
- DTMF, MF, and R2MF tone generation and detection
- Global Tone Generation and Detection

PerfectCall Call Analysis, Speed and Volume Control, Global Tone Detection, and Global Tone Generation are supported on the DSP-based D/xxx boards (D/21D, D41D, D/21E, D/41ESC, D81A, D/121, D/121A, D/121B, D/160SC-LS, D/240SC, D240SC-T1, D300SC-E1, and D/320SC).

The DTI/241SC, DTI/301SC, LSI/81SC, and LSI/161SC boards support the following Voice Processing features:

- Call handling
- Call Analysis - Basic and Enhanced
- DTMF, MF, and R2MF tone generation and detection
- Global Tone Generation and Detection

PerfectCall Call Analysis, Speed and Volume Control, Global Tone Detection, and Global Tone Generation are supported on the DSP-based D/xxx boards (D/21D, D41D, D/21E, D/41ESC, D81A, D/121, D/121A, D/121B, D/160SC-LS, D/240SC, D240SC-T1, D300SC-E1, and D/320SC).

User-defined tones are CST events, but detection for these events is enabled using `dx_addtone()` or `dx_enbtone()`. See the Voice Features Guide for information on Global Tone Detection functions.

Boards are treated as board devices and channels are treated as channel devices or board subdevices by the Voice Driver.

The SCbus is a real-time, high speed, time division multiplexed (TDM) communications bus connecting Signal Computing System Architecture (SCSA) voice, telephone network interface and other technology resource boards together. SCbus boards are treated as board devices with on-board voice and/or telephone network interface devices that are identified by a board and channel (time slot for...
digital network channels) designation, such as a voice channel, analog channel or digital channel.

For more information on the SCbus and SCbus routing, refer to the SCbus Routing Guide and the SCbus Routing Function Reference for Windows NT.

1.4. Voice Libraries

The voice libraries provide the interface to the Voice Driver. The voice libraries for single threaded and multithreaded applications include:

- libdxxmt.lib - the main Voice Library
- libsrml.lib - the Standard Runtime Library

These "C" function libraries can be used to:

- Utilize all the voice board features
- Write applications using a Single Threaded Synchronous or Multithreaded Asynchronous programming model
- Configure devices
- Handle events that occur on the devices
- Return device information

The voice library libdxxmt.lib, which contains most of these functions, is described in more detail in 2. Using the Voice Reference Library.

The Standard Runtime Library libsrml.lib is described in the Standard Runtime Library Programmer's Guide for Windows NT. This library provides a set of common system functions that are device independent and are applicable to all Dialogic devices (e.g., D/240SC-T1 and FAX/120 boards). You can use these functions to simplify application development by writing common event handlers to be used by all devices.
1. **Voice Software Reference Overview**

1.4.1. **Single Threaded Asynchronous Programming Model**

Single threaded asynchronous programming enables a single program to control multiple voice channels within a single thread. This allows the development of complex applications where multiple tasks must be coordinated simultaneously. The asynchronous programming model supports both polled and callback event management.


1.4.2. **Multithreaded Synchronous Programming Model**

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


1.4.3. **Extended Asynchronous Programming Model**

This model is similar to the asynchronous except it is implemented using the `sr_waitEvEx()` function. This allows an application to have different threads waiting on events on different devices. As with the basic asynchronous model, functions initiated asynchronously from a different thread and the completion event picked up the `sr_waitEvEx()` thread.

## 2. Using the Voice Reference Library

This chapter provides a description of the voice library and the programming requirements. The following topics are included:

- Voice Library and its function categories (*Section 2.1*)
- Programming requirements for the Voice Library (*Section 2.2*).

### 2.1. Voice Library

The Voice Library functions provide an interface to the Voice Device Driver. The functions can be divided into the following major categories:

<table>
<thead>
<tr>
<th>Category</th>
<th>Functions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Management</td>
<td>open and close devices</td>
</tr>
<tr>
<td>Configuration</td>
<td>alter configuration of devices</td>
</tr>
<tr>
<td>I/O</td>
<td>transfer data to and from devices</td>
</tr>
<tr>
<td>Convenience</td>
<td>simplify play and record</td>
</tr>
<tr>
<td>Call Status Transition</td>
<td>set and monitor events on devices</td>
</tr>
<tr>
<td>Event</td>
<td>for SCbus boards, connect the receive (listen) channel of an SCbus board</td>
</tr>
<tr>
<td>Route</td>
<td>to an SCbus time slot; the transmit of each channel device is connected to</td>
</tr>
<tr>
<td></td>
<td>a unique and unchangeable SCbus time slot at system initiation and download.</td>
</tr>
<tr>
<td>Global Tone Detection</td>
<td>enable user-defined tone detection</td>
</tr>
<tr>
<td>Global Tone Generation</td>
<td>enable user-defined tone generation</td>
</tr>
<tr>
<td>R2MF Convenience</td>
<td>detect and generate R2MF tones</td>
</tr>
<tr>
<td>Speed and Volume</td>
<td>enable play-speed and play-volume control</td>
</tr>
<tr>
<td>Convention</td>
<td>convenience functions for adjusting speed and volume control</td>
</tr>
<tr>
<td>Structure Clearance</td>
<td>clear data structures</td>
</tr>
<tr>
<td>Extended Attribute</td>
<td>retrieve device information</td>
</tr>
</tbody>
</table>

This section lists the functions that belong to each category and describes the characteristics of each category.
In the *Function Reference* (3. *Voice Function Reference*) each function is described in detail, and the function header includes the category to which the function belongs.

### 2.1.1. Device Management Functions

<table>
<thead>
<tr>
<th>Function</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>dx_close()</code></td>
<td>close a board or channel</td>
</tr>
<tr>
<td><code>dx_open()</code></td>
<td>open a board or channel</td>
</tr>
</tbody>
</table>

The Device Management functions open and close devices (boards and channels). For SCbus configurations using a D/240SC-T1 or D/300SC-E1 board, each board comprises a digital interface device with independent channels/time slots (dtiBxTx) and a voice device with independent channels (dxxxBxCx); where B is followed by the unique board number, C is followed by the number of the voice device channel (1 to 4) and T is followed by the number of the digital interface device time slot (digital channel)(1 to 24 for T-1; 1 to 30 for E-1).

Before you can use any of the other library functions on a device, that device must be opened. When the device is opened using `dx_open()` the function returns a unique Dialogic device handle. The handle is the only way the device can be identified once it has been opened. The `dx_close()` function closes a device via its handle.

Device Management functions do not cause a device to be busy. In addition, the Device Management functions will work on a device whether the device is busy or idle.

**NOTES:**

1. Issuing a `dt_open()`, `dx_open()`, `dt_close()` or `dx_close()` while the device is being used by another process will not affect the current operation of the device.

2. The device handle which is returned is Dialogic defined. The device handle is not a standard Windows NT file descriptor. Any attempts to use operating system commands such as `read()`, `write()`, or `ioctl()` will produce unexpected results.

3. In an application that starts a process, the device handle is not inheritable by the child process. Devices must be opened in the child process.
2. Using the Voice Reference Library

2.1.2. Configuration Functions

<table>
<thead>
<tr>
<th>Function</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dx_clrdigbuf()</td>
<td>clear the firmware digit buffer</td>
</tr>
<tr>
<td>dx_getparm()</td>
<td>get a board/channel device parameter</td>
</tr>
<tr>
<td>dx_setdigtyp()</td>
<td>set digit collection type</td>
</tr>
<tr>
<td>dx_sethook()</td>
<td>set hookswitch state</td>
</tr>
<tr>
<td>dx_setparm()</td>
<td>set device parameters</td>
</tr>
<tr>
<td>dx_wtring()</td>
<td>wait for number of rings</td>
</tr>
</tbody>
</table>

Configuration functions allow you to alter, examine, and control the physical configuration of an open device. The configuration functions operate on a device only if the device is idle. All configuration functions cause a device to be busy and return the device to an idle state when the configuration is complete. See Section 2.2.3. Busy and Idle States for information about busy and idle states.

**NOTE:** The `dx_sethook()` function can also be classified as an I/O function and can be run asynchronously or synchronously.

2.1.3. I/O Functions

<table>
<thead>
<tr>
<th>Function</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dx_dial()</td>
<td>(enable/disable call analysis)</td>
</tr>
<tr>
<td>dx_getdig()</td>
<td>get digits from channel digit buffer</td>
</tr>
<tr>
<td>dx_play()</td>
<td>play voice data from one or more sources</td>
</tr>
<tr>
<td>dx_playiottdata()</td>
<td>play voice data from multiple sources</td>
</tr>
<tr>
<td>dx_rec()</td>
<td>record voice data to one or more destinations</td>
</tr>
<tr>
<td>dx_recioottdata()</td>
<td>record voice data to multiple destinations</td>
</tr>
<tr>
<td>dx_setdigbuf()</td>
<td>set digit buffering mode</td>
</tr>
<tr>
<td>dx_stopch()</td>
<td>stop current I/O</td>
</tr>
<tr>
<td>dx_wink()</td>
<td>wink a channel</td>
</tr>
</tbody>
</table>

**NOTES:**
1. `dx_playtone()`, which is grouped with the Global Tone generation
functions, is also an I/O function and all I/O characteristics apply.

2. **dx_sethook()**, which is grouped with the Configuration functions, is also an I/O function and all I/O characteristics apply.

3. **dx_wink()**, cannot be called for a digital T-1 configuration that includes a D/240SC-T1 board. Transparent signaling for SCbus digital interface devices is not supported in System Release 4.1SC.

The purpose of an I/O function is to transfer data to and from an open idle channel. All I/O functions cause a channel to be busy while data transfer is taking place and return the channel to an idle state when data transfer is complete. The **dx_stopch()** function stops any other I/O function, except **dx_dial()** (see **dx_dial()** and **dx_stopch()** in the Chapter 3. Voice Function Reference for information).

I/O functions can be run synchronously or asynchronously. When running synchronously, they return after completing successfully or after an error. When running asynchronously they will return immediately to indicate successful initiation (or an error), and continue processing until a termination condition is satisfied. See the Standard Runtime Library Programmer's Guide for Windows NT, for a full discussion on asynchronous and synchronous operation.

A set of termination conditions can be specified for I/O functions (except **dx_stopch()** and **dx_wink()**). These conditions dictate what events will cause an I/O function to terminate. The termination conditions are specified just before the I/O function call is made. Obtain termination reasons for I/O functions by calling the Extended Attribute function **ATDX_TERMMSK()**. See Section 2.2.4. I/O Terminations for information on I/O terminations.

**NOTE:** The **dx_stopch()** function will not stop all I/O functions. Do not use this function to stop **dx_wink()** or **dx_dial()** (without Call Analysis enabled). See Chapter 3. Voice Function Reference for more information on these functions.
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2.1.4. Convenience Functions

<table>
<thead>
<tr>
<th>Function</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dx_playf()</td>
<td>play voice data from a single file</td>
</tr>
<tr>
<td>dx_playvox()</td>
<td>play a VOX file</td>
</tr>
<tr>
<td>dx_playwav()</td>
<td>play a WAVE file</td>
</tr>
<tr>
<td>dx_recf()</td>
<td>record voice data to a single file</td>
</tr>
<tr>
<td>dx_recvox()</td>
<td>record voice data to a single VOX file</td>
</tr>
<tr>
<td>dx_recwav()</td>
<td>record voice data to a single WAVE file</td>
</tr>
</tbody>
</table>

These functions simplify synchronous play and record.

`dx_playf()` performs a playback from a single file by specifying the filename. The same operation can be done by using `dx_play()` and supplying 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_playvox()`, `dx_playwav()`, `dx_recvox()`, `dx_recwav()`, and `dx_recf()` provide the same single-file convenience for the `dx_playiottdata()`, `dx_reciottdata()`, and `dx_rec()` function.

Source code is included for `dx_playf()` and `dx_recf()` in the function descriptions in Chapter 3, Voice Function Reference.

**NOTE:** `dx_play()`, `dx_playvox()`, `dx_playwav()`, `dx_recf()`, `dx_recvoxx()`, and `dx_recwav()` run synchronously only.

2.1.5. Call Status Transition Event Functions

<table>
<thead>
<tr>
<th>Function</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dx_getevt()</td>
<td>get call status transition event</td>
</tr>
<tr>
<td>dx_setevtsk()</td>
<td>set call status transition event notification</td>
</tr>
</tbody>
</table>

Call Status Transition (CST) Event functions set and monitor Call Status Transition events that can occur on a device. Call Status Transition events indicate changes in the status of the call. For example, if rings were detected, if the line went onhook or offhook, or if a tone was detected. The full list of Call Status
Transition events is contained in Section 4.1.3. DX_CST - call status transition structure which describes the Call Status Transition structure (DX_CST).

\texttt{dx\_setevtmsk()} enables detection of CST event(s).

\texttt{dx\_getevt()} retrieves events in a synchronous environment. To retrieve CST events in an asynchronous environment, use the Standard Runtime Library’s Event Management functions.

### 2.1.6. SCbus Routing Functions

See the \textit{SCbus Routing Function Reference for Windows NT} for function descriptions and the nomenclature used to identify devices, channels and time slots in an SCbus configuration. The SCbus routing functions can only be used in SCbus configurations.

### 2.1.7. Global Tone Detection Functions

<table>
<thead>
<tr>
<th>Function</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>\texttt{dx_addtone()}</td>
<td>add a user-defined tone</td>
</tr>
<tr>
<td>\texttt{dx_blddt()}</td>
<td>build a dual frequency tone description</td>
</tr>
<tr>
<td>\texttt{dx_bldtcad()}</td>
<td>build a dual frequency tone cadence description</td>
</tr>
<tr>
<td>\texttt{dx_bldest()}</td>
<td>build a single frequency tone description</td>
</tr>
<tr>
<td>\texttt{dx_bldestcad()}</td>
<td>build a single frequency tone cadence description</td>
</tr>
<tr>
<td>\texttt{dx_deltones()}</td>
<td>delete user-defined tones</td>
</tr>
<tr>
<td>\texttt{dx_enbtone()}</td>
<td>enable detection of user-defined tones</td>
</tr>
<tr>
<td>\texttt{dx_distone()}</td>
<td>disable detection of user-defined tones</td>
</tr>
<tr>
<td>\texttt{dx_setgttdamp()}</td>
<td>sets amplitudes used by Global Tone Detection (GTD)</td>
</tr>
</tbody>
</table>

Use the Global Tone Detection (GTD) functions to define and enable detection of single and dual frequency tones that fall outside those automatically provided with the Voice Driver. This includes tones outside the standard DTMF range of 0-9, a-d, *, and #.
2. Using the Voice Reference Library

The GTD \texttt{dx_blddt( )}, \texttt{dx_blddtcad( )}, \texttt{dx_bldst( )}, and \texttt{dx_bldstcad( )} functions define tones which can then be added to the channel using \texttt{dx_addtone( )}. This enables detection of the tone on that channel.

See the \textit{Voice Features Guide for Windows NT} for a full description of Global Tone Detection.

2.1.8. Global Tone Generation Functions

| \texttt{dx_bldtngen( )} | • build a user-defined tone generation template |
| \texttt{dx_playtone( )} | • play a user-defined tone |

Use Global Tone Generation functions to define and play single and dual tones other than those automatically provided with the Voice driver.

\texttt{dx_bldtngen( )} defines a tone template structure, \textit{TN\_GEN}. \texttt{dx_playtone( )} can then be used to generate the tone.

See the \textit{Voice Features Guide for Windows NT} for a full description of Global Tone Generation, and see 4. Voice Data Structures and Device Parameters for a description of the \textit{TN\_GEN} structure.

2.1.9. R2MF Convenience Functions

| \texttt{r2_creatfsig( )} | • create R2MF forward signal tone |
| \texttt{r2_playbsig( )} | • play R2MF backward signal tone |

These are convenience functions which enable detection of R2MF forward signals on a channel, and play R2MF backward signals in response. For more information about Voice Support for R2MF, see the \textit{Voice Features Guide for Windows NT}.

2.1.10. Speed and Volume Functions

| \texttt{dx_adjsv( )} | • adjust speed or volume |
| \texttt{dx_clrsvcond( )} | • clear speed or volume digit adjustment conditions |
**Voice Programmer’s Guide for Windows NT**

**dx_setsvcond( )**  • set speed or volume digit adjustment conditions
**dx_getcursv( )**  • get current speed and volume settings
**dx_getsvmt( )**  • get Speed/Volume Modification Table
**dx_setsvmt( )**  • set Speed/Volume Modification Table

**NOTE:** Speed and Volume Control are available on D/41ESC, D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1, and D/320SC boards.

Use these functions to adjust the speed and volume of the play. A 21-entry Speed Modification Table and Volume Modification Table is associated with each channel. This table can be used for increasing or decreasing the speed or volume. This table has default values which can be changed using the **dx_setsvmt( )** function.

**dx_adjsv( )** and **dx_setsvcond( )** both use the Modification Table to adjust speed or volume; **dx_adjsv( )** adjusts speed or volume immediately, and
**dx_setsvcond( )** sets conditions (such as a digit) for speed or volume adjustment. **dx_clrsvcond( )** to clear the speed or volume conditions.

**dx_getcursv( )** retrieves the current speed or volume settings. **dx_getsvmt( )** retrieves the settings of the current Speed or Volume Adjustment Table.

See the *Voice Features Guide for Windows NT* for more information about voice software support for speed and volume.

### 2.1.11. Speed and Volume Convenience Functions

**dx_addspddig( )**  • add speed adjustment digit
**dx_addvoldig( )**  • add volume adjustment digit

**dx_addspddig( )** and **dx_addvoldig( )** are convenience functions that specify a digit and an adjustment to occur on that digit, without having to set any data structures. These functions use the default settings of the Speed/Volume Modification Tables.
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### 2.1.12. PerfectCall Call Analysis Functions

<table>
<thead>
<tr>
<th>Function</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dx_chgdur()</td>
<td>change PerfectCall Call Analysis signal duration</td>
</tr>
<tr>
<td>dx_chgfreq()</td>
<td>change PerfectCall Call Analysis signal frequency</td>
</tr>
<tr>
<td>dx_chgrepct()</td>
<td>change PerfectCall Call Analysis signal repetition count</td>
</tr>
<tr>
<td>dx_initcallp()</td>
<td>initialize PerfectCall Call Analysis on a channel</td>
</tr>
<tr>
<td>dx_chg()</td>
<td>functions can be used to change the definition of default PerfectCall Call Analysis tones.</td>
</tr>
<tr>
<td>dx_initcallp()</td>
<td>enables PerfectCall Call Analysis.</td>
</tr>
</tbody>
</table>

### 2.1.13. Structure Clearance Functions

<table>
<thead>
<tr>
<th>Function</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dx_clrcap()</td>
<td>clear DX_CAP structure</td>
</tr>
<tr>
<td>dx_clrtpt()</td>
<td>clear DV_TPT structure</td>
</tr>
</tbody>
</table>

These functions do not affect a device. The `dx_clrcap()` and `dx_clrtpt()` functions provide a convenient method for clearing the DX_CAP and DV_TPT Voice Library data structures. These structures are discussed in *Chapter 4. Voice Data Structures and Device Parameters*.

### 2.1.14. Extended Attribute Functions

<table>
<thead>
<tr>
<th>Function</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ATDX_ANSRSIZ()</td>
<td>Returns duration of answer detected during Call Analysis</td>
</tr>
<tr>
<td>ATDX_BDNAMEP()</td>
<td>Returns pointer to the device name string</td>
</tr>
<tr>
<td>ATDX_BDTYPE()</td>
<td>Returns board type</td>
</tr>
<tr>
<td>ATDX_BUFDIGS()</td>
<td>Returns number of digits in firmware since last <code>dx_getdig()</code> for a given channel</td>
</tr>
<tr>
<td>ATDX_CHNAMES()</td>
<td>Returns pointer to an array of channel name strings</td>
</tr>
<tr>
<td>ATDX_CHNUM()</td>
<td>Returns channel number on board associated with the channel device handle</td>
</tr>
<tr>
<td>Function</td>
<td>Description</td>
</tr>
<tr>
<td>----------------------------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>ATDX_CONNTYPE()</td>
<td>Returns connection type for a call</td>
</tr>
<tr>
<td>ATDX_CPERROR()</td>
<td>Returns call analysis error</td>
</tr>
<tr>
<td>ATDX_CPTERM()</td>
<td>Returns last call analysis termination</td>
</tr>
<tr>
<td>ATDX_CRTNID()</td>
<td>Returns the identifier of the tone that caused the most recent Call Analysis termination</td>
</tr>
<tr>
<td>ATDX_DEVTYPE()</td>
<td>Returns device type</td>
</tr>
<tr>
<td>ATDX_DTNFAIL()</td>
<td>Returns the dial tone character that indicates which dial tone Call Analysis failed to detect</td>
</tr>
<tr>
<td>ATDX_FRQDUR()</td>
<td>Returns duration of first frequency</td>
</tr>
<tr>
<td>ATDX_FRQDUR2()</td>
<td>Returns duration of 2nd SIT tone frequency</td>
</tr>
<tr>
<td>ATDX_FRQDUR3()</td>
<td>Returns duration of 3rd SIT tone frequency detected</td>
</tr>
<tr>
<td>ATDX_FRQHZ()</td>
<td>Returns frequency of first detected tone</td>
</tr>
<tr>
<td>ATDX_FRQHZ2()</td>
<td>Returns frequency of second detected SIT tone</td>
</tr>
<tr>
<td>ATDX_FRQHZ3()</td>
<td>Returns frequency of third detected SIT tone</td>
</tr>
<tr>
<td>ATDX_FRQOUT()</td>
<td>Returns % of frequency out of bounds detected during Call Analysis</td>
</tr>
<tr>
<td>ATDX_FWVER()</td>
<td>Returns firmware version</td>
</tr>
<tr>
<td>ATDX_HOOKST()</td>
<td>Returns current hook status</td>
</tr>
<tr>
<td>ATDX_LINEST()</td>
<td>Returns current line status</td>
</tr>
<tr>
<td>ATDX_LONGLOW()</td>
<td>Returns duration of longer silence detected during Call Analysis</td>
</tr>
<tr>
<td>ATDX_PHYADDR()</td>
<td>Returns physical address of board</td>
</tr>
<tr>
<td>ATDX_SHORTLOW()</td>
<td>Returns duration of shorter silence detected during Call Analysis</td>
</tr>
<tr>
<td>ATDX_SIZEHI()</td>
<td>Returns duration of non-silence detected during Call Analysis</td>
</tr>
<tr>
<td>ATDX_STATE()</td>
<td>Returns current state of the device</td>
</tr>
<tr>
<td>ATDX_TERMMSK()</td>
<td>Returns termination bitmap</td>
</tr>
</tbody>
</table>
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<table>
<thead>
<tr>
<th>Function</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ATDX_TONEID()</td>
<td>Returns the tone id</td>
</tr>
<tr>
<td>ATDX_TRCOUNT()</td>
<td>Returns last record or play transfer count</td>
</tr>
</tbody>
</table>

Voice Library Extended Attribute functions return information specific to the Voice device indicated in the function call. Many are related to specific Voice features:

Basic Call Analysis uses:

- ATDX_ANSRSIZ()
- ATDX_CPERROR()
- ATDX_CPTERM()
- ATDX_FRQ()
- ATDX_LONGLOW()
- ATDX_SHORTLOW()
- ATDX_SIZEHI()

PerfectCall Call Analysis uses:

- ATDX_ANSRSIZ()
- ATDX_CPERROR()
- ATDX_CPTERM()
- ATDX_FRQ()
- ATDX_CRTNID()
- ATDX_DTNFAIL()

The Call Status Transition event detection uses:

- ATDX_HOOKST()  

Global Tone Detection uses:

- ATDX_TONEID()
2.2. Voice Programming Requirements

This section contains information that is required when using the Voice Library and many of its functions. The following topics are covered:

- Opening and Using Devices (Section 2.2.1)
- Opening and Using Voice Channels (Section 2.2.2)
- Busy and Idle Device States (Section 2.2.3)
- I/O Terminations (Section 2.2.4)
- Error Handling (Section 2.2.5)
- Voice Library Include Files (Section 2.2.6)
- Compiling Applications (Section 2.2.7)

2.2.1. Opening and Using Devices

When you open a file under Windows NT, it returns a unique file descriptor for that file. The following is an example of a file descriptor:

```c
int file_descriptor;
file_descriptor = open(filename, mode);
```

Any subsequent action you wish to perform on that file is accomplished by identifying the file using `file_descriptor`. No action can be performed on the file until it is first opened.

Dialogic boards and channels work in a similar manner. You must first open a Voice device using `dx_open()` before you can perform any operation on it. When you open a channel using `dx_open()`, the value returned is a unique Dialogic device handle for that particular open process on that channel. The Dialogic channel device handle is referred to as `chdev`, where

```c
int chdev;
chdev = dx_open(channel_name, mode)
```

Any time you wish to use a Voice library function on the channel, you must identify the channel with its Dialogic channel device handle, `chdev`. The channel name is used only when opening a channel, and all actions after opening must use
the handle `chdev`. Board devices are opened by following the same procedure, where `bddev` refers to the Dialogic board device handle.

NOTE: As stated above, boards and channels are considered separate devices under Windows NT. It is possible to open and use a channel without ever opening the board it is on. There is no board-channel hierarchy imposed by the driver.

To enable users to control the boards and the channels under the Windows NT operating system, Dialogic provides a library of C language functions. For details on opening and closing channels and boards refer to the function references for `dx_open()` and `dx_close()` in Section 3.2. Voice Library Function Descriptions.

CAUTION
Dialogic devices should never be opened using the Windows NT `open()`.

2.2.2. Opening and Using Voice Files

The Voice library provides a set of standard I/O routines. Although applications may use the routines provided with the Microsoft C Runtime Library, Dialogic recommends that the application use the Dialogic file handling routines when manipulating voice files. These routines are `dx_fileopen()` for opening voice files, `dx_fileclose()` for closing voice files, and `dx_fileseek()`, `dx_fileread()`, and `dx_filewrite()` for searching for, reading, or writing directly to a file. The arguments for these functions are identical to the equivalent “C” runtime functions.

2.2.3. Busy and Idle States

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. The categories of functions and their state dependencies are described in the following sections.

2.2.4. I/O Terminations

Pass a set of termination conditions as one of the function parameters when an I/O function is issued. Termination conditions are events monitored during the I/O process that cause an I/O function to terminate. When the termination condition is met, `ATDX_TERMMSK()` returns the reason for termination. I/O functions can terminate under the following conditions:

- byte transfer count is satisfied
- device has stopped due to `dx_stopch()`
- end of file is reached during a play
- loop current has dropped for a period of time
- maximum delay between DTMF digits is detected
- maximum number of digits has been received
- maximum period of non-silence (noise or meaningful sound) has been detected
- maximum period of silence has been detected
- pattern of silence and non-silence (noise or meaningful sound) has been detected
- specific digit has been received
- I/O function has been executing for a maximum period of time
- user-defined digit has been received
- user-defined tone-on or tone-off has been detected (GTD)

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. The DV_TPT structure, which is defined in `srllib.h`, is described in detail in the `Standard Runtime Library Programmer’s Guide for Windows NT`. Voice board
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values for the DV_TPT are contained Appendix A. 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. See Section 4.1.5. DX_IOTT - I/O transfer table, for information about setting the number of bytes in the DX_IOTT.

**Stop Occurred** - dx_stopch() terminates any I/O function, except for dx_dial() without Call Analysis enabled, and dx_wink(). 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 causes 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. See Section 4.1.5. DX_IOTT - I/O transfer table for information about setting the DX_IOTT. ATDX_TERMMSK() returns the termination reason TM_EOD when this termination condition is met.

**Loop Current Drop** - In some central offices, switches, and PBX’s, a drop in loop current indicates disconnect supervision. An I/O function can terminate if the loop current drops for a specified amount of time. Specify the amount of time in the tp_length field of a DV_TPT structure in 100 ms units (default) or 10 ms units. Specify 10 ms in the tp_flags field of the DV_TPT structure. ATDX_TERMMSK() returns the termination reason TM_LCOFF when this termination condition is met.

**Maximum Delay Between Digits** - 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. Specify the amount of time in 100 ms units (default) or 10 ms units for the tp_length field or 10 ms units for the tp_flags field. ATDX_TERMMSK() returns the termination reason TM_IDDTIME when this termination condition is met.

**Maximum Digits Received** - 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 condition counts the digits remaining in the buffer as well. To set the maximum number of digits received before termination, place a number from 1 to 31 in the `tp_length` field of a DV_TPT. `ATDX_TERMMSK()` returns the termination reason TM_MAXDTMF when this termination condition is met.

**Maximum Length of Non-silence** - Non-silence is the absence of silence: noise or meaningful sound, such as a person speaking. Enable this condition by setting the `tp_length` field of a DV_TPT to a specific period of time. When the application detects non-silence for this length of time, the I/O function terminates. This termination condition is frequently used to detect dial tone or the howler tone that is used by central offices to indicate that a phone has been off-hook for an extended period of time. Specify the amount of time in 100 ms units (default) or 10 ms units in the `tp_length` field or 10 ms units in the `tp_flags` field of the DV_TPT structure. `ATDX_TERMMSK()` returns the termination reason TM_MAXNOSIL when this termination condition is met.

**Maximum Length of Silence** - Enable this termination condition 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 ms units (default) or 10 ms units for the `tp_length` field or 10 ms units in the `tp_flags` field of the DV_TPT structure. `ATDX_TERMMSK()` returns the termination reason TM_MAXSIL when this termination condition is met.

**Pattern of Silence and Non-silence** - A known pattern of silence and non-silence can terminate a function. A pattern can be specified by specifying DX_PMON and DX_PMOFF in the `tp_termno` field in two separate DV_TPT structures, where one represents a period of silence and one represents a period of non-silence. `ATDX_TERMMSK()` returns the termination reason TM_PATTERN when this termination condition is met.

DX_PMOFF/DX_PMON termination conditions must be used together. The DX_PMON terminating condition must directly follow the DX_PMOFF terminating condition. A combination of both DV_TPT structures using these conditions is used to form a single termination condition. A detailed description of how to set these termination conditions is described in Appendix A in a section called "Using DX_PMON and DX_PMOFF."
Specific Digit Received - An application collects the digits received during an I/O function in a channel’s digit buffer. If the buffer is not empty before an I/O function executes, the application treats the digits in the buffer as if the digits were received during the I/O execution. Enable this termination condition 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. `ATDX_TERMMSK()` returns the termination reason TM_DIGIT when this termination condition is met.

Maximum Function Time - Place a time limit on the I/O function by setting the `tp_length` field of a DV_TPT to a specific length of time in 100 ms units. The I/O function terminates when it executes longer than this period of time. Specify the amount of time in 100 ms units (default) or 10 ms units for the `tp_length` field and 10 ms units in the `tp_flags` field of the DV_TPT. `ATDX_TERMMSK()` returns the termination reason TM_MAXTIME when this termination condition is met.

User-Defined Digit Received - An application collects user-defined digits in a channel’s digit buffer during an I/O function. If the buffer is not empty before an I/O function executes, the application treats the digits in the buffer as if 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 terminates. `ATDX_TERMMSK()` returns the termination reason TM_DIGIT when this termination condition is met.

User-Defined Tone On/Off Event Detected - Use this termination condition with Global Tone Detection. Before specifying a user-defined tone as a termination condition, define the tone using the GTD `dx_bld...( )` functions, and enable the tone detection on the channel using the `dx_addtone()` or `dx_enbtone()` functions. 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. `ATDX_TERMMSK()` returns the termination reason TM_TONE when this termination condition is met.

The application may clear the DV_TPT structure using `dx_clrtpt()` before initializing the structure and passing a pointer to it as a function parameter.
Refer to the *Standard Runtime Library Programmer’s Guide for Windows NT* for a complete discussion of the DV_TPT structure, and to Appendix A for a description of Voice software values for the DV_TPT.

### 2.2.5. Error Handling

All the Dialogic Voice Library functions return a value to indicate success or failure of the function. All Voice Library functions indicate success by a return value of zero or a non-negative number.

Extended Attribute functions that return pointers return a pointer to the ASCIIZ string "Unknown device" if they fail.

Extended Attribute functions that don’t return pointers, return a value of AT_FAILURE if they fail.

All other functions return a value of -1 to indicate a failure.

If a function fails, call the Standard Attribute functions `ATDV_LASTERR()` and `ATDV_ERRMSGP()` for the reason for failure. These functions are described in the *Standard Runtime Library Programmer’s Guide for Windows NT*.

The errors that can be returned by a Voice Library function are listed in Table 1. These errors are also listed for reference in Appendix B.

**NOTES:**
1. `dx_open()` and `dx_close()` are exceptions to the above error handling rules. If these functions fail, the return code is -1 and the specific error is found in the `errno` variable contained in `errno.h`.
2. If `ATDV_LASTERR()` returns the error EDX_SYSTEM, a Windows NT system error has occurred. Check the global variable `errno` contained in `errno.h`.

**Table 1. Voice Library Function Errors**

<table>
<thead>
<tr>
<th>Error Define</th>
<th>Error String</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_AMPLGEN</td>
<td>Invalid Amplitude Value in Tone Generation</td>
</tr>
<tr>
<td></td>
<td>Template [+2dB - -40dB]</td>
</tr>
</tbody>
</table>

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2. Using the Voice Reference Library

<table>
<thead>
<tr>
<th>Error Define</th>
<th>Error String</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_ASCII</td>
<td>Invalid ASCII Value in Tone Template Description</td>
</tr>
<tr>
<td>EDX_BADDEV</td>
<td>Invalid Device Descriptor</td>
</tr>
<tr>
<td>EDX_BADIOTT</td>
<td>Invalid Entry in the \textit{DX_IOTT}</td>
</tr>
<tr>
<td>EDX_BADPARM</td>
<td>Invalid Parameter in Function Call</td>
</tr>
<tr>
<td>EDX_BADPROD</td>
<td>Function Not Supported on this Board</td>
</tr>
<tr>
<td>EDX_BADTPT</td>
<td>Invalid Entry in the DX_TPT</td>
</tr>
<tr>
<td>EDX_BADWAVFILE</td>
<td>Invalid WAV file</td>
</tr>
<tr>
<td>EDX_BUSY</td>
<td>Device is Already Busy</td>
</tr>
<tr>
<td>EDX_CADENCE</td>
<td>Invalid Cadence Component Values in Tone Template Description</td>
</tr>
<tr>
<td>EDX_CHANNUM</td>
<td>Invalid Channel Number Specified</td>
</tr>
<tr>
<td>EDX_DIGTYPE</td>
<td>Invalid Dig_type Value in Tone Template Description</td>
</tr>
<tr>
<td>EDX_FLAGGEN</td>
<td>Invalid tn_dflag field in Tone Generation Template</td>
</tr>
<tr>
<td>EDX_FREQDET</td>
<td>Invalid Freq Component Values in Tone Template Description</td>
</tr>
<tr>
<td>EDX_FREQGEN</td>
<td>Invalid Frequency Component in Tone Generation Template [200hz - 2000hz]</td>
</tr>
<tr>
<td>EDX_FWERROR</td>
<td>Firmware Error</td>
</tr>
<tr>
<td>EDX_IDLE</td>
<td>Device is Idle</td>
</tr>
<tr>
<td>EDX_INVSUBCM</td>
<td>Invalid Sub Command Number</td>
</tr>
<tr>
<td>EDX_MAXTMLT</td>
<td>Max number of Templates Exists</td>
</tr>
<tr>
<td>EDX_MSGSTATUS</td>
<td>Invalid Message Status Setting</td>
</tr>
</tbody>
</table>
### Error Define | Error String
---|---
EDX_NOERROR | No Errors
EDX_NONZEROSIZE | Reset to Default was Requested but size was non-zero
EDX_SH_BADCMD | Unsupported command or WAV file format
EDX_SPDVOL | Must Specify either SV_SPEEDTBL or SV_VOLMETBL
EDX_SVADJBLKS | Invalid Number of Speed/Volume Adjustment Blocks
EDX_SVMTRANGE | An Entry in DX_SVMT was out of Range
EDX_SVMTSIZE | Invalid Table Size Specified
EDX_SYSTEM | System Error
EDX_TIMEOUT | Function Timed Out
EDX_TONEID | Bad Tone Template ID

#### 2.2.6. Voice Library Include Files

The following lines must be included in application code prior to calling Voice Library functions:

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

**NOTE:** `srllib.h` must be included in code before all other Dialogic header files.

The libraries are located in the following default directories:

```
<install drive:><install directory>\dialogic\inc
```
2. Using the Voice Reference Library

2.2.7. Compiling Applications

Application programs developed using the Voice Library for Windows NT should be linked with the following libraries.

Libraries for multithreaded applications are located in the following default directories:

<install drive:><install directory>\dialogic\i386\lib\libdxxmt.lib - the main Voice Library
<install drive:><install directory>\dialogic\i386\lib\libsrlmt.lib - the Standard Runtime Library

They should be linked in the order specified above.
Voice Programmer's Guide for Windows NT
3. Voice Function Reference

3.1. Voice Function Reference Overview

This chapter provides a description of the Voice Library functions provided with the voice software for Windows NT systems. These functions are listed alphabetically for ease of use.

Some Voice Library functions use special structures. These structures are defined in the dxxxlib.h and srllib.h header files and are described in Chapter 4. Voice Data Structures and Device Parameters. They include the following:

- **DV_DIGIT**: User Digit Buffer Structure
- **DV_TPT**: Termination Parameter Table Structure
- **DX_CAP**: Call Analysis Parameter Structure
- **DX_EBLK**: Event Block Structure
- **DX_IOTT**: I/O Transfer Table Structure
- **DX_UIO**: I/O User-definable I/O Structure

The **dx_getparm( )** and **dx_setparm( )** functions use defined masks to specify parameters. The definitions of the masks are found in Chapter 4. Voice Data Structures and Device Parameters.

Applications that use the Voice Library must include the dxxxlib.h and srllib.h header files.

**NOTE:** The srllib.h header file must always be listed before any other Dialogic header file.

SCbus Functions

See the SCbus Routing Function Reference for Windows NT for function descriptions and the nomenclature used to identify devices, channels and time slots in an SCbus configuration. The SCbus routing functions can only be used in SCbus configurations.

3.2. Voice Library Function Descriptions
ATDX_ANSRSIZ() returns the duration of the answer

<table>
<thead>
<tr>
<th>Name:</th>
<th>long ATDX_ANSRSIZ(chdev)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>int chdev • valid Dialogic channel device handle</td>
</tr>
<tr>
<td>Returns:</td>
<td>answer duration in 10 ms units if successful AT_FAILURE if error</td>
</tr>
<tr>
<td>Includes:</td>
<td>srllib.h dxxxlib.h</td>
</tr>
<tr>
<td>Category:</td>
<td>Extended Attribute</td>
</tr>
</tbody>
</table>

**Description**

The ATDX_ANSRSIZ() function returns the duration of the answer that occurs when dx_dial() with Basic Call Analysis enabled is called on a channel. An answer is considered the period of non-silence that begins after cadence is broken and a connection is made. This measurement is taken before a connect event is returned. The duration of the answer can be used to determine if the call was answered by a person or an answering machine. This feature is based on the assumption that an answering machine typically answers a call with a longer greeting than a live person does.

See the section called "Cadence Detection Parameters’ Affect on a Connect" in the Voice Features Guide for Windows NT for information about distinguishing between a person and answering machine.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using dx_open().</td>
</tr>
</tbody>
</table>

**Cautions**

None.

**Example**

```c
/* Call Analysis with user-specified parameters */
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
```
returns the duration of the answer ATDX_ANSRSIZ()

#include <windows.h>

main()
{
    int cares, chdev;
    DX_CAP capp;

    /* open the channel using dx_open(). Obtain channel device descriptor in */
    /* chdev */
    if ((chdev = dx_open("dxxxB1C1", NULL)) == -1) {
        /* process error */
    }

    /* take the phone off-hook */
    if (dx_sethook(chdev, DX_OFFHOOK, EV_SYNC) == -1) {
        /* process error */
    }

    /* Set the DX_CAP structure as needed for call analysis. Perform the */
    /* outbound dial with call analysis enabled */
    if ((cares = dx_dial(chdev, "5551212", &capp, DX_CALLP|EV_SYNC)) == -1) {
        /* perform error routine */
    }

    switch (cares) {
    case CR_CNCT:     /* Call Connected, get some additional info */
        printf("Duration of short low - %ld ms", ATDX_SHORTLOW(chdev)*10);
        printf("Duration of long low  - %ld ms", ATDX_LONGLOW(chdev)*10);
        printf("Duration of answer    - %ld ms", ATDX_ANSRSIZ(chdev)*10);
        break;
    case CR_CEPT:     /* Operator Intercept detected */
        printf("Frequency detected - %ld Hz", ATDX_FRQHZ(chdev));
        printf("% of Frequency out of bounds - %ld Hz", ATDX_FRQOUT(chdev));
        break;
    case CR_BUSY:
    }

    switch (cares) {
    case CR_CNCT:     /* Call Connected, get some additional info */
        printf("Duration of short low - %ld ms", ATDX_SHORTLOW(chdev)*10);
        printf("Duration of long low  - %ld ms", ATDX_LONGLOW(chdev)*10);
        printf("Duration of answer    - %ld ms", ATDX_ANSRSIZ(chdev)*10);
        break;
    case CR_CEPT:     /* Operator Intercept detected */
        printf("Frequency detected - %ld Hz", ATDX_FRQHZ(chdev));
        printf("% of Frequency out of bounds - %ld Hz", ATDX_FRQOUT(chdev));
        break;
    case CR_BUSY:
    }

    Errors

    This function will fail and return AT_FAILURE if an invalid channel device
    handle is specified in chdev.

    See Also

    Related to Call Analysis:
ATDX_ANSRSIZ() returns the duration of the answer

- dx_dial()
- DX_CAP. (Chapter 4. Voice Data Structures and Device Parameters)
- "Call Analysis" (Voice Features Guide for Windows NT)
ATDX_BDNAMESP() returns a pointer

Name: char * ATDX_BDNAMESP(chdev)

Inputs: int chdev • valid Dialogic channel device handle

Returns: pointer to Board device name string if successful
        pointer to ASCIIZ string "Unknown device" if error

Includes: srllib.h
dxxxlib.h

Category: Extended Attribute

Description

The ATDX_BDNAMESP() function returns a pointer to the name of the board device on which the channel accessed by chdev resides.

As illustrated in the example, this may be used to open the board device that corresponds to a particular channel device prior to setting board parameters.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using dx_open().</td>
</tr>
</tbody>
</table>

Cautions

None.

Example

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    int chdev, bddev;
    char *bdnamep;
    /* Open the channel device */
    if ((chdev = dx_open("dxxxB1C1", NULL)) == -1) {
        /* Process error */
    }
}
```
ATDX_BDNAMEP( )        returns a pointer

    /* Display board name */
    bdnamep = ATDX_BDNAMEP(chdev);
    printf(“The board device is: %s
", bdnamep);

    /* Open the board device */
    if ((bddev = dx_open(bdnamep, NULL)) == -1) {
        /* Process error */
        
    }

■ Errors

This function will fail and return a pointer to "Unknown device" if an invalid channel device handle is specified in chdev.
returns the device type

Name: long ATDX_BDTYPE(dev)
Inputs: int dev
• valid Dialogic board or channel device handle
Returns: board or channel device type if successful
AT_FAILURE if error
Includes: srllib.h
dxxlib.h
Category: Extended Attribute

Description

The ATDX_BDTYPE() function returns the device type of the board or channel dev.

A typical use would be to determine whether or not the device can support particular features, such as Call Analysis.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dev:</td>
<td>specifies the valid Dialogic device handle obtained when a board or channel was opened using dx_open().</td>
</tr>
</tbody>
</table>

Possible return values are the following:

DL_D20BD • D/20 Board Device
DL_D21BD • D/21 Board Device
DL_D40BD • D/40 Board Device
DL_D41BD • D/41 Board Device
DL_D20CH • D/20 Channel Device
DL_D21CH • D/21 Channel Device
DL_D40CH • D/40 Channel Device
DL_D41CH • D/41 Channel Device

NOTE: DL_D41BD and DL_D41CH will be returned for the D/121 board, which emulates three D/41 boards. DL_D41BD and DL_D41CH will be returned: for the D/160SC-LS board which emulates four D/41 boards; for the D/240SC and D/240SC-T1 boards which emulate six D/41 boards; and for the D/300SC-E1 and D/320SC boards which emulate eight D/41
ATDX_BDTYPE() returns the device type

boards

■ Cautions

None.

■ Example

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
#define ON 1

main()
{
  int bddev;
  long bdtype;
  int call_analysis=0;

  /* Open the board device */
  if ((bddev = dx_open("dxxxB1",NULL)) == -1) {
    /* Process error */
  }

  if((bdtype = ATDX_BDTYPE(bddev)) == AT_FAILURE) {
    /* Process error */
  }

  if(bdtype == DI_D41BD) {
    printf("Device is a D/41 Board\n");
    call_analysis = ON;
  }

  ...
}
```

■ Errors

This function will fail and return AT_FAILURE if an invalid board or channel device handle is specified in dev.

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returns the number of uncollected digits

**Name:** long ATDX_BUFDIGS(chdev)

**Inputs:** int chdev • valid Dialogic channel device handle

**Returns:** number of uncollected digits in the firmware buffer if successful
            AT_FAILURE if error

**Includes:** srllib.h
dxxxlib.h

**Category:** Extended Attribute

### Description

The **ATDX_BUFDIGS()** function returns the number of uncollected digits in the firmware buffer for channel **chdev**. This is the number of digits that have arrived since the last call to **dx_getdig()** or the last time the buffer was cleared using **dx_clrdigbuf()**. The digit buffer contains a maximum of 31 digits and a null terminator.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <strong>dx_open()</strong>.</td>
</tr>
</tbody>
</table>

### Cautions

Digits that adjust speed and volume (see **dx_setsvcond()**) will not be passed to the digit buffer.

### Example

```c
#include <fcntl.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
  int chdev;
  long bufdigs;
  DX_IOTT iott;
  DV_TPT  tpt[2];
```
ATDX_BUFDIGS( ) returns the number of uncollected digits

/* Open the device using dx_open( ). Get channel device descriptor in */
/* chdev. */
if ((chdev = dx_open("dxxdB1Cl",NULL)) == -1) {
    /* process error */
}

/* set up DX_IOTT */
io_type = IO_DEV|IO_EOT;
iott.io_bufp = 0;
iott.io_offset = 0;
iott.io_length = -1; /* play till end of file */

if((iott.io_fhandle = dx_fileopen("prompt.vox", O_RDONLY)) == -1) {
    /* process error */
}

/* set up DV_TPT */
dx_clrtpt(tpt,2);
tpt[0].tp_type = IO_CONT;
tpt[0].tp_termno = DX_MAXDTMF; /* Maximum digits */
tpt[0].tp_length = 4; /* terminate on 4 digits */
tpt[0].tp_flags = TF_MAXDTMF; /* Use the default flags */
tpt[1].tp_type = IO_EOT;
tpt[1].tp_termno = DX_DIGMASK; /* Digit termination */
tpt[1].tp_length = DM_5; /* terminate on the digit "5" */
tpt[1].tp_flags = TF_DIGMASK; /* Use the default flags */

/* Play a voice file. Terminate on receiving 4 digits, the digit "5" or */
/* at end of file.*/
if (dx_play(chdev,&iott,tpt,EV_SYNC) == -1) {
    /* process error */
}
/* Check # of digits collected and continue processing. */
if((bufdigs = ATDX_BUFDIGS(chdev)) ==AT_FAILURE) {
    /* process error */
}

Errors

This function will fail and return AT_FAILURE if an invalid channel device handle is specified in chdev.

See Also

Other digit functions:
• dx_getdig( )
• dx_clrdigbuf( )

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**ATDX_ChNAMES( )**

<table>
<thead>
<tr>
<th>Name:</th>
<th>char ** ATDX_ChNAMES(bddev)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>int bddev</td>
</tr>
<tr>
<td></td>
<td>• valid Dialogic board device handle</td>
</tr>
<tr>
<td>Returns:</td>
<td>pointer to array of channel names if successful</td>
</tr>
<tr>
<td></td>
<td>pointer to array of pointers that point to &quot;Unknown device&quot; if error</td>
</tr>
<tr>
<td>Includes:</td>
<td>srllib.h</td>
</tr>
<tr>
<td></td>
<td>dxxxlib.h</td>
</tr>
<tr>
<td>Category:</td>
<td>Extended Attribute</td>
</tr>
</tbody>
</table>

### Description

The **ATDX_ChNAMES( )** function returns a pointer to an array of channel names associated with the designated board device handle `bddev`.

A possible use for this attribute would be to display the names of the channel devices associated with a particular board device.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>bddev:</td>
<td>specifies the valid board device handle obtained when the board was opened using <code>dx_open()</code></td>
</tr>
</tbody>
</table>

### Cautions

None.

### Example

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
  int bddev, cnt;
  char **chnames;
  long subdevs;
  .
  /* Open the board device */
  .
} /* Open the board device */
```
if ((bddev = dx_open("dxxxB1", NULL)) == -1) {
   /* Process error */
   .
   .
   /* Display channels on board */
   chnames = ATDX_CHNAMES(bddev);
   subdevs = ATDV_SUBDEVS(bddev);  /* number of sub-devices on board */
   printf("Channels on this board are:\n");
   for (cnt=0; cnt<subdevs; cnt++) {
      printf("%s\n", *(chnames + cnt));
   }
   .
   /* Call dx_open( ) to open each of the
    * channels and store the device descriptors
    */
   .
}

Errors

This function will fail and return the address of a pointer to "Unknown device" if an invalid board device handle is specified in bddev.
Description

The ATDX_CHNUM() function returns the channel number of the channel chdev on the Voice board. Channel numbering starts at 1.

For example, use the channel as an index into an array of channel-specific information.

Cautions

None.

Example

```c
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
maint()
{
    int chdev;
    long chno;

    /* Open the channel device */
    if ((chdev = dx_open("dxxxB1C1", NULL)) == -1) {
        /* Process error */
    }

    /* Get Channel number */
    if((chno = ATDX_CHNUM(chdev)) == AT_FAILURE) {
        /* Process error */
    }
    /* Use chno for application-specific purposes */
```

returns the channel number ATDX_CHNUM()

Name: long ATDX_CHNUM(chdev)  
Inputs: int chdev • valid Dialogic channel device handle  
Returns: channel number if successful  
          AT_FAILURE if error  
Includes: srllib.h  
dxxxlib.h  
Category: Extended Attribute
**ATDX_CHNUM( )**  
returns the channel number

### Errors

This function will fail and return AT_FAILURE if an invalid channel device handle is specified in `chdev`. 
returns the connection

ATDX_CONNTYPE()

Name: long ATDX_CONNTYPE(chdev)
Inputs: int chdev • valid Dialogic channel device handle
Returns: connection type AT_FAILURE if error
Includes: srllib.h dxxxlib.h
Category: Extended Attribute

Description

The ATDX_CONNTYPE() function returns the connection for a call on the channel chdev. Use this function when a CR_CNCT is returned by ATDX_CPTERM() after termination of dx_dial() with Call Analysis enabled.

Possible return values are the following:

- CON_CAD • Connection due to cadence break
- CON_LPC • Connection due to loop current
- CON_PVD • Connection due to Positive Voice Detection
- CON_PAMD • Connection due to Positive Answering Machine Detection

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using dx_open().</td>
</tr>
</tbody>
</table>

Cautions

None.

Example

```
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
```
main()
{
    int dxxxdev;
    int cares;

    /*
     * Open the Voice Channel Device and Enable a Handler
     */
    if ( ( dxxxdev = dx_open( "dxxxB1C1", NULL ) ) == -1 ) {
        perror( "dxxxB1C1" );
        exit( 1 );
    }

    /*
     * Delete any previous tones
     */
    if ( dx_deltones(dxxxdev) < 0 ) {
        /* handle error */
    }

    /*
     * Now enable Enhanced call progress with above changed settings.
     */
    if (dx_initcallp( dxxxdev )) {
        /* handle error */
    }

    /*
     * Take the phone off-hook
     */
    if ( dx_sethook( dxxxdev, DX_OFFHOOK, EV_SYNC ) == -1 ) {
        printf( "Unable to set the phone off-hook\n" );
        printf( "Lasterror = %d  Err Msg = %s\n",
            ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
        dx_close( dxxxdev );
        exit( 1 );
    }

    /*
     * Perform an outbound dial with call analysis, using
     * the default call analysis parameters.
     */
    if ( (cares=dx_dial( dxxxdev, ",84",(DX_CAP *)NULL, DX_CALLP ) ) == -1 ) {
        printf( "Outbound dial failed - reason = %d\n",
            ATDX_CPERROR( dxxxdev ) );
        dx_close( dxxxdev );
        exit( 1 );
    }

    printf( "Call Analysis returned %d\n", cares );
    if ( cares == CR_CNCT ) {
        switch ( ATDX_CONNTYPE( dxxxdev ) ) {
            case CON_CAD:
                printf( "Cadence Break\n" );
                break;
            case CON_LPC:
                printf( "Loop Current Drop\n" );
                break;
            case CON_PVD:
                printf( "Positive Voice Detection\n" );
                break;
        }
    }

    printf( "46-CD\n" );
returns the connection  

```c
ATDX_CONNTYPE()
```

```c
break;

case CON_PAMD:
    printf("Positive Answering Machine Detection\n");
    break;

default:
    printf("Unknown connection type\n");
    break;
}

/*
 * Continue Processing
 * :
 * :
 * /

/*
 * Close the opened Voice Channel Device
 */
if ( dx_close( dxxxdev ) != 0 ) {
    perror("close");
}

/* Terminate the Program */
exit( 0 );
}

### Errors

This function will fail and return AT_FAILURE if an invalid channel device handle is specified in `chdev`.

### See Also

Related to Call Analysis:

- `dx_dial()`
- `ATDX_CPTERM()`
- `DX_CAP` structure (Chapter 4. Voice Data Structures and Device Parameters)
- "Call Analysis" (Voice Features Guide for Windows NT)
**ATDX_CPERROR( )**

*returns the error*

<table>
<thead>
<tr>
<th>Name:</th>
<th>long ATDX_CPERROR(chdev)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>int chdev • valid Dialogic channel device handle</td>
</tr>
<tr>
<td>Returns:</td>
<td>Call Analysis error AT_FAILURE if function fails</td>
</tr>
<tr>
<td>Includes:</td>
<td>srllib.h dxxxlib.h</td>
</tr>
<tr>
<td>Category:</td>
<td>Extended Attribute</td>
</tr>
</tbody>
</table>

### Description

The **ATDX_CPERROR( )** function returns the error that caused **dx_dial( )** to terminate when checking for operator intercept SIT tones.

When **dx_dial( )** terminates due to a Call Analysis error, **CR_ERROR** is returned by **ATDX_CPTERM( )**.

If **CR_ERROR** is returned, use **ATDX_CPERROR( )** to determine the Call Analysis error. One of the following values will be returned:

- **CR_LGTUERR** • lower frequency greater than upper frequency
- **CR_MEMERR** • out of memory when creating temporary SIT tone templates
- **CR_MXFRQERR** • invalid ca_maxtimefreq field in DX_CAP
- **CR_OVRLPERR** • overlap in selected SIT tones
- **CR_TMOUTOFF** • timeout waiting for SIT tone to terminate
- **CR_TMOUTON** • timeout waiting for SIT tone to commence
- **CR_UNEXPTN** • unexpected SIT tone
- **CR_UPFRQERR** • invalid upper frequency selection

The function parameter is defined as follows:
Parameter Description

chdev: specifies the valid channel device handle obtained when the channel was opened using dx_open().

■ Cautions

None.

■ Example

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

int main() {
    int dxxxdev;
    int cares;

    /* Open the Voice Channel Device and Enable a Handler */
    if ( (dxxxdev = dx_open( "dxxxB1C1", NULL) ) == -1 ) {
        perror( "dxxxB1C1" );
        exit( 1 );
    }

    /* Take the phone off-hook */
    if ( dx_sethook( dxxxdev, DX_OFFHOOK, EV_SYNC ) == -1 ) {
        printf( "Unable to set the phone off-hook\n" );
        printf( "Lasterror = %d  Err Msg = %s\n",
                ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
        dx_close( dxxxdev );
        exit( 1 );
    }

    /* Perform an outbound dial with call analysis, using *
     * the default call analysis parameters. */
    if((cares = dx_dial( dxxxdev,"84",(DX_CAP *) NULL, DX_CALLP )) == -1 ) {
        printf( "Outbound dial failed - reason = %d\n",
                ATDX_CPERROR( dxxxdev ) );
        dx_close( dxxxdev );
        exit( 1 );
    }
}
```

returns the error ATDX_CPERROR()
ATDX_CPERROR() returns the error

* Continue Processing
*.
*.
*/

/* Close the opened Voice Channel Device */
if ( dx_close( dxxxdev ) != 0 ) {
    perror( "close" );
}

/* Terminate the Program */
exit( 0 );

See Also

Related to Call Analysis:

- dx_dial()
- ATDX_CPTERM()
- DX_CAP structure (Chapter 4. Voice Data Structures and Device Parameters)
returns last Call Analysis termination  

**ATDX_CPTERM( )**

<table>
<thead>
<tr>
<th>Name:</th>
<th>long ATDX_CPTERM(chdev)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>int chdev</td>
</tr>
<tr>
<td></td>
<td>• valid Dialogic channel device handle</td>
</tr>
<tr>
<td>Returns:</td>
<td>last Call Analysis termination if successful</td>
</tr>
<tr>
<td></td>
<td>AT_FAILURE if error</td>
</tr>
<tr>
<td>Includes:</td>
<td>srllib.h</td>
</tr>
<tr>
<td></td>
<td>dxxxlib.h</td>
</tr>
<tr>
<td>Category:</td>
<td>Extended Attribute</td>
</tr>
</tbody>
</table>

### Description

The **ATDX_CPTERM( )** function returns last Call Analysis termination on the channel **chdev**. Call this function to determine the call status after dialing out with Call Analysis enabled.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <strong>dx_open( )</strong>.</td>
</tr>
</tbody>
</table>

Possible return values are the following:

- CR_BUSY • Busy
- CR_CEPT • Operator intercept
- CR_CNCT • Connect
- CR_FAXTONE • Called line answered by fax machine or modem
- CR_NOANS • No answer
- CR_NODIALTONE • Timeout occurred while waiting for dial tone
- CR_NORB • No ringback
- CR_STOPD • Stopped
- CR_ERROR • Error

### Cautions

None.
ATDX_CPTERM( )

returns last Call Analysis termination

Example

/* Call Analysis with user-specified parameters */
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
  int chdev;
  DX_CAP capp;
  ...
  /* open the channel using dx_open( ). Obtain channel device descriptor in */
  * chdev */
  if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
    /* process error */
  }

  /* take the phone off-hook */
  if (dx_sethook(chdev,DX_OFFHOOK,EV_SYNC) == -1) {
    /* process error */
  } else {
    /* Clear DX_CAP structure */
    dx_clrcap(&capp);

    /* Set the DX_CAP structure as needed for call analysis. */
    /* Allow 3 rings before no answer. */
    capp.ca_nbrdna = 3;

    /* Perform the outbound dial with call analysis enabled. */
    if (dx_dial(chdev,"5551212", &capp,DX_CALLP|EV_SYNC) == -1) {
      /* perform error routine */
    }
  }

  /* Examine last call progress termination on the device */
  switch (ATDX_CPTERM(chdev)) {
    case CR_CNCT: /* Call Connected, get some additional info */
      ....
      break;
    case CR_CEPT: /* Operator Intercept detected */
      ....
      break;
    ....
    case AT_FAILURE: /* Error */
      ....
    }
}
returns last Call Analysis termination

ATDX_CPTERM()

■ Errors

This function will fail and return AT_FAILURE if an invalid channel device handle is specified in chdev.

■ See Also

Related to Call Analysis:

- dx_dial()
- DX_CAP structure (Chapter 4. Voice Data Structures and Device Parameters)
- "Call Analysis" (Voice Features Guide for Windows NT)
**ATDX_CRTNID()**

Name: long ATDX_CRTNID(chdev)

*Inputs:*
- int chdev
  - valid channel device handle

*Returns:* identifier of the tone that caused the most recent Call Analysis termination, if successful (null if failure)

*Includes:* srllib.h
dxxxlib.h

*Category:* Extended Attribute

---

### Description

The **ATDX_CRTNID()** function returns the tone identifier of the tone that caused the most recent Call Analysis termination of the channel device. This function is supported under PerfectCall Call Analysis on DSP boards only. See the **Voice Features Guide for Windows NT** for a description of PerfectCall Call Analysis.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid Dialogic device handle obtained when a board or channel was opened using dx_open().</td>
</tr>
</tbody>
</table>

Possible return values are the following:

- TID_DIAL_LCL • Local dial tone
- TID_DIAL_INTL • International dial tone
- TID_DIAL_XTRA • Special (“Extra”) dial tone
- TID_BUSY1 • First signal busy
- TID_BUSY2 • Second signal busy
- TID_RINGBK1 • Ringback
- TID_FAX1 • First fax or modem tone
- TID_FAX2 • Second fax or modem tone

### Cautions

None.

---

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returns the tone identifier ATDX_CRTNID()

Example

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main() {
    DX_CAP cap_s;
    int ddd, car;
    char *chnam, *dialstrg;

    chnam = "dxxxB1C1";
    dialstrg = "L1234";

    /*
     * Open channel
     */
    if ((ddd = dx_open( chnam, NULL )) == -1 ) {
        /* handle error */
    }

    /*
     * Delete any previous tones
     */
    if ( dx_deltones(ddd) < 0 ) {
        /* handle error */
    }

    /*
     * Now enable Enhanced call progress with above changed settings.
     */
    if (dx_initcallp( ddd )) {
        /* handle error */
    }

    /*
     * Set off Hook
     */
    if ((dx_sethook( ddd, DX_OFFHOOK, EV_SYNC )) == -1) {
        /* handle error */
    }

    /*
     * Dial
     */
    printf("Dialing %s\n", dialstrg);
    car = dx_dial(ddd,dialstrg,(DX_CAP *)&cap_s,DX_CALLP|EV_SYNC);
    if (car == -1) {
        /* handle error */
    }

    switch( car ) {
    case CR_NODIALTONE:
        switch ( ATDX_DTNFAIL(ddd) ) {
            case 'L':
                printf(" Unable to get Local dial tone\n");
        }
    ```
ATDX_CRTNID() returns the tone identifier

break;
case 'I':
    printf("Unable to get International dial tone\n");
    break;
case 'X':
    printf("Unable to get special eXtra dial tone\n");
    break;
}
break;

case CR_BUSY:
    printf("%s engaged - %s detected\n", dialstrg,
        ATDX_CRTNID(ddd) == TID_BUSY1 ? "Busy 1" : "Busy 2" ) );
    break;

case CR_CNCT:
    printf("Successful connection to %s\n", dialstrg );
    break;

default:
    break;
}

/*
 * Set on Hook
*/
if ((dx_sethook( ddd, DX_ONHOOK, EV_SYNC )) == -1) { /* handle error */
}

dx_close( ddd );
}
**Name:**
long ATDX_DEVTYPE(dev)

**Inputs:**
int dev  
• valid Dialogic board or channel device handle

**Returns:**
device type if successful
AT_FAILURE if error

**Includes:**
srllib.h
dxxxlib.h

**Category:**
Extended Attribute

---

**Description**

The `ATDX_DEVTYPE()` function returns device type of the board or channel `dev`.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
</table>
| dev:      | specifies the valid Dialogic device handle obtained when a board or channel was opened using `dx_open()`.

Possible return values are the following:

- DT_DXBD  • Board device
- DT_DXCH  • Channel

**Cautions**

None.

**Example**

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    int bddev;
```
ATDX_DEVTYPE() returns device type

long devtype;

/* Open the board device */
if ((bddev = dx_open("dxxxb1",NULL)) == -1) {
    /* Process error */
}

if ((devtype = ATDX_DEVTYPE (bddev)) == AT_FAILURE) {
    /* Process error */
}

if (devtype == DT_XXBD) {
    printf("Device is a Board\n");
}

/* Continue processing */
/* Continue processing */
/* Continue processing */

■ Errors

This function will fail and return AT_FAILURE if an invalid board or channel device handle is specified in dev.
returns character for dial tone

ATDX_DTNFAIL()

Name: long ATDX_DTNFAIL(chdev)

Inputs: int chdev • valid channel device handle

Returns: code for the dial tone that failed to appear
AT_FAILURE if error

Includes: srllib.h
dxxxlib.h

Category: Extended Attribute

■ Description

The ATDX_DTNFAIL() function returns character for dial tone that PerfectCall Call Analysis failed to detect. This attribute is supported under PerfectCall Call Analysis only.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid Dialogic device handle obtained when a board or channel was opened using dx_open().</td>
</tr>
</tbody>
</table>

Possible return values are the following:

- L • Local dial tone
- I • International dial tone
- X • Special ("extra") dial tone

■ Cautions

None.

■ Example

```c
#include <stdio.h>

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

main()
```
ATDX_DTNFAIL( )
returns character for dial tone

{  
   DX_CAP cap_s;
   int ddd, car;
   char *chnam, *dialstrg;

   chnam = "dxxxB1C1";
   dialstrg = "L1234";

   /*
   *  Open channel
   */
   if ((ddd = dx_open( chnam, NULL )) == -1 ) {
      /* handle error */
   }

   /*
   * Delete any previous tones
   */
   if ( dx_deltones(ddd) < 0 ) {
      /* handle error */
   }

   /*
   * Now enable Enhanced call progress with above changed settings.
   */
   if (dx_initcallp( ddd )) {
      /* handle error */
   }

   /*
   * Set off Hook
   */
   if ( dx_sethook( ddd, DX_OFFHOOK, EV_SYNC )) == -1) { 
      /* handle error */
   }

   /*
   * Dial
   */
   printf("Dialing \n", dialstrg );
   car = dx_dial(ddd,dialstrg,(DX_CAP *)&cap_s,DX_CALLP|EV_SYNC);
   if (car == -1) {
      /* handle error */
   }

   switch( car ) {  
   case CR_NODIALTONE:  
      switch( ATDX_DTNFAIL(ddd) ) {  
         case 'L':
            printf(" Unable to get Local dial tone\n");
            break;
         case 'I':
            printf(" Unable to get International dial tone\n");
            break;
         case 'X':
            printf(" Unable to get special eXtra dial tone\n");
            break;
      }
      break;

   }
returns character for dial tone

ATDX_DTNFAIL()

case CR_BUSY:
    printf("%s engaged - %s detected\n", dialstrg,
        ATDX_CRTNID(ddd) == TID_BUSY1 ? "Busy 1" : "Busy 2") ;
    break;

case CR_CNCT:
    printf("Successful connection to %s\n", dialstrg); 
    break;

default:
    break;
}

/*@ 
 * Set on Hook 
 */
if ((dx_sethook( ddd, DX_ONHOOK, EV_SYNC )) == -1) { 
    /* handle error */
}

dx_close( ddd );
}
**ATDX_FRQDUR( )** can be used to return the duration

<table>
<thead>
<tr>
<th>Name:</th>
<th>long ATDX_FRQDUR(chdev)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>int chdev • valid Dialogic channel device handle</td>
</tr>
<tr>
<td>Returns:</td>
<td>first frequency duration in 10ms units AT_FAILURE if error</td>
</tr>
<tr>
<td>Includes:</td>
<td>srllib.h dxxlib.h</td>
</tr>
<tr>
<td>Category:</td>
<td>Extended Attribute</td>
</tr>
</tbody>
</table>

### Description

When `dx_dial()` terminates due to Operator Intercept; the `ATDX_FRQDUR()` function can be used to return the duration of the first detected SIT tone frequency in 10 ms units.

Termination due to Operator Intercept is indicated by `ATDX_CPTERM()` returning CR_CEPT.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <code>dx_open()</code>.</td>
</tr>
</tbody>
</table>

### Cautions

None.

### Example

This example illustrates `ATDX_FRQDUR()`, `ATDX_FRQDUR2()`, and `ATDX_FRQDUR3()`.

```c
/* Call Analysis with user-specified parameters */
#include <stdio.h>
#include <srllib.h>
#include <dxxlib.h>
#include <windows.h>

main()
```

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can be used to return the duration $ATDX_FRQDUR()$.

```c
int cares, chdev;
DX_CAP capp;

/* open the channel using dx_open(). Obtain channel device descriptor in
 * chdev */
if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
    /* process error */
}

/* take the phone off-hook */
if (dx_sethook(chdev,DX_OFFHOOK,EV_SYNC) == -1) {
    /* process error */
}

/* Set the DX_CAP structure as needed for call analysis. Perform the
 * outbound dial with call analysis enabled */
if ((cares = dx_dial(chdev,"5551212",&capp,DX_CALLP|EV_SYNC)) == -1) {
    /* perform error routine */
}

switch (cares) {
    case CR_CNCT: /* Call Connected, get some additional info */
        printf("\nDuration of short low - %ld ms, ATDX_SHORTLOW(chdev)\*10;
        printf("\nDuration of long low - %ld ms, ATDX_LONGLOW(chdev)\*10;
        printf("\nDuration of answer - %ld ms, ATDX_ANSRSIZ(chdev)\*10;
        break;
    case CR_CEPT: /* Operator Intercept detected */
        printf("\nFirst frequency detected - %ld Hz, ATDX_FRQHZ(chdev);
        printf("\nSecond frequency detected - %ld Hz, ATDX_FRQHZ2(chdev);
        printf("\nThird frequency detected - %ld Hz, ATDX_FRQHZ3(chdev);
        printf("\nDuration of first frequency - %ld ms, ATDX_FRQDUR(chdev));
        printf("\nDuration of second frequency - %ld ms, ATDX_FRQDUR2(chdev));
        printf("\nDuration of third frequency - %ld ms, ATDX_FRQDUR3(chdev));
        break;
    case CR_BUSY:
        break;
    .
    .
}
```

### See Also

#### Related to Call Analysis:

- `dx_dial()`
- `ATDX_CPTERM()`
- `DX_CAP` structure (Chapter 4. Voice Data Structures and Device Parameters)
- "Call Analysis" (Voice Features Guide for Windows NT)
ATDX_FRQDUR( ) can be used to return the duration

SIT Tone Detection - All Boards:
- ATDX_FRQHZ( )

SIT Tone Detection - DSP Boards
- ATDX_FRQDUR2( )
- ATDX_FRQDUR3( )
- ATDX_FRQHZ2( )
- ATDX_FRQHZ3( )
Name: long ATDX_FRQDUR2(chdev)

Inputs: int chdev • valid Dialogic channel device handle

Returns: second frequency duration in 10 ms units
AT_FAILURE if error

Includes: srllib.h
dxxlib.h

Category: Extended Attribute

Description

When dx_dial() terminates due to Operator Intercept, the ATDX_FRQDUR2() function can be used to return the duration of the second detected frequency in 10 ms units.

NOTE: For more information on tri-tone SIT sequences, see Frequency Detection in the Voice Features Guide for Windows NT.

Termination due to Operator Intercept is indicated by ATDX_CPTERM() returning CR_CEPT.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using dx_open().</td>
</tr>
</tbody>
</table>

Cautions

None.

Example

See the example for ATDX_FRQDUR().

See Also

Related to Call Analysis:
ATDX_FRQDUR2( ) can be used to return the duration

- dx_dial()
- ATDX_CPTERM()
- DX_CAP structure (Chapter 4. Voice Data Structures and Device Parameters)
- "Call Analysis" (Voice Features Guide for Windows NT)

SIT Tone Detection - All Boards:

- ATDX_FRQHZ()
- ATDX_FRQDUR()
- "Frequency Detection" (Voice Features Guide for Windows NT)

SIT Tone Detection - DSP Boards:

- ATDX_FRQDUR3()
- ATDX_FRQHZ2()
- ATDX_FRQHZ3()
can be used to return the duration

ATDX_FRQDUR3()

<table>
<thead>
<tr>
<th>Name:</th>
<th>long ATDX_FRQDUR3(chdev)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>int chdev • valid Dialogic channel device handle</td>
</tr>
<tr>
<td>Returns:</td>
<td>third frequency duration in 10 ms units AT_FAILURE if error</td>
</tr>
<tr>
<td>Includes:</td>
<td>srlib.h dlxxxlib.h</td>
</tr>
<tr>
<td>Category:</td>
<td>Extended Attribute</td>
</tr>
</tbody>
</table>

■ Description

When dx_dial() terminates due to Operator Intercept, the ATDX_FRQDUR3() function can be used to return the duration of the third detected frequency in 10 ms units.

**NOTE:** For more information about tri-tone SIT tone detection, see Frequency Detection in the Voice Features Guide for Windows NT.

Termination due to Operator Intercept is indicated by ATDX_CPTERM() returning CR_CEPT.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using dx_open().</td>
</tr>
</tbody>
</table>

■ Cautions

None.

■ Example

See the example for ATDX_FRQDUR().
See Also

Related to Call Analysis:

- dx_dial()
- ATDX_CPTERM()
- DX_CAP structure (Chapter 4. Voice Data Structures and Device Parameters)
- "Call Analysis" (Voice Features Guide for Windows NT)

SIT Tone Detection - All Boards:

- ATDX_FRQHZ()
- ATDX_FRQDUR()
- "Frequency Detection" (Voice Features Guide for Windows NT)

SIT Tone Detection - DSP Boards

- ATDX_FRQDUR2()
- ATDX_FRQHZ2()
- ATDX_FRQHZ3()
**ATDX_FRQHZ()**

**Name:** long ATDX_FRQHZ(chdev)

**Inputs:** int chdev • valid Dialogic channel device handle

**Returns:** first tone frequency in hz

AT_FAILURE if error

**Includes:** srllib.h
dxxxlib.h

**Category:** Extended Attribute

### Description

When `dx_dial()` with Call Analysis terminates due to Operator Intercept, the `ATDX_FRQHZ()` function can return frequency of answered signal in Hz, (such as the first detected SIT tone that occurs due to operator intercept).

**NOTE:** Termination due to Operator Intercept is indicated by `ATDX_CPTERM()` returning CR_CEPT.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
</table>
| chdev:    | specifies the valid channel device handle obtained when the channel was opened using `dx_open()`.

### Cautions

None.

### Example

This example illustrates the use of `ATDX_FRQHZ()`, `ATDX_FRQHZ2()` , and `ATDX_FRQHZ3()`.

```c
/* Call Analysis with user-specified parameters */
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    int cares, chdev;
}```
ATDX_FRQHZ()  return frequency of answered signal

DX_CAP capp;

/* open the channel using dx_open(). Obtain channel device descriptor in
 * chdev */
if ((chdev = dx_open("dxxxB1C1",NULL)) == -1)  {
    /* process error */
}

/* take the phone off-hook */
if (dx_sethook(chdev,DX_OFFHOOK,EV_SYNC) == -1) {
    /* process error */
}

/* Set the DX_CAP structure as needed for call analysis. Perform the
 * outbound dial with call analysis enabled */
if ((cares = dx_dial(chdev,"5551212",&capp,DX_CALLP|EV_SYNC)) == -1) {
    /* perform error routine */
}

switch (cares) {
    case CR_CNCT: /* Call Connected, get some additional info */
        printf("\nDuration of short low - %ld ms",ATDX_SHORTLOW(chdev)*10);
        printf("\nDuration of long low  - %ld ms",ATDX_LONGLOW(chdev)*10);
        printf("\nDuration of answer    - %ld ms",ATDX_ANSRSIZ(chdev)*10);
        break;
    case CR_CEPT: /* Operator Intercept detected */
        printf("\nFirst frequency detected - %ld Hz", ATDX_FRQHZ(chdev));
        printf("\nSecond frequency detected - %ld Hz", ATDX_FRQHZ2(chdev));
        printf("\nThird frequency detected - %ld Hz", ATDX_FRQHZ3(chdev));
        printf("\nDuration of first frequency - %ld ms", ATDX_FRQDUR(chdev));
        printf("\nDuration of second frequency - %ld ms", ATDX_FRQDUR2(chdev));
        printf("\nDuration of third frequency - %ld ms", ATDX_FRQDUR3(chdev));
        break;
    case CR_BUSY:
        break;
    .
}

See Also

Related to Call Analysis:

- dx_dial()
- ATDX_CPTERM()
- DX_CAP structure (Chapter 4. Voice Data Structures and Device Parameters)
- "Call Analysis" (Voice Features Guide for Windows NT)
return frequency of answered signal

ATDX_FRQHZ( )

SIT Tone Detection - All Boards:

• ATDX_FRQDUR( )

SIT Tone Detection - DSP Boards:

• ATDX_FRQDUR2( )
• ATDX_FRQDUR3( )
• ATDX_FRQHZ2( )
• ATDX_FRQHZ3( )
**Name:** long ATDX_FRQHZ2(chdev)

**Inputs:**
- int chdev
  - valid Dialogic channel device handle

**Returns:**
- second tone frequency in hz
- AT_FAILURE if error

**Includes:**
- srllib.h
- dxxxlib.h

**Category:** Extended Attribute

### Description

When `dx_dial()` terminates due to Operator Intercept, the `ATDX_FRQHZ2()` function can return frequency of second detected tone in Hz.

Termination due to Operator Intercept is indicated by `ATDX_CPTERM()` returning CR_CEPT.

**NOTE:** For more information about tri-tone SIT tone detection, see *Frequency Detection* in the *Voice Features Guide for Windows NT*.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
</table>
| chdev:    | specifies the valid channel device handle obtained when the channel was opened using `dx_open()`.

### Cautions

None.

### Example

See the example for `ATDX_FRQHZ()`.
return frequency of second detected tone \texttt{ATDX\_FRQHZ2()} 

\begin{itemize}
\item See Also

Related to Call Analysis:
\begin{itemize}
\item \texttt{dx\_dial()}
\item \texttt{ATDX\_CPTERM()}
\item DX\_CAP structure (Chapter 4. \textit{Voice Data Structures and Device Parameters})
\item "Call Analysis" (\textit{Voice Features Guide for Windows NT})
\end{itemize}

SIT Tone Detection - All Boards:
\begin{itemize}
\item \texttt{ATDX\_FRQDUR()} 
\item \texttt{ATDX\_FRQHZ()} 
\item "Frequency Detection" (\textit{Voice Features Guide for Windows NT})
\end{itemize}

SIT Tone Detection - DSP Boards:
\begin{itemize}
\item \texttt{ATDX\_FRQDUR2()} 
\item \texttt{ATDX\_FRQDUR3()}
\item \texttt{ATDX\_FRQHZ3()}
\end{itemize}
**ATDX_FRQHZ3( )**

*return frequency of third detected tone*

<table>
<thead>
<tr>
<th>Name:</th>
<th>long ATDX_FRQHZ3(chdev)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>int chdev</td>
</tr>
<tr>
<td>Returns:</td>
<td>third tone frequency in hz</td>
</tr>
<tr>
<td>Includes:</td>
<td>srllib.h</td>
</tr>
<tr>
<td>Category:</td>
<td>Extended Attribute</td>
</tr>
</tbody>
</table>

### Description

When `dx_dial()` terminates due to Operator Intercept, the `ATDX_FRQHZ3()` function can return frequency of third detected tone in Hz.

Termination due to Operator Intercept is indicated by `ATDX_CPTERM()` returning CR_CEPT.

**NOTE:** For more information about tri-tone SIT tone detection, see *Frequency Detection* in the *Voice Features Guide for Windows NT*.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
</table>
| chdev:    | specifies the valid channel device handle obtained when the channel was opened using `dx_open()`.

### Cautions

None.

### Example

See the example for `ATDX_FRQHZ()`.
return frequency of third detected tone \texttt{ATDX\_FRQHZ3()}

- See Also

Related to Call Analysis:

- \texttt{dx\_dial()}
- \texttt{ATDX\_CPTERM()}
- \texttt{DX\_CAP} structure (Chapter 4, Voice Data Structures and Device Parameters)
- "Call Analysis" (Voice Features Guide for Windows NT)

SIT Tone Detection - DSP Boards

- \texttt{ATDX\_FRQDUR2()}
- \texttt{ATDX\_FRQDUR3()}
- \texttt{ATDX\_FRQHZ2()}

75-CD
ATDX_FRQOUT() returns percentage of a single tone frequency

**Name:** long ATDX_FRQOUT(chdev)

**Inputs:**
- int chdev
- valid Dialogic channel device handle

**Returns:**
- percentage frequency out-of bounds
- AT_FAILURE if error

**Includes:**
- srllib.h
- dxxxlib.h

**Category:** Extended Attribute

### Description

The ATDX_FRQOUT() function returns percentage of a single tone frequency that was not within the specified range in the DX_CAP structure.

Upon detection of a frequency within the range specified by `ca_upperfrq` and lower `ca_lowerfrq`, use this function to optimize the `ca_refctfrq` parameter (which sets the percentage of *time* that the frequency can be out of bounds) in the DX_CAP structure.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <code>dx_open()</code></td>
</tr>
</tbody>
</table>

### Cautions

This function is only for use with non-DSP boards. If you call it on a DSP board, it will return zero.

### Example

```c
/* Call Analysis with user-specified parameters */
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    int cares, chdev;
}
```

76-CD
returns percentage of a single tone frequency

\[ \text{ATDX\_FRQOUT() } \]

```c
DX\_CAP capp;

/* open the channel using dx\_open( ).  Obtain channel device descriptor in chdev */
if ((chdev = dx\_open("dxxxB1C1",NULL)) == -1) {
    /* process error */
}

/* take the phone off-hook */
if (dx\_sethook(chdev,DX\_OFFHOOK,EV\_SYNC) == -1) {
    /* process error */
}

/* Set the DX\_CAP structure as needed for call analysis. Perform the * outbound dial with call analysis enabled. */
if ((cares = dx\_dial(chdev,"5551212",&capp,DX\_CALLP|EV\_SYNC)) == -1) {
    /* perform error routine */
}

switch (cares) {
    case CR\_CNCT: /* Call Connected, get some additional info */
        printf("\n\nDuration of short low - %ld ms\n",ATDX\_SHORTLOW(chdev)*10);
        printf("\nDuration of long low  - %ld ms\n",ATDX\_LONGLOW(chdev)*10);
        printf("\nDuration of answer    - %ld ms\n",ATDX\_ANSRSIZ(chdev)*10);
        break;
    case CR\_CEPT: /* Operator Intercept detected */
        printf("\nFrequency detected - %ld Hz\n",ATDX\_FRQHZ(chdev));
        printf("\n%% of Frequency out of bounds - %ld Hz\n",ATDX\_FRQOUT(chdev));
        break;
    case CR\_BUSY:
        break;
    
    }
```

### Errors

This function will fail and return AT\_FAILURE if an invalid channel device handle is specified in \texttt{chdev}.

### See Also

Related to Call Analysis:

- \texttt{dx\_dial()}
- \texttt{ATDX\_CPTERM()}
- DX\_CAP structure (Chapter 4. Voice Data Structures and Device Parameters)
- "Call Analysis" (Voice Features Guide for Windows NT)
# ATDX_FWVER()

Name: long ATDX_FWVER(bddev)

Inputs: int bddev
  • valid Dialogic board device handle

Returns: D/4x Firmware version if successful
          AT_FAILURE if error

Includes: srllib.h
dxxxlib.h

Category: Extended Attribute

## Description

The ATDX_FWVER() function returns version number of D/4x firmware. On a D/41ESC or a D/xxxSC board the emulated D/4x firmware version is returned.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>bddev:</td>
<td>specifies the valid board device handle obtained when the board was opened using dx_open().</td>
</tr>
</tbody>
</table>

## Cautions

None.

## Example

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
  int bddev;
  long fwver;

  /* Open the board device */
  if ((bddev = dx_open("dxxxB1",NULL)) == -1) {
    /* Process error */
  }

  /* Display Firmware version number */
  if ((fwver = ATDX_FWVER(bddev)) == AT_FAILURE) { 
```

78-CD
returns version number of D/4x firmware

`ATDX_FWVER()`

```c
    /* Process error */
    printf("Firmware version %ld\n",fwver);
```

- **Errors**

  This function will fail and return AT_FAILURE if an invalid device handle is specified in `bddev`. 
ATDX_HOOKST()  returns the current hook state

Name:     long ATDX_HOOKST(chdev)
Inputs:   int chdev           • valid Dialogic channel device handle
Returns:  current hook state of channel if successful
          AT_FAILURE if error
Includes: srllib.h
          dxxxlib.h
Category: Extended Attribute

Description

The ATDX_HOOKST() function returns the current hook state of the channel chdev.

NOTE: Do not call this function for a digital T-1 or E-1 SCbus configuration that includes a D/240SC, D/240SC-T1, D/320SC D/300SC-E1, DTI/241SC, or DTI/301SC board. Transparent signaling for SCbus digital interface devices is not supported in System Release 4.1SC.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using dx_open().</td>
</tr>
</tbody>
</table>

Possible return values are the following:

- DX_OFFHOOK • Channel is off-hook
- DX_ONHOOK  • Channel is on-hook

Cautions

None.

Example

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

80-CD
returns the current hook state

```c
ATDX_HOOKST()
```

```c
main()

int chdev;
long hookst;

/* Open the channel device */
if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
    /* Process error */
}

/* Examine Hook state of the channel. Perform application specific action */
if ((hookst = ATDX_HOOKST(chdev)) == AT_FAILURE) {
    /* Process error */
}

if (hookst == DX_OFFHOOK) {
    /* Channel is Off-hook */
}
```

**Errors**

This function will fail and return AT_FAILURE if an invalid channel device handle is specified in `chdev`.

**See Also**

- `dx_sethook()`
- `DX_CST()` (*Chapter 4. Voice Data Structures and Device Parameters*)
**ATDX_LINEST()** returns a bitmapped representation of activity

<table>
<thead>
<tr>
<th>Name:</th>
<th>long ATDX_LINEST(chdev)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>int chdev</td>
</tr>
<tr>
<td></td>
<td>• valid Dialogic channel device handle</td>
</tr>
<tr>
<td>Returns:</td>
<td>current line status of channel if successful</td>
</tr>
<tr>
<td></td>
<td>AT_FAILURE if error</td>
</tr>
<tr>
<td>Includes:</td>
<td>srllib.h</td>
</tr>
<tr>
<td></td>
<td>dxxxlib.h</td>
</tr>
<tr>
<td>Category:</td>
<td>Extended Attribute</td>
</tr>
</tbody>
</table>

### Description

The **ATDX_LINEST( )** function returns a bitmapped representation of activity on the line at that instant connected to the channel `chdev`.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>chdev</code>:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <strong>dx_open()</strong>.</td>
</tr>
</tbody>
</table>

Possible return values are the following:

- RLS_SILENCE • Silence on the line
- RLS_DTMF • present
- RLS_LCSENSE • not present
- RLS_RING • Ring not present
- RLS_HOOK • Channel is on-hook
- RLS_RINGBK • Audible ringback detected

### Cautions

None.

### Example

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

main()
```

82-CD
returns a bitmapped representation of activity

ATDX_LINEST()

```c
int chdev;
long linest;

/* Open the channel device */
if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
    /* Process error */
}

/* Examine line status bitmap of the channel. Perform application-specific action */
if((linest = ATDX_LINEST(chdev)) == AT_FAILURE) {
    /* Process error */
}

if(linest & RLS_LCSENSE) {
    /* No loop current */
}
```

### Errors

This function will fail and return AT_FAILURE if an invalid channel device handle is specified in `chdev`. 

83-CD
The **ATDX_LONGLOW( )** function returns duration of the longer silence, in 10 ms units, of the initial signal that occurred during Call Analysis on the channel `chdev`. This function can be used in conjunction with **ATDX_SIZEHI( )** and **ATDX_SHORTLOW( )** to determine the elements of an established cadence. See the *Voice Features Guide for Windows NT* for further information.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>chdev</code></td>
<td>specifies the valid channel device handle obtained when the channel was opened using <code>dx_open( )</code>.</td>
</tr>
</tbody>
</table>

### Cautions

None.

### Example

```c
/* Call Analysis with user-specified parameters */
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
  int cares, chdev;
  DX_CAP capp;
  
  /* open the channel using `dx_open()`.
  Obtain channel device descriptor in... */
```
returns duration of the longer silence \texttt{ATDX\_LONGLOW( )}

```c
  */
  if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {  /* process error */
  }
  /* take the phone off-hook */
  if (dx_sethook(chdev,DX\_OFFHOOK,EV\_SYNC) == -1) {  /* process error */
  }
  /* Set the DX\_CAP structure as needed for call analysis. Perform the */
  /* outbound dial with call analysis enabled */
  if ((cares = dx_dial(chdev,"5551212",&capp,DX\_CALLP|EV\_SYNC)) == -1) {  /* perform error routine */
    switch (cares) {
      case CR\_CNCT: /* Call Connected, get some additional info */
        printf("Duration of short low - %ld ms\n",ATDX\_SHORTLOW(chdev)*10);
        printf("Duration of long low  - %ld ms\n",ATDX\_LONGLOW(chdev)*10);
        printf("Duration of answer    - %ld ms\n",ATDX\_ANSRSIZ(chdev)*10);
        break;
      case CR\_CEPT: /* Operator Intercept detected */
        printf("Frequency detected - %ld Hz\n",ATDX\_FRQHZ(chdev));
        printf("\% of Frequency out of bounds - %ld Hz\n",ATDX\_FRQOUT(chdev));
        break;
      case CR\_BUSY:
        "
    }

  Errors

  This function will fail and return AT\_FAILURE if an invalid channel device handle is specified in \texttt{chdev}.

  See Also

  Related to Call Analysis::
  \begin{itemize}
    \item \texttt{dx\_dial( )}
    \item \texttt{ATDX\_CPTERM( )}
    \item \texttt{DX\_CAP structure (Chapter 4. Voice Data Structures and Device Parameters)}
    \item "Call Analysis" (Voice Features Guide for Windows NT)
    \item "Cadence Detection" (Voice Features Guide for Windows NT)
  \end{itemize}
ATDX_PHYADDR() returns the physical address

Name: long ATDX_PHYADDR(bddev)
Inputs: int bddev • valid Dialogic board device handle
Returns: physical address of board if successful
          AT_FAILURE if error
Includes: srllib.h
dxxxlib.h
Category: Extended Attribute

Description

The ATDX_PHYADDR() function returns the physical address of the board bddev.

The function parameter is defined as follows:

Parameter Description
bddev: specifies the valid board device handle obtained when the board was opened using dx_open().

Cautions

None.

Example

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    int bddev;
    long phyaddr;

    /* Open the board device */
    if ((bddev = dx_open("dxxxB1",NULL)) == -1) {
        /* Process error */
    }

    if((phyaddr = ATDX_PHYADDR(bddev)) == AT_FAILURE) {
        /* Process error */
    }
```

86-CD
returns the physical address \texttt{ATDX_PHYADDR()} \%

```c
}

printf("Board is at address \%X\n",phyaddr);
}
```

\section*{Errors}

This function will fail and return \texttt{AT\_FAILURE} if an invalid board device handle is specified in \texttt{bddev}.  

\hfill

\textbf{87-CD}
**ATDX_SHORTLOW( )**  
returns duration of shorter silence

| Name:          | long ATDX_SHORTLOW(chdev) |
| Inputs:        | int chdev                 |
|               | • valid Dialogic channel device handle |
| Returns:       | duration of shorter silence if successful |
|               | AT_FAILURE if error     |
| Includes:      | srlib.h                  |
|               | dxxlib.h                 |
| Category:      | Extended Attribute       |

### Description

The **ATDX_SHORTLOW( )** function returns duration of shorter silence of the initial signal that occurred during Call Analysis on the channel **chdev**. This function can be used in conjunction with **ATDX_SIZEHI( )** and **ATDX_LONGLOW( )** to determine the elements of an established cadence. See the *Voice Features Guide for Windows NT* for further information.

Compare the results of this function with the DX_CAP field **ca_lo2rmin** to determine whether the cadence is a double or single ring.

If the result of **ATDX_SHORTLOW( )** is less than the **ca_lo2rmin** field this indicates a double ring cadence.

If the result of **ATDX_SHORTLOW( )** is greater than the **ca_lo2rmin** field this indicates a single ring.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <strong>dx_open( )</strong>.</td>
</tr>
</tbody>
</table>

### Cautions

None.

88-CD
returns duration of shorter silence  

ATDX_SHORTLOW( )

**Example**

```c
/* Call Analysis with user-specified parameters */
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    int cares, chdev;
    DX_CAP capp;
    .
    /* open the channel using dx_open( ). Obtain channel device descriptor
     * in chdev */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
        /* process error */
    }
    /* take the phone off-hook */
    if (dx_sethook(chdev,DX_OFFHOOK,EV_SYNC) == -1) {
        /* process error */
    }
    /* Set the DX_CAP structure as needed for call analysis. Perform the
     * outbound dial with call analysis enabled */
    if ((cares = dx_dial(chdev,"5551212",&capp,DX_CALLP|EV_SYNC)) == -1) {
        /* perform error routine */
    }
    switch (cares) {
    case CR_CNCT: /* Call Connected, get some additional info */
        printf("Duration of short low - \ld ms\n",ATDX_SHORTLOW(chdev)*10);
        printf("Duration of answer - \ld ms\n",ATDX_ANSRSIZ(chdev)*10);
        break;
    case CR_CEPT:     /* Operator Intercept detected */
        printf("\nFrequency detected - \ld Hz\n",ATDX_FRQHZ(chdev));
        printf("\n%% of Frequency out of bounds - \ld Hz\n",ATDX_FRQOUT(chdev));
        break;
    case CR_BUSY:
        .
    }
}
```

**Errors**

This function will fail and return AT_FAILURE if an invalid channel device handle is specified in `chdev`. 
ATDX_SHORTLOW() returns duration of shorter silence

- See Also
  - dx_dial()
  - ATDX_LONGLOW()
  - ATDX_SIZEHI()
  - ATDX_CPTERM()
  - DX_CAP structure (Chapter 4. Voice Data Structures and Device Parameters)
  - "Call Analysis" (Voice Features Guide for Windows NT)
  - "Cadence Detection" (Voice Features Guide for Windows NT)
returns duration of initial non-silence

*ATDX_SIZEHI*( )

**Name:** long ATDX_SIZEHI(chdev)

**Inputs:**
- int chdev
- valid Dialogic channel device handle

**Returns:**
- non-silence duration in 10 ms units if successful
- AT_FAILURE if error

**Includes:**
- srllib.h
- dxxxlib.h

**Category:** Extended Attribute

### Description

The *ATDX_SIZEHI*( ) function returns duration of initial non-silence, in 10 ms units, during Call Analysis on the channel `chdev`. This function can be used in conjunction with *ATDX_SIZEHI*( ) and *ATDX_LONGLOW*( ) to determine the elements of an established cadence. See the Voice Features Guide for Windows NT for further information.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>chdev</code></td>
<td>specifies the valid channel device handle obtained when the channel was opened using <code>dx_open()</code>.</td>
</tr>
</tbody>
</table>

### Cautions

None.

### Example

```c
/* Call Analysis with user-specified parameters */
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()

    {
        int cares, chdev;
        DX_CAP capp;
        ...
        /* open the channel using dx_open(). Obtain channel device descriptor */
    }
```

91-CD
ATDX_SIZEHI( )

returns duration of initial non-silence

* in chdev

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

/* take the phone off-hook */
if (dx_sethook(chdev,DX_OFFHOOK, EV_SYNC) == -1) {
    /* process error */
}

/* Set the DX_CAP structure as needed for call analysis. Perform the
 * outbound dial with call analysis enabled */
if ((cares = dx_dial(chdev,"5551212",&capp,DX_CALLP|EV_SYNC)) == -1) {
    /* perform error routine */
} switch (cares) {
    case CR_CNCT:     /* Call Connected, get some additional info */
        printf("\nDuration of short low - %ld ms",ATDX_SHORTLOW(chdev)*10);
        print("\nDuration of long low - %ld ms",ATDX_LONGLOW(chdev)*10);
        printf("\nDuration of non-silence - %ld ms",ATDX_SIZEHI(chdev)*10);
        break;
    case CR_CEPT:     /* Operator Intercept detected */
        printf("\nFrequency detected - %ld Hz",ATDX_FRQHZ(chdev));
        printf("\n%% of Frequency out of bounds - %ld Hz",ATDX_FRQOUT(chdev));
        break;
    case CR_BUSY:    /* */
        .
    .
    .
}

Errors

This function will fail and return AT_FAILURE if an invalid channel device
handle is specified in chdev.

See Also

- dx_dial()
- ATDX_LONGLOW()
- ATDX_SHORTLOW()
- ATDX_CPTERM()
- DX_CAP structure (Chapter 4. Voice Data Structures and Device
  Parameters)
- "Call Analysis" (Voice Features Guide for Windows NT)
- "Cadence Detection" (Voice Features Guide for Windows NT)
returns the current state

Name: long ATDX_STATE(chdev)

Inputs: int chdev • valid Dialogic channel device handle

Returns: current state of channel if successful
AT_FAILURE if error

Includes: srllib.h
dxx_xlib.h

Category: Extended Attribute

Description

The ATDX_STATE() function returns the current state of the channel chdev.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using dx_open().</td>
</tr>
</tbody>
</table>

Possible return values are the following:

- CS_DIAL • Dial state
- CS_CALL • Call state
- CS_GTDIG • Get Digit state
- CS_HOOK • Hook state
- CS_IDLE • Idle state
- CS_PLAY • Play state
- CS_RECD • Record state
- CS_STOPD • Stopped state
- CS_TONE • Playing tone state
- CS_WINK • Wink state

**NOTE:** When a Voice board is being used with a FAX/xxx board to send and receive faxes the following states may be returned:

- CS_SENDFAX • Channel is in a fax transmission state.
- CS_RECVFAX • Channel is in a fax reception state.

**NOTE:** A device is idle if there is no I/O function active on it.
ATDX_STATE() returns the current state

■ Cautions

None.

■ Example

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    int chdev;
    long chstate;

    /* Open the channel device */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
        /* Process error */
    }

    /* Examine state of the channel. Perform application specific action based
     * on state of the channel
     */
    if((chstate = ATDX_STATE(chdev)) == AT_FAILURE) {
        /* Process error */
    }

    printf("current state of channel %s = %ld\n", ATDX_NAMEP(chdev), chstate);
}
```

■ Errors

This function will fail and return AT_FAILURE if an invalid channel device handle is specified in `chdev`. 
returns a bitmap

**Name:** long ATDX_TERMMSK(chdev)

**Inputs:** int chdev • valid Dialogic channel device handle

**Returns:** channel’s last termination bitmap if successful
AT_FAILURE if error

**Includes:** srllib.h
dxxxlib.h

**Category:** Extended Attribute

### Description

The **ATDX_TERMMSK**() function returns a bitmap containing the reason(s) for the last termination on the channel **chdev**. The bitmap is reset when an I/O function terminates.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <strong>dx_open</strong>().</td>
</tr>
</tbody>
</table>

Possible return values are the following:

- **TM_NORMTERM** • Normal Termination (for **dx_dial**(), **dx_sethook**())
- **TM_MAXDTMF** • Maximum DTMF count
- **TM_MAXSIL** • Maximum period of silence
- **TM_MAXNOSIL** • Maximum period of non-silence
- **TM_LCOFF** • Loop current off
- **TM_IDDDTIME** • Inter-digit delay
- **TM_MAXTIME** • Maximum function time
- **TM_DIGIT** • Specific digit received
- **TM_PATTERN** • Pattern matched silence off
- **TM_USRSTOP** • Function stopped by user
- **TM_EOD** • End of Data reached on playback
- **TM_TONE** • Tone-on/off event
- **TM_ERROR** • I/O Device Error
ATDX_TERMMSK( ) returns a bitmap

■ Cautions

If several termination conditions are met at the same time, several bits will be set in the termination bitmap.

■ Example

```
#include <stdio.h>
#include <fcntl.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    int chdev;
    long term;
    DX_IOTT iott;
    DV_TPT tpt[4];

    /* Open the channel device */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
        /* Process error */
    }

    /* Record a voice file. Terminate on receiving a digit, silence, loop
     * current drop, max time, or reaching a byte count of 50000 bytes.
     */

    /* set up DX_IOTT */
    iott.io_type = IO_DEV|IO_EOT;
    iott.io_bufp = 0;
    iott.io_offset = 0;
    iott.io_length = 50000;

    if((iott.io_fhandle = dx_fileopen("file.vox", O_RDWR)) == -1) {
        /* process error */
    }

    /* set up DV_TPTs for the required terminating conditions */

    dx_clrpt(tpt,4);
    tpt[0].tp_type = IO_CONT;
    tpt[0].tp_termno = DX_MAXDTMF; /* Maximum digits */
    tpt[0].tp_length = 1; /* terminate on the first digit */
    tpt[0].tp_flags = TF_MAXDTMF; /* Use the default flags */
    tpt[1].tp_type = IO_CONT;
    tpt[1].tp_termno = DX_MAXTIME; /* Maximum time */
    tpt[1].tp_length = 100; /* terminate after 10 secs */
    tpt[1].tp_flags = TF_MAXTIME; /* Use the default flags */
    tpt[2].tp_type = IO_CONT;
    tpt[2].tp_termno = DX_MAXSIL; /* Maximum Silence */
    tpt[2].tp_length = 30; /* terminate on 3 sec silence */
    tpt[2].tp_flags = TF_MAXSIL; /* Use the default flags */
```
returns a bitmap  

```
ATDX_TERMMASK(

  tpt[3].tp_type = IO_EOL;     /* last entry in the table */
  tpt[3].tp_termno = DX_LCOFF; /* terminate on loop current drop */
  tpt[3].tp_length = 10;       /* terminate on 1 sec silence */
  tpt[3].tp_flags = TF_LCOFF;  /* Use the default flags */
/
/* Now record to the file */
if (dx_rec(chdev,&iott,tpt,EV_SYNC) == -1) {
   /* process error */
}
/* Examine bitmap to determine if digits caused termination */
if((term = ATDX_TERMMASK(chdev)) == AT_FAILURE) {
   /* Process error */
}
if(term & TM_MAXDTMF) {
   printf("Terminated on digits\n");
   .
}
```

■ Errors

This function will fail and return AT_FAILURE if an invalid channel device handle is specified in chdev.

■ See Also

Setting Termination Conditions:
- DV_TPT (Appendix A)

Retrieving Termination Events - asynchronously:
- Event Management functions (Standard Runtime Library Programmer’s Guide for Windows NT)
ATDX_TONEID( )

returns the user-defined tone id

Name: long ATDX_TONEID(chdev)

Inputs: int chdev

• valid Dialogic channel device handle

Returns: channel’s last termination bitmap if successful

AT_FAILURE if error

Includes: srllib.h
dxxxlib.h

Category: Extended Attribute

Description

The ATDX_TONEID( ) function returns the user-defined tone id. Use this function to determine which tone occurred when ATDX_TERMMSK( ) returns DX_TONE to indicate that an I/O function terminated due to the occurrence of a user-specified tone.

The function parameter is defined as follows:

Parameter Description

chdev: specifies the valid channel device handle obtained when the channel was opened using dx_open( ).

Cautions

None.

Example

#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
#define TID_1 101

main()
{
    TN_GEN tngen;
    DV_TPT tpt[5];
    int chdev;

    98-CD
returns the user-defined tone id

```c
/* Open the D/xxx Channel Device and Enable a Handler */
if ( ( chdev = dx_open("dxxxB1C1", NULL ) ) == -1 ) {
    error("dxxxB1C1");
    exit( 1 );
}

/* Describe a Simple Dual Tone Frequency Tone of 950-
* 1050 Hz and 475-525 Hz using leading edge detection. */
if ( dx_blddt( TID_1, 1000, 50, 500, 25, TN_LEADING )== -1 ) {
    printf( "Unable to build a Dual Tone Template\n" );
}

/* Add the Tone to the Channel */
if ( dx_addtone( chdev, NULL, 0 ) == -1 ) {
    printf( "Unable to Add the Tone %d\n", TID_1 );
    ATDV_LASTERR( chdev ), ATDV_ERRMSGP( chdev );
    dx_close( chdev );
    exit( 1 );
}

/* Build a Tone Generation Template. */
* This template has Frequency1 = 1140,
* Frequency2 = 1020, amplitude at -10dB for
* both frequencies and duration of 100 * 10 msecs.
* dx_bldtngen( &tngen, 1140, 1020, -10, -10, 100 );

/* Set up the Terminating Conditions */
t[0].tp_type = IO_CONT;
t[0].tp_termno = DX_TUNE;
t[0].tp_length = TID_1;
t[0].tp_flags = TF_TONE;
t[0].tp_data = DX_TONEON;
t[1].tp_type = IO_CONT;
t[1].tp_termno = DX_TUNE;
t[1].tp_length = TID_1;
t[1].tp_flags = TF_TONE;
t[1].tp_data = DX_TONEOFF;
t[2].tp_type = IO_EOT;
t[2].tp_termno = DX_MAXTIME;
t[2].tp_length = 6000;
t[2].tp_flags = TF_MAXTIME;

if (dx_playtone( chdev, &tngen, tpt, EV_SYNC ) == -1 ){
    printf( "Unable to Play the Tone\n" );
    ATDV_LASTERR( chdev ), ATDV_ERRMSGP( chdev );
    dx_close( chdev );
    exit( 1 );
}
```

99-CD
ATDX_TONEID( ) returns the user-defined tone id

if ( ATDX_TERMMSK( chdev ) & TM_TONE ) {
    printf( "Terminated by Tone Id = %d\n", ATDX_TONEID( chdev ) );
}

/* Continue Processing */

/* Close the opened D/xxx Channel Device */
if ( dx_close( chdev ) != 0 ) {
    perror( "close" );
}

/* Terminate the Program */
exit( 0 );

Errors

This function will fail and return AT_FAILURE if an invalid channel device handle is specified in chdev.
returns number of bytes transferred

ATDX_TRCOUNT()

| Name:        | long ATDX_TRCOUNT(chdev) |
| Inputs:      | int chdev • valid Dialogic channel device handle |
| Returns:     | last play/record transfer count if successful |
|             | AT_FAILURE if error |
| Includes:    | srllib.h  |
|             | dxxxlib.h |
| Category:    | Extended Attribute |

Description

The ATDX_TRCOUNT() function returns number of bytes transferred during the last play or record on the channel chdev.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using dx_open().</td>
</tr>
</tbody>
</table>

Cautions

None.

Example

```c
#include <stdio.h>
#include <fcntl.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    int chdev;
    long trcount;
    DX_IOTT iott;
    DV_TPT tpt[2];

    /* Open the channel device */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
        /* Process error */
    }
```
/* Record a voice file. Terminate on receiving a digit, max time, */
/* or reaching a byte count of 50000 bytes. */

/* set up DX_IOTT */
io_type = IO_DEV|IO_EOT;
io_bufp = 0;
io_offset = 0L;
io_length = 50000L;
if((io_fhandle = dx_fileopen("file.vox", O_RDWR)) == -1) {
    /* process error */
}
/* set up DV_TPTs for the required terminating conditions */
dx_clrtpt(tpt,2);
tpt[0].tp_type = IO_CONT;
tpt[0].tp_termno = DX_MAXDTMF;       /* Maximum digits */
tpt[0].tp_length = 1;                /* terminate on the first digit */
tpt[0].tp_flags = TF_MAXDTMF;        /* Use the default flags */
tpt[1].tp_type = IO_EOT;
tpt[1].tp_termno = DX_MAXTIME;       /* Maximum time */
tpt[1].tp_length = 100;              /* terminate after 10 secs */
tpt[1].tp_flags = TF_MAXTIME;        /* Use the default flags */
/* Now record to the file */
if (dx_rec(chdev,&io,tpt,EV_SYNC) == -1) {
    /* process error */
}
/* Examine transfer count */
if((trcount = ATDX_TRCOUNT(chdev)) == AT_FAILURE) {
    /* Process error */
}
printf("%ld bytes recorded\n", trcount);

Errors

This function will fail and return AT_FAILURE if an invalid channel device
handle is specified in chdev.

 ATDX_TRCOUNT( ) returns number of bytes transferred

102-CD
**Description**

`dx_addspddig()` is a convenience function that sets a DTMF digit to adjust speed by a specified amount, immediately and for all subsequent plays on the specified channel (until changed or cancelled).

**NOTES:**
1. Calls to this function are cumulative. To reset a digit condition, you need to clear all adjustment conditions using a `dx_clrsvcond()` function, and then reset the new condition.

This function assumes that the Speed Modification Table has not been modified using the `dx_setsvmt()` function.

For information about the speed and volume functions and the Speed and Volume Modification Tables, see the *Voice Features Guide for Windows NT*. For information about the speed and volume data structures see *Sections DV_TPT*.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained by a call to <code>dx_open()</code></td>
</tr>
</tbody>
</table>
| digit:    | specifies a DTMF digit (0-9, *,#) that will modify speed by the amount specified in `adjval`.  
To start play-speed at the origin, set `digit` to NULL and set `adjval` to `SV_NORMAL`. |
**dx_addspddig( )**

Sets a DTMF digit to adjust speed

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>adjval:</td>
<td>specifies one of the following the speed adjustment values to take effect whenever the digit specified in digit occurs:</td>
</tr>
<tr>
<td>SV_ADD10PCT</td>
<td>Increase play - speed by 10%</td>
</tr>
<tr>
<td>SV_ADD20PCT</td>
<td>Increase play - speed by 20%</td>
</tr>
<tr>
<td>SV_ADD30PCT</td>
<td>Increase play - speed by 30%</td>
</tr>
<tr>
<td>SV_ADD40PCT</td>
<td>Increase play - speed by 40%</td>
</tr>
<tr>
<td>SV_ADD50PCT</td>
<td>Increase play - speed by 50%</td>
</tr>
<tr>
<td>SV_SUB10PCT</td>
<td>Decrease play - speed by 10%</td>
</tr>
<tr>
<td>SV_SUB20PCT</td>
<td>Decrease play - speed by 20%</td>
</tr>
<tr>
<td>SV_SUB30PCT</td>
<td>Decrease play - speed by 30%</td>
</tr>
<tr>
<td>SV_SUB40PCT</td>
<td>Decrease play - speed by 40%</td>
</tr>
<tr>
<td>SV_NORMAL</td>
<td>Set play - speed to origin (regular speed) when the play begins. digit must be set to NULL.</td>
</tr>
</tbody>
</table>

**Cautions**

1. This function is cumulative. To reset or remove any condition, you should clear all conditions, and reset if required (e.g., If DTMF digit "1" has already been set to increase play-speed by one step, a second call that attempts to redefine "1" to the origin, will have no effect on speed or volume but it will be added to the array of conditions. The digit will retain its original setting).  
2. The digit that causes the play adjustment will not be passed to the digit buffer, so it cannot be retrieved using dx_getdig( ) or ATDX_BUFDIGS( )
3. Digits that are used for play adjustment will not be used as a terminating condition. If a digit is defined as both, then the play adjustment will take priority.
4. Speed control is supported on all the D/21D, D/21E, D/41D, D/41E, D/41ESC, D/81A, D/121B, D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1, and D/320SC boards.

**Example**

```c
#include <stdio.h>

104-CD
```
sets a DTMF digit to adjust speed $dx_addspddig()$

```c
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

/*
 * Global Variables
 */

main()
{
    int dxxxdev;

    /*
    * Open the Voice Channel Device and Enable a Handler
    */
    if ( ( dxxxdev = dx_open( "dxxxB1C1", NULL ) ) == -1 ) {
        perror( "dxxxB1C1" );
        exit( 1 );
    }

    /*
    * Add a Speed Adjustment Condition - increase the
    * playback speed by 30% whenever DTMF key 1 is pressed.
    */
    if ( dx_addspddig( dxxxdev, '1', SV_ADD30PCT ) == -1 ) {
        printf("Unable to Add a Speed Adjustment Condition\n");
        printf( "Lasterror = %d  Err Msg = %s\n",
                ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
        dx_close( dxxxdev );
        exit( 1 );
    }

    /*
    * Continue Processing
    */

    /*
    * Close the opened Voice Channel Device
    */
    if ( dx_close( dxxxdev ) != 0 ) {
        perror( "close" );
    }

    /* Terminate the Program */
    exit( 0 );
}

■ Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR() and ATDV_ERRMSGP() to retrieve one of the following error reasons:
dx_addspddig() sets a DTMF digit to adjust speed

- EDX_BADPARM: Invalid Parameter
- EDX_BADPROD: Function not supported on this board
- EDX_SYSTEM: Windows NT system error - check errno
- EDX_SVADJBLK: Invalid Number of Play Adjustment Blocks

See Also

- dx_addvoldig()
- dx_adjsv()
- dx_clrsvcond()
- dx_getcursv()
- dx_getsvmt()
- dx_setsvcond()
- dx_setsvmt()
- "Speed and Volume Modification Tables" (Voice Features Guide for Windows NT)
- DX_SVCB data structure (Chapter 4. Voice Data Structures and Device Parameters)
The \texttt{dx_addtone( )} function adds the tone that was defined by the most recent \texttt{dx_bldtt( )} (or other Global Tone Detection build-tone) function call, to the specified channel. Adding a user-defined tone to a channel downloads it to the board and enables detection of tone-on and tone-off events for that tone by default.

Use \texttt{dx_distone( )} to disable detection of the tone, without removing the tone from the channel. Detection can be enabled again using \texttt{dx_enbtone( )}. For example, if you only want to be notified of tone-on events, you should call \texttt{dx_distone( )} to disable detection of tone-off events.

**Detection Notification**

Tone-on and tone-off events are call status transition events. Retrieval of these events is handled differently for asynchronous and synchronous applications. Table 2 outlines the different processes:

<table>
<thead>
<tr>
<th>Synchronous</th>
<th>Asynchronous</th>
</tr>
</thead>
<tbody>
<tr>
<td>1. Call \texttt{dx_addtone( )}, or \texttt{dx_enbtone( )}</td>
<td>Call \texttt{dx_addtone( )} or \texttt{dx_enbtone( )} to enable tone-on/off detection.</td>
</tr>
<tr>
<td>2. Call \texttt{dx_getevt( )} to wait for CST</td>
<td>Use SRL to asynchronously wait for</td>
</tr>
</tbody>
</table>
**dx_addtone()**  *adds the tone*

Event(s). Events are returned in the **DX_EBLK** structure

**3. N/A**  
Use **sr_getevtdatap()** to retrieve **DX_CST** 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_setevtmsk()**.

You can optionally specify an associated ASCII digit (and digit type) with the tone. When the digit is detected, it is placed in the digit buffer and can be used for termination.

### Setting User-Defined Tones as Termination Conditions

Detection of a user-defined tone can be specified as a termination condition for I/O functions. Set the **tp_termno** field in the **DV_TPT** to **DX_TONE**, and specify **DX_TONEON** or **DX_TONEOFF** in the **tp_data** field.

The function parameters are described below.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <strong>dx_open()</strong>.</td>
</tr>
<tr>
<td>digit:</td>
<td>(optional) specifies the digit to associate with the tone. When the tone is detected, the digit will be placed in the <strong>DV_DIGIT</strong> digit buffer. These digits can be retrieved using <strong>dx_getdig()</strong> (i.e., they can be used in the same way as DTMF digits, for example). If you do not specify a digit, the tone will be indicated by a <strong>DE_TONEON</strong> event or <strong>DE_TONEOFF</strong> event.</td>
</tr>
</tbody>
</table>
| digtype:  | specifies the type of digit the channel will detect. Specify one of the following values.  
|           | - DG_USER1  
|           | - DG_USER2  
|           | - DG_USER3  
|           | - DG_USER4  
|           | - DG_USER5  |

---

108-CD
**Parameter**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Up to twenty digits can be associated with each of these digit types.</td>
</tr>
</tbody>
</table>

**NOTE:** These types can be specified in addition to the digit types already defined for the Voice Library (e.g., DTMF, MF) which are specified using `dx_setdigitp()`.

**Cautions**

1. Ensure that `dx_bldtt()` (or another appropriate "build tone" function) has been called to define a tone prior to adding it to the channel using `dx_addtone()`; otherwise an error will occur.

2. The `dx_addtone()` function may not be used to change a tone that has previously been added.

3. The number of tones that can be added to a channel is dependent on the type of board. See the *Voice Features Guide for Windows NT* for details.

**Example**

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

#define TID_1   101
#define TID_2   102
#define TID_3   103
#define TID_4   104

main()
{
    int dxxxdev;

    /* Open the Voice Channel Device and Enable a Handler */
    if ( ( dxxxdev = dx_open( "dxxxB1C1", NULL) ) == -1 ) {
        perror( "dxxxB1C1" );
        exit( 1 );
    }

    /* Describe a Simple Dual Tone Frequency Tone of 950-1050 Hz and 475-525 Hz using leading edge detection. */
```
dx_addtone() adds the tone

if ( dx_bldtt( TID_1, 1000, 50, 500, 25, TN_LEADING ) == -1 ) {
    printf( "Unable to build a Dual Tone Template\n" );
}

/*
 * Bind the Tone to the Channel
*/
if ( dx_addtone( dxxxdev, NULL, 0 ) == -1 ) {
    printf( "Unable to Bind the Tone %d\n", TID_1 );
    printf( "%s\n", ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
    dx_close( dxxxdev );
    exit( 1 );
}

/*
 * Describe a Dual Tone Frequency Tone of 950-1050 Hz
 * and 475-525 Hz. On between 190-210 msecs and off
 * 990-1010 msecs and a cadence of 3.
*/
if ( dx_bldttcad( TID_2, 1000, 50, 500, 25, 20, 1, 100, 1, 3 ) == -1 ) {
    printf( "Unable to build a Dual Tone Cadence Template\n" );
}

/*
 * Bind the Tone to the Channel
*/
if ( dx_addtone( dxxxdev, 'A', DG_USER1 ) == -1 ) {
    printf( "Unable to Bind the Tone %d\n", TID_2 );
    printf( "%s\n", ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
    dx_close( dxxxdev );
    exit( 1 );
}

/*
 * Describe a Simple Single Tone Frequency Tone of
 * 950-1050 Hz using trailing edge detection.
*/
if ( dx_bldst( TID_3, 1000, 50, TN_TRAILING ) == -1 ) {
    printf( "Unable to build a Single Tone Template\n" );
}

/*
 * Bind the Tone to the Channel
*/
if ( dx_addtone( dxxxdev, 'D', DG_USER2 ) == -1 ) {
    printf( "Unable to Bind the Tone %d\n", TID_3 );
    printf( "%s\n", ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
    dx_close( dxxxdev );
    exit( 1 );
}

/*
 * Describe a Single Tone Frequency Tone of 950-1050 Hz.
 * On between 190-210 msecs and off 990-1010 msecs and
 * a cadence of 3.
*/
if ( dx_bldstcad( TID_4, 1000, 50, 20, 1, 100, 1, 3 ) == -1 ) {
    printf("Unable to build a Single Tone Cadence Template\n");
}

110-CD
adds the tone  

```c
/*
 * Bind the Tone to the Channel
 */
if ( dx_addtone( dxxxdev, NULL, 0 ) == -1 ) {
    printf( "Unable to Bind the Tone %d\n", TID_4 );
    ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev );
    dx_close( dxxxdev );
    exit( 1 );
}

/*
 * Continue Processing
 */

/*
 * Close the opened Voice Channel Device
 */
if ( dx_close( dxxxdev ) != 0 ) {
    perror( "close" );
}

/* Terminate the Program */
exit( 0 );
```

---

**Errors**

If this function returns -1 to indicate failure, use `ATDV_LASTERR()` and `ATDV_ERRMSGP()` to retrieve one of the following error reasons:

<table>
<thead>
<tr>
<th>Code</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_ASCII</td>
<td>Invalid ASCII value in tone template description</td>
</tr>
<tr>
<td>EDX_BADPARM</td>
<td>Invalid parameter</td>
</tr>
<tr>
<td>EDX_BADPROD</td>
<td>Function not supported on this board</td>
</tr>
<tr>
<td>EDX_CADENCE</td>
<td>Invalid cadence component value</td>
</tr>
<tr>
<td>EDX_DIGTYPE</td>
<td>Invalid Dig_Type value in tone template description</td>
</tr>
<tr>
<td>EDX_FREQDET</td>
<td>Invalid tone frequency</td>
</tr>
<tr>
<td>EDX_INVSUBCMD</td>
<td>Invalid sub-command</td>
</tr>
<tr>
<td>EDX_MAXTMPLT</td>
<td>Maximum number of user-defined tones for the board</td>
</tr>
<tr>
<td>EDX_SYSTEM</td>
<td>Windows NT System error - check <code>errno</code></td>
</tr>
<tr>
<td>EDX_TONEID</td>
<td>Invalid tone template ID</td>
</tr>
</tbody>
</table>

---

111-CD
See Also

Global Tone Detection functions:
- `dx_bldtt()`, `dx_bldst()`, `dx_bldttcad()`, `dx_bldstcad()`
- `dx_distone()`
- `dx_enbtone()`
- "Global Tone Detection" (Voice Features Guide for Windows NT)

Event Retrieval:
- `dx_getevt()`
- `DX_CST` data structure

Digit Retrieval:
- `dx_getdig()`
- `dx_setdigtyp()`
- `DV_DIGIT`
sets a DTMF digit to immediately adjust volume  

\[ \text{dx_addvoldig( )} \]

- **Name:** int dx_addvoldig(chdev,digit,adjval)
- **Inputs:**
  - int chdev • valid Dialogic channel device handle
  - char digit • DTMF digit
  - short adjval • volume adjustment value
- **Returns:** 0 if success
  - -1 if failure
- **Includes:** srllib.h
dxxxlib.h
- **Category:** Speed and Volume Convenience

### Description

**dx_addvoldig( )** is a convenience function that sets a DTMF digit to immediately adjust volume by a specified amount for all subsequent plays on the specified channel (until changed or cancelled).

**NOTES:**
1. Calls to this function are cumulative. To reset a digit condition, you need to clear all adjustment conditions using a **dx_clrsvcond( )**, and then reset the new condition.


This function assumes that the Volume Modification Table has not been modified using the **dx_setsvmt( )** function.

For information about the speed and volume functions and the Speed and Volume Modification Tables, see the Voice Features Guide for Windows NT.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained by a call to <strong>dx_open( )</strong>.</td>
</tr>
</tbody>
</table>
| digit:    | specifies a DTMF digit (0-9, *,#) that will modify volume by the amount specified in **adjval**.  
To start play-volume at the origin, set digit to NULL and set adjval to SV_NORMAL. |
| adjval:   | specifies one of the following the speed adjustment values |
**dx_addvoldig()**  
**sets a DTMF digit to immediately adjust volume**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>to take effect whenever the digit specified in <strong>digit</strong> occurs:</td>
</tr>
<tr>
<td>SV_ADD2DB</td>
<td>Increase play-volume by 2DB</td>
</tr>
<tr>
<td>SV_ADD4DB</td>
<td>Increase play-volume by 4DB</td>
</tr>
<tr>
<td>SV_ADD6DB</td>
<td>Increase play-volume by 6DB</td>
</tr>
<tr>
<td>SV_ADD8DB</td>
<td>Increase play-volume by 8DB</td>
</tr>
<tr>
<td>SV_SUB2DB</td>
<td>Decrease play-volume by 2DB</td>
</tr>
<tr>
<td>SV_SUB4DB</td>
<td>Decrease play-volume by 4DB</td>
</tr>
<tr>
<td>SV_SUB6DB</td>
<td>Decrease play-volume by 6DB</td>
</tr>
<tr>
<td>SV_SUB8DB</td>
<td>Decrease play-volume by 8DB</td>
</tr>
<tr>
<td>SV_NORMAL</td>
<td>Set play-volume to origin when the play begins (<strong>digit</strong> must be set to NULL)</td>
</tr>
</tbody>
</table>

**Cautions**

1. This function is cumulative. To reset or remove any condition, you should clear all adjustment conditions, and reset them if required. (e.g., If DTMF digit "1" has already been set to increase play-volume by one step, a second call that attempts to redefine "1" to the origin (regular volume), will have no effect on the volume, but will add it to the condition array. The digit will retain its original setting).

2. The digit that causes the play adjustment will not be passed to the digit buffer, so it cannot be retrieved using **dx_getdig**() and will not be included in the result of **ATDX_BUFDIGS**() which retrieves the number of digits in the buffer.

3. Digits that are used for play adjustment will not be used as a terminating condition. If a digit is defined as both, then the play adjustment will take priority.


**Example**

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
```

*114-CD*
sets a DTMF digit to immediately adjust volume \hspace{1cm} dx_addvoldig()

```
#include <doxxlib.h>
#include <windows.h>

/*
 * Global Variables
*/
main()
{
    int dxdev;

    /*
     * Open the Voice Channel Device and Enable a Handler
     */
    if ( (dxdev = dx_open( "doxxB1C1", NULL ) ) == -1 ) {
        perror( "doxxB1C1" );
        exit( 1 );
    }

    /*
     * Add a Speed Adjustment Condition - decrease the
     * playback volume by 2dB whenever DTMF key 2 is pressed.  */
    if ( dx_addvoldig( dxdev, '2', SV_SUB2DB ) == -1 ) {
        printf( "Unable to Add a Volume Adjustment Condition\n" );
        printf( "Lasterror = %d  Err Msg = %s\n",
                ATDV_LASTERR( dxdev ), ATDV_ERRMSGP( dxdev ) );
        dx_close( dxdev );
        exit( 1 );
    }

    /*
     * Continue Processing
     */

    /*
     * Close the opened Voice Channel Device
     */
    if ( dx_close( dxdev ) != 0 ) {
        perror( "close" );
    }

    /* Terminate the Program */
    exit( 0 );
}
```

### Errors

If this function returns -1 to indicate failure, use \texttt{ATDV\_LASTERR()} and \texttt{ATDV\_ERRMSGP()} to retrieve one of the following error reasons:

- \texttt{EDX\_BADPARM} • Invalid Parameter
- \texttt{EDX\_BADPROD} • Function not supported on this board
dx_addvoldig()  sets a DTMF digit to immediately adjust volume

<table>
<thead>
<tr>
<th>EDX_SVADJBLKS</th>
<th>• Invalid Number of Play Adjustment Blocks</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_SYSTEM</td>
<td>• Windows NT system error - check errno</td>
</tr>
</tbody>
</table>

■ See Also

Related Speed and Volume functions:

- dx_addspddig( )
- dx_adjsv( )
- dx_clrsvcond( )
- dx_getcursv( )
- dx_getsvmt( )
- dx_setsvcond( )
- dx_setsvmt( )
adjusts speed or volume \[dx\_adjsv()\]

**Name:** int \(dx\_adjsv(chdev,tabletype,action,adjsize)\)

**Inputs:**
- int \(chdev\)
  • valid channel device handle
- unsigned short \(tabletype\)
  • table to set (speed or volume)
- unsigned short \(action\)
  • how to adjust (absolute position, relative change or toggle)
- unsigned short \(adjsize\)
  • adjustment size

**Returns:**
- 0 if successful
- -1 if failure

**Includes:** srllib.h
dxxllib.h

**Category:** Speed and Volume

### Description

The \(dx\_adjsv()\) function adjusts speed or volume immediately, and for all subsequent plays on a specified channel (until changed or cancelled). Speed or volume can be set to a specific value, adjusted incrementally, or can be set to toggle. See the **action** parameter description for information.

\(dx\_adjsv()\) utilizes the Speed and Volume Modification Tables to make adjustments to play-speed or play-volume. These tables have 21 entries that represent different levels of speed or volume. There are up to ten levels above and below the regular speed or volume. These tables can be set with explicit values using \(dx\_setsvmt()\) or default values can be used. Refer to the Voice Features Guide for Windows NT for detailed information about these tables.

**NOTES:**

1. This function is similar to \(dx\_setsvcnd()\). Use \(dx\_adjsv()\) to explicitly adjust the play immediately, and use \(dx\_setsvcnd()\) to adjust the play in response to specified conditions. See the description of \(dx\_setsvcnd()\) for more information.

2. Whenever a play is started its speed and volume is based on the most recent modification.

**Parameter** | **Description**
--- | ---
chdev: | specifies the valid channel device handle obtained when the channel was opened using \(dx\_open()\).
tabletype: | specifies whether to modify the play-back using a value from the Speed or the Volume Modification Table.
**dx_adjsv()**  
*adjusts speed or volume*

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>SV_SPEEDTBL</td>
<td>Use the Speed Modification Table</td>
</tr>
<tr>
<td>SV_VOLUMEETBL</td>
<td>Use the Volume Modification Table</td>
</tr>
</tbody>
</table>

**action:** specifies the type of adjustment to make. Set to one of the following:

- **SV_ABSPOS**  
  Set speed or volume to a specified position in the appropriate table. (The position is set using the `adjsize` parameter).

- **SV_RELCURPOS**  
  Adjust speed or volume by the number of steps specified using the `adjsize` parameter.

- **SV_TOGGLE**  
  Toggle between values specified using the `adjsize` parameter.

**adjsize:** specifies the size of the adjustment. `adjsize` has a different value depending on how the adjustment type is set using the action parameter. Set `adjsize` to one of the following:

<table>
<thead>
<tr>
<th>For this action value</th>
<th>Choose this <code>adjsize</code> value</th>
</tr>
</thead>
<tbody>
<tr>
<td>SV_ABSPOS</td>
<td>Specify the position between -10 to +10 in the Speed/Volume Modification Table that contains the required speed or volume adjustment. The origin (regular speed or volume) has a value of 0 in the table.</td>
</tr>
<tr>
<td>SV_RELCURPOS</td>
<td>Specify how many positive or negative &quot;steps&quot; in the Speed/Volume Modification Table by which to adjust the speed or volume. For example, specify -2 to lower the speed or volume by 2 steps in the Speed/Volume Modification Table.</td>
</tr>
<tr>
<td>SV_TOGGLE</td>
<td>Set the values between which speed or volume will toggle.</td>
</tr>
</tbody>
</table>
**SV_TOG ORIGIN** - sets the speed/volume to toggle between the origin and the last modified level of speed/volume.

**SV_CUR ORIGIN** - resets the current speed/volume level to the origin (i.e. regular speed/volume).

**SV_CU RLASTMOD** - sets the current speed/volume to the last modified speed/volume level.

**SV_RESETORIG** - resets the current speed/volume to the origin and the last modified speed/volume to the origin.

### Cautions

Speed and volume control are supported on the D/21D, D/21E, D/41D, D/41E, D/41ESC, D/81A, D/121B, D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1 and D/320SC boards only. Do not use the Speed and Volume control functions to control speed on the D/120, D/121, or D/121A boards.

### Example

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

int main() {
    int dxxxdev;

    /* Open the Voice Channel Device and Enable a Handler */
    if ( (dxxxdev = dx_open( "dxxxB1C1", NULL) ) == -1 ) {
        perror( "dxxxB1C1" );
        exit( 1 );
    }

    /* Modify the Volume of the playback so that it is 4dB higher than normal. */
    if ( dx_adjsv( dxxxdev, SV_VOLU MEBEL, SV_ABSPOS, SV_ADD4DB ) == -1 ) {
        printf( "Unable to Increase Volume by 4dB\n" );
        printf( "Lasterror = %d  Err Msg = %s",
```
dx_adjsv()  \textit{adjusts speed or volume}

\begin{verbatim}
    ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev )
    dx_close( dxxxdev );
    exit( 1 );
}

    /* Continue Processing
    */
    /* Close the opened Voice Channel Device
    */
    if ( dx_close( dxxxdev ) != 0 ) {
        perror "close";
    }

    /* Terminate the Program */
    exit( 0 );
\end{verbatim}

\section*{Errors}

If this function returns -1 to indicate failure, use ATDV_LASTERR() and ATDV_ERRMSGP() to retrieve one of the following error reasons:

\begin{itemize}
    \item EDX_BADPARM  \hspace{1cm}  Invalid Parameter
    \item EDX_BADPROD  \hspace{1cm}  Function not supported on this board
    \item EDX_SYSTEM  \hspace{1cm}  Windows NT system error - check \texttt{errno}
\end{itemize}

\section*{See Also}

Related to Speed and Volume:
\begin{itemize}
    \item dx_setsvmt()
    \item dx_getcursv()
    \item dx_getsvmt()
    \item "Speed and Volume Modification Tables" (Voice Features Guide for Windows NT)
    \item DX_SVMT structure (Chapter 4. Voice Data Structures and Device Parameters)
\end{itemize}

Setting Speed and Volume conditions:
\begin{itemize}
    \item dx_setsvcond()
\end{itemize}

120-CD
* adjusts speed or volume \[ dx_{\text{adjsv}}() \]

- \[ dx_{\text{clrsvcond}}() \]
The `dx_blddt()` function defines a simple dual frequency tone. Subsequent calls to `dx_addtone()` will enable detection of this tone, until another tone is defined.

Issuing a `dx_blddt()` defines a new tone but `dx_addtone()` must be used to add the tone to the channel.

Use `dx_addtone()` to enable detection of the tone on a channel.

Parameter | Description
---|---
tid: | specifies a unique identifier for the tone.\n\n**NOTE:** If you are using R2MF tone detection, reserve the use of tone IDs 101 to 115 for the R2MF tones. See `r2_creatfsig()` for further information.
freq1: | specifies the first frequency (in Hz) for the tone
frq1dev: | specifies the allowable deviation for the first frequency (in Hz).
freq2: | specifies the second frequency (in Hz) for the tone
frq2dev: | specifies the allowable deviation for the second frequency (in Hz)
mode: | specifies whether tone detection notification will occur on the leading or trailing edge of the tone. Set to one of the following: • **TN_LEADING**
defines a simple dual frequency tone

Parameter  Description

- TN_TRAILING

■ Cautions

1. Only one tone per process can be defined at any time. Ensure that `dx_blddt()` is called for each `dx_addtone()` . The tone is not created until `dx_addtone()` is called, and a second consecutive call to `dx_blddt()` will replace the previous tone definition for the channel. If you call `dx_addtone()` without calling `dx_blddt()` an error will occur.

2. If you are using R2MF tone detection, reserve the use of tone ID’s 101 to 115 for the R2MF tones. See `r2_creatfsig()` for further information.

■ Example

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

#define TID_1 101

main()
{
    int dxopdev;
    /*
     * Open the Voice Channel Device and Enable a Handler
     */
    if ( ( dxopdev = dx_open( "dxxxB1C1", NULL ) ) == -1 ) {
        perror( "dxxxB1C1" );
        exit( 1 );
    }
    /*
     * Describe a Simple Dual Tone Frequency Tone of 950-1050 Hz and 475-525 Hz using leading edge detection.
     */
    if ( dx_blddt( TID_1, 1000, 50, 500, 25, TN_LEADING ) == -1 ) {
        printf( "Unable to build a Dual Tone Template\n" );
    }
    /*
     * Continue Processing
     */
}

/*
 * Close the opened Voice Channel Device
 */
```

123-CD
dx_blddt() defines a simple dual frequency tone

if ( dx_close( dxxxdev ) != 0 ) {
    perror( "close" );
}

/* Terminate the Program */
exit( 0 );

Errors
None.

See Also

Global Tone Detection:
  "Global Tone Detection" (Voice Features Guide for Windows NT)

Building Tones:
  • dx_blst()
  • dx_bldtcad()
  • dx_bldstcad()

Enabling Tone Detection:
  • dx_addtone()
  • dx_distone()
  • dx_enhtone()

R2MF Tones:
  • r2_creatfsig()
  • r2_playbsig()
defines a simple dual frequency cadence tone \( \text{dx_blddtcad()} \)

| Name: | \( \text{int dx_blddtcad(tid, freq1, freq1dev, freq2, freq2dev, ontime, ontime_dev, offtime, offtime_dev, repcnt)} \) |
| Inputs: | | |
| | unsigned int \( tid \) | • tone ID to assign. |
| | unsigned int \( freq1 \) | • frequency 1 in Hz |
| | unsigned int \( freq1dev \) | • frequency 1 deviation in Hz |
| | unsigned int \( freq2 \) | • frequency 2 in Hz |
| | unsigned int \( freq2dev \) | • frequency 2 deviation in Hz |
| | unsigned int \( ontime \) | • tone-on time in 10ms |
| | unsigned int \( ontime_dev \) | • tone-on time deviation in 10ms |
| | unsigned int \( offtime \) | • tone-off time in 10ms |
| | unsigned int \( offtime_dev \) | • tone-off time deviation in 10ms |
| | unsigned int \( repcnt \) | • number of repetitions if cadence |
| Returns: | 0 if success |
| | -1 if failure |
| Includes: | srllib.h |
| | dxxxlib.h |
| Category: | Global Tone Detection |

## Description

The \( \text{dx_blddtcad()} \) function defines a simple dual frequency cadence tone. Subsequent calls to \( \text{dx_addtone()} \) will use this tone, until another tone is defined.

A dual frequency cadence tone has dual frequency signals with specific on/off characteristics.

Issuing a \( \text{dx_blddtcad()} \) defines a new tone but \( \text{dx_addtone()} \) must be used to add the tone to the channel.

Use \( \text{dx_addtone()} \) to enable detection of the user-defined tone on a channel.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>( tid )</td>
<td>specifies a unique identifier for the tone.</td>
</tr>
<tr>
<td>( freq1 )</td>
<td>specifies the first frequency (in Hz) for the tone</td>
</tr>
</tbody>
</table>

\( \text{NOTE:} \) If you are using R2MF tone detection, reserve the use of tone IDs 101 to 115 for the R2MF tones. See \( \text{r2_creatfsig()} \) for further information.
dx_blddtcad() \textit{defines a simple dual frequency cadence tone}

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>frq1dev:</td>
<td>specifies the allowable deviation for the first frequency (in Hz).</td>
</tr>
<tr>
<td>freq2:</td>
<td>specifies the second frequency (in Hz) for the tone</td>
</tr>
<tr>
<td>frq2dev:</td>
<td>specifies the allowable deviation for the second frequency (in Hz).</td>
</tr>
<tr>
<td>ontime:</td>
<td>specifies the length of time for which the cadence is &quot;on&quot; (in 10ms units)</td>
</tr>
<tr>
<td>ontddev:</td>
<td>specifies the allowable deviation for &quot;on&quot; time. (in 10ms units)</td>
</tr>
<tr>
<td>offtime:</td>
<td>specifies the length of time for which the cadence is &quot;off&quot; (in 10ms units)</td>
</tr>
<tr>
<td>offtddev:</td>
<td>specifies the allowable deviation for &quot;off&quot; time (in 10ms units).</td>
</tr>
<tr>
<td>repcnt:</td>
<td>specifies the number of repetitions for the cadence (i.e. the number of times that an on/off signal is repeated).</td>
</tr>
</tbody>
</table>

\textbf{Cautions}

1. Only 1 user-defined tone per process can be defined at any time. \texttt{dx_blddtcad() } will replace the previous user-defined tone definition.

2. If you are using R2MF tone detection, reserve the use of tone ID’s 101 to 115 for the R2MF tones. See \texttt{r2_creatfsig()} for further information.

\textbf{Example}

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
#define TID_2   102

main()
{
    int dxxxdev;

    /*
     * Open the Voice Channel Device and Enable a Handler
     */
    if ( ( dxxxdev = dx_open( "dxxxB1C1", NULL ) ) == -1 ) {
```
defines a simple dual frequency cadence tone \textit{dx\_bldtcad()} 

```c
 perror("dxxxB1C1" );
 exit( 1 );
}
/*
 * Describe a Dual Tone Frequency Tone of 950-1050 Hz
 * and 475-525 Hz. On between 190-210 msecs and off
 * 990-1010 msecs and a cadence of 3.
 */
if ( dx_bldtcad( TID\_2, 1000, 50, 500, 25, 20, 1,
         100, 1, 3 ) == -1 ) {
    printf( "Unable to build a Dual Tone Cadence" );
    printf( " Template\n");
}
/*
 * Continue Processing
 * .
 * .
 */
/*
 * Close the opened Voice Channel Device
 */
if ( dx_close( dxxxdev ) != 0 ) {
    perror( "close\n" );
}
/* Terminate the Program */
exit( 0 );
```

## Errors

None.

## See Also

### Global Tone Detection:

- "Global Tone Detection" (Voice Features Guide for Windows NT)

### Building Tones:

- \textit{dx\_bldst()} 
- \textit{dx\_bldtt()} 
- \textit{dx\_bldtscad()}
defines a simple dual frequency cadence tone

Enabling Tone Detection:

- `dx_addtone()`
- `dx_distone()`
- `dx_enhtone()`

R2MF Tones:

- `r2_creatfsig()`
- `r2_playbsig()`
**defines a simple single frequency tone**

```
Name: int dx_bldst(tid,freq,fqdev,mode)
Inputs:
unsigned int tid • tone ID to assign.
unsigned int freq • frequency in Hz
unsigned int fqdev • frequency deviation in Hz
unsigned int mode • leading or trailing edge
Returns: 0 if success
           -1 if failure
Includes: srllib.h
dxxxlib.h
Category: Global Tone Detection
```

---

### Description

The `dx_bldst()` function defines a simple single frequency tone. Subsequent calls to `dx_addtone()` will use this tone, until another tone is defined.

Issuing a `dx_bldst()` defines a new tone but `dx_addtone()` must be used to add the tone to the channel.

Use `dx_addtone()` to enable detection of the user-defined tone on a channel.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>tid:</td>
<td>specifies a unique identifier for the tone.</td>
</tr>
<tr>
<td></td>
<td><strong>NOTE:</strong> If you are using R2MF tone detection, reserve the use of tone IDs 101 to 115 for the R2MF tones. See <code>r2_creatfsig()</code> for further information.</td>
</tr>
<tr>
<td>freq:</td>
<td>specifies the frequency (in Hz) for the tone</td>
</tr>
<tr>
<td>frqdev:</td>
<td>specifies the allowable deviation for the frequency (in Hz).</td>
</tr>
<tr>
<td>mode:</td>
<td>specifies whether detection is on the leading or trailing edge of the tone. Set to one of the following:</td>
</tr>
<tr>
<td></td>
<td>• TN_LEADING</td>
</tr>
<tr>
<td></td>
<td>• TN_TRAILING</td>
</tr>
</tbody>
</table>
**dx_bldst( )**

defines a simple single frequency tone

---

### Cautions

1. Only 1 tone per application may be defined at any time. **dx_bldst( )** will replace the previous user-defined tone definition.

2. If you are using R2MF tone detection, reserve the use of tone IDs 101 to 115 for the R2MF tones. See **r2_creatfsig( )** for further information.

---

### Example

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

#define TID_3   103

main()
{
    int dxxdev;

    /*
     * Open the Voice Channel Device and Enable a Handler
     */
    if ( ( dxxdev = dx_open( "dxxxB1C1", NULL ) ) == -1 ) {
        perror( "dxxxB1C1" );
        exit( 1 );
    }

    /*
     * Describe a Simple Single Tone Frequency Tone of
     * 950-1050 Hz using trailing edge detection.
     */
    if ( dx_bldst( TID_3, 1000, 50, TN_TRAILING ) == -1 ) {
        printf( "Unable to build a Single Tone Template\n" );
    }

    /*
     * Continue Processing
     *   .
     *   .
     *   .
     */

    /*
     * Close the opened Voice Channel Device
     */
    if ( dx_close( dxxdev ) != 0 ) {
        perror( "close" );
    }

    /* Terminate the Program */
    exit( 0 );
}
```

---

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defines a simple single frequency tone \( dx_{\text{bldst}}() \)

- **Errors**

  None.

- **See Also**

  **Global Tone Detection:**
  - "Global Tone Detection" (*Voice Features Guide for Windows NT*)

  **Building Tones:**
  - \( dx_{\text{blddtcad}}() \)
  - \( dx_{\text{blddt}}() \)
  - \( dx_{\text{bldstcad}}() \)

  **Enabling Tone Detection:**
  - \( dx_{\text{addtone}}() \)
  - \( dx_{\text{distone}}() \)
  - \( dx_{\text{enbtone}}() \)

  **R2MF Tones:**
  - \( r2_{\text{creatfsig}}() \)
  - \( r2_{\text{playbsig}}() \)
**dx_bldstcad()** defines a simple single frequency cadence tone

### Name:

```c
int dx_bldstcad(tid, freq, fqdev, ontime, ontdev, offtime, offtdev, repcnt)
```

### Inputs:

- `tid`: tone ID to assign.
- `freq`: frequency in Hz
- `fqdev`: frequency deviation in Hz
- `ontime`: tone on time in 10ms
- `ontdev`: "on" time deviation in 10ms
- `offtime`: tone off time in 10ms
- `offtdev`: "off" time deviation in 10ms
- `repcnt`: repetitions if cadence

### Returns:

- 0 if success
- -1 if failure

### Includes:

- srllib.h
- dxxxlib.h

### Category:

Global Tone Detection

---

**Description**

The `dx_bldstcad()` function 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.

Issuing a `dx_bldstcad()` defines a new tone but `dx_addtone()` must be used to add the tone to the channel.

Use `dx_addtone()` to enable detection of the user-defined tone on a channel.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>tid</code></td>
<td>specifies a unique identifier for the tone.</td>
</tr>
<tr>
<td></td>
<td><strong>NOTE:</strong> If you are using R2MF tone detection, reserve the use of tone IDs 101 to 115 for the R2MF tones. See <code>r2_creatfsig()</code> for further information.</td>
</tr>
<tr>
<td><code>freq</code></td>
<td>specifies the frequency (in Hz) for the tone</td>
</tr>
<tr>
<td><code>fqdev</code></td>
<td>specifies the allowable deviation for the frequency (in Hz).</td>
</tr>
<tr>
<td><code>ontime</code></td>
<td>specifies the length of time for which the cadence is &quot;on.&quot;</td>
</tr>
</tbody>
</table>
defines a simple single frequency cadence tone  \textit{dx\_bldstcad( )}

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>\textit{ontdev:}</td>
<td>specifies the allowable deviation for &quot;on&quot; time in 10 ms units. (10 ms units)</td>
</tr>
<tr>
<td>\textit{offtime:}</td>
<td>specifies the length of time for which the cadence is &quot;off.&quot; (10 ms units)</td>
</tr>
<tr>
<td>\textit{offtdev:}</td>
<td>specifies the allowable deviation for &quot;off&quot; time in 10 ms units.</td>
</tr>
<tr>
<td>\textit{repcnt:}</td>
<td>specifies the number of repetitions for the cadence (i.e., the number of times that an on/off signal is repeated).</td>
</tr>
</tbody>
</table>

\textbf{Example}

```
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

#define TID_4   104

main()
{
    int  dxxxdev;

    /*
    * Open the Voice Channel Device and Enable a Handler
    */
    if ( ( dxxxdev = dx_open( "dxxxB1C1", NULL) ) == -1 ) {
        perror( "dxxxB1C1" );
        exit( 1 );
    }

    /*
    * Describe a Single Tone Frequency Tone of 950-1050 Hz.
    * On between 190-210 msecs and off 990-1010 msecs and
    * a cadence of 3.
    */
    if ( dx_bldstcad( TID_4, 1000, 50, 20, 1, 100, 1, 3 ) == -1 ) {
        printf( "Unable to build a Single Tone Cadence\n" );
        printf( " Template\n" );
    }

    /*
    * Continue Processing
    */
    ...
    ...
    ...
}
```
defines a simple single frequency cadence tone

```c
/*
* Close the opened Voice Channel Device
*/
if ( dx_close( dxxxdev ) != 0 ) {
    perror( "close" );
}

/* Terminate the Program */
exit( 0 );
```

### Cautions

1. Only 1 tone per application may be defined at any time. `dx_bldstcad( )` will replace the previous user-defined tone definition.

2. If you are using R2MF tone detection, reserve the use of tone IDs 101 to 115 for the R2MF tones. See `r2_creatfsig( )` for further information.

### Errors

None.

### See Also

**Global Tone Detection:**
- "Global Tone Detection" *(Voice Features Guide for Windows NT)*

**Building Tones:**
- `dx_bldstcad( )`
- `dx_blddtcad( )`
- `dx_bldtt( )`
- `dx_bldst( )`

**Enabling Tone Detection:**
- `dx_addtone( )`
- `dx_distone( )`
- `dx_enbtone( )`

**R2MF Tones:**
- `r2_creatfsig( )`

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defines a simple single frequency cadence tone  

- r2_playbsig()
dx_bldtngen()  

**sets up tone generation template**

**Name:**  
void dx_bldtngen(tngenp,freq1,freq2,ampl1,ampl2,duration)

**Inputs:**  
*TN_GEN* *tngenp*  
• pointer to tone generation structure  
unsigned short *freq1*  
• frequency of tone 1 in Hz  
unsigned short *freq2*  
• frequency of tone 2 in Hz  
short *ampl1*  
• amplitude of tone 1 in dB  
short *ampl2*  
• amplitude of tone 2 in dB  
short *duration*  
• duration of tone in 10 ms units

**Returns:**  
none

**Includes:**  
srllib.h  
dxxxlib.h

**Category:**  
Global Tone Generation

---

**Description**

dx_bldtngen() is a convenience function that sets up tone generation template (*TN_GEN*) by assigning 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.

**Parameter** | **Description**
--- | ---
*tngenp:* | points to the TN_GEN data structure where the tone generation template is output.
*freq1:* | specifies the frequency of tone 1 in Hz. Valid range is 200 to 3000 Hz.
*freq2:* | specifies the frequency of tone 2 in Hz. Valid range is 200 to 3000 Hz. To define a single tone, set *freq1* to the desired frequency and set *freq2* to 0.
*ampl1:* | specifies the amplitude of tone 1 in dB. Valid range is 0 to -40 dB. Calling this function with *ampl1* set to R2_DEFAMPL will set the amplitude to -10 dB.
*ampl2:* | specifies the amplitude of tone 2 in dB. Valid range is 0 to -40 dB. Calling this function with *ampl2* set to R2_DEFAMPL will set the amplitude to -10 dB.
*duration:* | specifies the duration of the tone in 10 ms units. A value of -1 specifies infinite duration (the tone will only terminate upon an external terminating condition).
Generating a tone with a high frequency component (approximately 700 Hz or higher) will cause the amplitude of the tone to increase. The increase will be approximately 1 dB at 1000 Hz. Also, the amplitude of the tone will increase by 2 dB if an analog (loop start) device is used, such as the LSI/120 board or the analog device on a D/4xD, D/41E, D/41ESC, or D/160SC-LS board.

### Cautions

None.

### Example

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    TN_GEN     tngen;
    int        dxxxdev;

    /* Open the Voice Channel Device and Enable a Handler */
    if ( ( dxxxdev = dx_open( "dxxxB1C1", NULL) ) == -1 ) {
        perror( "dxxxB1C1" );
        exit( 1 );
    }

    /* Build a Tone Generation Template. 
    * This template has Frequency1 = 1140,
    * Frequency2 = 1020, amplitude at -10dB for
    * both frequencies and duration of 100 * 10 msecs.
    */
    dx_bldtngen( &tngen, 1140, 1020, -10, -10, 100 );

    /* Continue Processing */
    
    /* Close the opened Voice Channel Device */
    if ( dx_close( dxxxdev ) != 0 ) {
        perror( "close" );
    }  
}
```
dx_bldtngen() sets up tone generation template

/* Terminate the Program */
exit( 0 );
}

■ Errors

None.

■ See Also

Generating Tones:

- TN_GEN (Chapter 4, Voice Data Structures and Device Parameters)
- dx_playtone()
- "Global Tone Generation" (Voice Features Guide for Windows NT)

R2MF functions:

- r2_creatfsig()
- r2_playbsig()
alters standard definition of duration component  

\textbf{Name:} \texttt{int dx\_chgdur(tonetype, ontime, ondev, offtime, offdev)}

\textbf{Inputs:} \begin{itemize}
\item \texttt{int tonetype}\quad \text{tone to modify} \\
\item \texttt{int ontime}\quad \text{on duration} \\
\item \texttt{int ondev}\quad \text{ontime deviation} \\
\item \texttt{int offtime}\quad \text{off duration} \\
\item \texttt{int offdev}\quad \text{offtime deviation}
\end{itemize}

\textbf{Returns:} \begin{itemize}
\item \texttt{0}\quad \text{success} \\
\item \texttt{-1}\quad \text{tone does not have cadence values} \\
\item \texttt{-2}\quad \text{unknown tone type}
\end{itemize}

\textbf{Includes:} \texttt{srlib.h} \quad \texttt{dxxxlib.h}

\textbf{Category:} PerfectCall Call Analysis

\section*{Description}

The Voice Driver comes with default definitions for each of the PerfectCall Call Analysis tones, which are identified by \texttt{tonetype}. The \texttt{dx\_chgdur()} function alters standard definition of duration component.

Changing a tone definition has no immediate effect on the behavior of an application. The \texttt{dx\_initcallp()} function takes the tone definitions and uses them to initialize a channel. Once a channel is initialized, subsequent changes to the tone definitions have no effect on that channel. For these changes to take effect, \texttt{dx\_deltones} must be called and then followed by calling \texttt{dx\_initcallp}.

\begin{tabular}{|l|l|}
\hline
\textbf{Parameter} & \textbf{Description} \\
\hline
\texttt{tonetype}: & specifies the identifier of the tone whose definition is to be modified. It may be one of the following: \\
\hline
& \begin{itemize}
\item \texttt{TID\_BUSY1}: Busy signal \\
\item \texttt{TID\_BUSY2}: Alternate busy signal \\
\item \texttt{TID\_DIAL\_INTL}: International dial tone \\
\item \texttt{TID\_DIAL\_LCL}: Local dial tone \\
\item \texttt{TID\_DIAL\_XTRA}: Special (“extra”) dial tone \\
\item \texttt{TID\_FAX1}: Fax or modem tone \\
\item \texttt{TID\_FAX2}: Alternate fax or modem tone \\
\item \texttt{TID\_RINGBK1}: Ringback \\
\end{itemize} \\
\hline
\texttt{ontime}: & is the length of time that the tone is on \\
\hline
\end{tabular}
dx_chgdur() alters standard definition of duration component

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ondev:</td>
<td>is the maximum permissible deviation from ontime (10 ms units).</td>
</tr>
<tr>
<td>offtime:</td>
<td>is the length of time that the tone is off (10 ms units).</td>
</tr>
<tr>
<td>offdev:</td>
<td>is the maximum permissible deviation from offtime (10 ms units).</td>
</tr>
</tbody>
</table>

### Example

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    DX_CAP  cap_s;
    int   ddd, car;
    char   *chnam, *dialstrg;

    chnam = "dxxxB1C1";
    dialstrg = "L1234";

    /*
     * Open channel
     */
    if ((ddd = dx_open( chnam, NULL )) == -1 ) {
        /* handle error */
    }

    /*
     * Delete any previous tones
     */
    if ( dx_deltones(ddd) < 0 ) {
        /* handle error */
    }

    /*
     * Change Enhanced call progress default local dial tone
     */
    if (dx_chgfreq( TID_DIAL_LCL, 425, 150, 0, 0 ) < 0) {
        /* handle error */
    }

    /*
     * Change Enhanced call progress default busy cadence
     */
}
```

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alters standard definition of duration component \( dx_{\text{chgdur}}() \)

```c
if (dx_chgdur( TID_BUSY1, 550, 400, 550, 400 ) < 0) { 
    /* handle error */
}

if (dx_chgrepcnt( TID_BUSY1, 4 ) < 0) { 
    /* handle error */
}

/* Now enable Enhanced call progress with above changed settings. 
if (dx_initcallp( ddd )) { 
    /* handle error */
}

/* Set off Hook */
if ((dx_sethook( ddd, DX_OFFHOOK, EV_SYNC )) == -1) { 
    /* handle error */
}

/* Dial */
if ((car = dx_dial( ddd, dialstrg,(DX_CAP *)&cap_s, DX_CALLP|EV_SYNC))==-1) { 
    /* handle error */
}

switch( car ) { 
    case CR_NODIALTONE: 
        printf(" Unable to get dial tone\n");
        break;
    case CR_BUSY: 
        printf(" %s engaged\n", dialstrg );
        break;
    case CR_CNCT: 
        printf(" Successful connection to %s\n", dialstrg );
        break;
    default: 
        break;
}

/* Set on Hook */
if ((dx_sethook( ddd, DX_ONHOOK, EV_SYNC )) == -1) { 
    /* handle error */
}

dx_close( ddd );
```
dx_chgdur( ) alters standard definition of duration component

■ Cautions

This function changes only the definition of a signal. The new definition does not apply to a channel until dx_deltones is called and then dx_initcallp( ) is called on that channel.

■ See Also

- dx_chgfreq( )
- dx_chgrepcent( )
- dx_deltones( )
- dx_initcallp( )
**changes the standard definition**

<table>
<thead>
<tr>
<th>Name:</th>
<th>int dx_chgfreq(tonetype, freq1, freq1dev, freq2, freq2dev)</th>
</tr>
</thead>
</table>
| Inputs: | int tonetype • tone to modify  
| | int freq1 • frequency of first tone  
| | int freq1dev • frequency deviation for first tone  
| | int freq2 • frequency of second tone  
| | int freq2dev • frequency deviation of second tone  |
| Returns: | 0 • success  
| | -1 • failure due to bad parameter(s) for tone type  
| | -2 • failure due to unknown tone type  |
| Includes: | srllib.h  
| | dxxxlib.h  |
| Category: | PerfectCall Call Analysis  |

### Description

The **dx_chgfreq( )** function changes the standard definition for one of the PerfectCall Call Analysis tones, identified by `tonetype`, by modifying its frequency component.

The Voice Driver comes with default definitions for each of the PerfectCall Call Analysis tones; this function alters the frequency component of one of the definitions.

PerfectCall Call Analysis supports both single-frequency and dual-frequency tones. For dual-frequency tones, the frequency and tolerance of each component may be specified independently. For single-frequency tones, specifications for the second frequency are set to zero.

Changing a tone definition has no immediate effect on the behavior of an application. The **dx_initcallp( )** function takes the tone definitions and uses them to initialize a channel. Once a channel is initialized, subsequent changes to the tone definitions have no effect on that channel. For these changes to take effect, **dx_deltones** must be called and then followed by calling **dx_initcallp**.
**dx_chgfreq()**

changes the standard definition

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>tonetype:</td>
<td>specifies the identifier of the tone whose definition is to be modified. It may be one of the following:</td>
</tr>
<tr>
<td></td>
<td>• TID_BUSY1: Busy signal</td>
</tr>
<tr>
<td></td>
<td>• TID_BUSY2: Alternate busy signal</td>
</tr>
<tr>
<td></td>
<td>• TID_DIAL_INTL: International dial tone</td>
</tr>
<tr>
<td></td>
<td>• TID_DIAL_LCL: Local dial tone</td>
</tr>
<tr>
<td></td>
<td>• TID_DIAL_XTRA: Special (&quot;extra&quot;) dial tone</td>
</tr>
<tr>
<td></td>
<td>• TID_FAX1: Fax or modem tone</td>
</tr>
<tr>
<td></td>
<td>• TID_FAX2: Alternate fax or modem tone</td>
</tr>
<tr>
<td></td>
<td>• TID_RINGBK1: Ringback</td>
</tr>
<tr>
<td>freq1:</td>
<td>is the frequency of the first tone (in Hz).</td>
</tr>
<tr>
<td>freq1dev:</td>
<td>is the maximum permissible deviation from freq1 (in Hz).</td>
</tr>
<tr>
<td>freq2:</td>
<td>is the frequency of the second tone, if any (in Hz). If there is only one frequency, freq2 is set to zero.</td>
</tr>
<tr>
<td>freq2dev:</td>
<td>is the maximum permissible deviation from freq2 (in Hz).</td>
</tr>
</tbody>
</table>

**Example**

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    DX_CAP  cap_s;
    int     ddd, car;
    char    *chnam, *dialstrg;

    chnam    = "dxxxB1C1";
    dialstrg = "L1234";

    /*
     * Open channel
     */
    if ( (ddd = dx_open( chnam, NULL )) == -1 ) {
        /* handle error */
    }

    /*
     * Delete any previous tones
     */
    if ( dx_deltones(ddd) < 0 ) {
```
changes the standard definition

dx_chgfreq()

    /* handle error */

    */
    * Change Enhanced call progress default local dial tone
    if (dx_chgfreq( TID_DIAL_LCL, 425, 150, 0, 0 ) < 0) {
        /* handle error */
    }

    */
    * Change Enhanced call progress default busy cadence
    if (dx_chgdur( TID_BUSY1, 550, 400, 550, 400 ) < 0) {
        /* handle error */
    }

    if (dx_chgrepcnt( TID_BUSY1, 4 ) < 0) {
        /* handle error */
    }

    */
    * Now enable Enhanced call progress with above changed settings.
    */
    if (dx_initcallp( ddd )) {
        /* handle error */
    }

    */
    * Set off Hook
    */
    if ((dx_sethook( ddd, DX_OFFHOOK, EV_SYNC )) == -1) {
        /* handle error */
    }

    */
    * Dial
    */
    if ((car = dx_dial( ddd, dialstrg,(DX_CAP *)&cap_s, DX_CALLP|EV_SYNC))==-1) {
        /* handle error */
    }

    switch( car ) {
    case CR_NODIALTONE:
        printf(" Unable to get dial tone\n");
        break;

    case CR_BUSY:
        printf(" %s engaged\n", dialstrg );
        break;

    case CR_CNCT:
        printf(" Successful connection to %s\n", dialstrg );
        break;

    default:
        break;
    }
/ Set on Hook
if (dx_sethook(ddd, DX_ONHOOK, EV_SYNC)) == -1) {
  /* handle error */
}
dx_close(ddd);
The `dx_chgrepcnt()` function changes the standard definition for one of the PerfectCall Call Analysis tones, identified by `tonetype`, by modifying its repetition count component (the number of times that the signal must repeat before being recognized as valid).

The Voice Driver comes with default definitions for each of the PerfectCall Call Analysis tones; this function alters the repetition count component of one of the definitions.

Changing a tone definition has no immediate effect on the behavior of an application. The `dx_initcallp()` function takes the tone definitions and uses them to initialize a channel. Once a channel is initialized, subsequent changes to the tone definitions have no effect on that channel. For these changes to take effect, `dx_deltone` must be called followed by calling `dx_initcallp()`.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>tonetype</code></td>
<td>specifies the identifier of the tone whose definition is to be modified. It may be one of the following:</td>
</tr>
<tr>
<td></td>
<td>• TID_BUSY1: Busy signal</td>
</tr>
<tr>
<td></td>
<td>• TID_BUSY2: Alternate busy signal</td>
</tr>
<tr>
<td></td>
<td>• TID_DIAL_INTL: International dial tone</td>
</tr>
<tr>
<td></td>
<td>• TID_DIAL_LCL: Local dial tone</td>
</tr>
<tr>
<td></td>
<td>• TID_DIAL_XTRA: Special (“extra”) dial tone</td>
</tr>
<tr>
<td></td>
<td>• TID_FAX1: Fax or modem tone</td>
</tr>
<tr>
<td></td>
<td>• TID_FAX2: Alternate fax or modem tone</td>
</tr>
</tbody>
</table>
dx_chgrepCnt() changes the standard definition

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>TID_RINGBK1: Ringback</td>
<td>repcnt: is the number of times that the signal must repeat.</td>
</tr>
</tbody>
</table>

**Example**

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    DX_CAP cap_s;
    int ddd, car;
    char *chnam, *dialstrg;

    chnam = "dxxxB1C1";
    dialstrg = "L1234";

    /*
    *  Open channel
    */
    if ((ddd = dx_open( chnam, NULL )) == -1 ) {
        /* handle error */
    }

    /*
    *  Delete any previous tones
    */
    if ( dx_deltones(ddd) < 0 ) {
        /* handle error */
    }

    /*
    *  Change Enhanced call progress default local dial tone
    */
    if (dx_chgfreq( TID_DIAL_LCL, 425, 150, 0, 0 ) < 0) {
        /* handle error */
    }

    /*
    *  Change Enhanced call progress default busy cadence
    */
    if (dx_chgdur( TID_BUSY1, 550, 400, 550, 400 ) < 0) {
        /* handle error */
    }

    if (dx_chgrepCnt( TID_BUSY1, 4 ) < 0) {
        /* handle error */
    }
}
```

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changes the standard definition  


\begin{verbatim}
/*
 *  Now enable Enhanced call progress with above changed settings.
 */
if (dx_initcallp( ddd )) {
    /* handle error */
}

/*
 *  Set off Hook
 */
if (dx_sethook( ddd, DX_OFFHOOK, EV_SYNC ) == -1) {
    /* handle error */
}

/*
 *  Dial
 */
if (dx_dial( ddd, dialstrg,(DX_CAP *)&cap_s, DX_CALLP|EV_SYNC) ==-1) {
    /* handle error */
}

switch( car ) {
    case CR_NODIALTONE:
        printf(" Unable to get dial tone\n");
        break;
    case CR_BUSY:
        printf(" %s engaged\n", dialstrg );
        break;
    case CR_CNCT:
        printf(" Successful connection to %s\n", dialstrg );
        break;
    default:
        break;
}

/*
 *  Set on Hook
 */
if (dx_sethook( ddd, DX_ONHOOK, EV_SYNC ) == -1) {
    /* handle error */
}

dx_close( ddd );
\end{verbatim}

### Cautions

This function changes only the definition of a tone. The new definition does not apply to a channel until \texttt{dx_initcallp()} is called on that channel.
dx_chgrepct() changes the standard definition

See Also

- dx_chgdur()
- dx_chgfreq()
- dx_deltones()
- dx_initcallp()
**dx_close()**

**Name:** int dx_close(dev)

**Inputs:**
- int dev
- valid Dialogic channel or board device handle

**Returns:**
- 0 if successful
- -1 if error

**Includes:** srllib.h
dxxxlib.h

**Category:** Device Management

---

**Description**

The **dx_close()** function closes Dialogic devices opened previously by using **dx_open()**. It releases the handle and breaks any link the calling process has with the device through this handle. It will release the handle whether the device is busy or idle.

**NOTE:** **dx_close()** disables the generation of all events. It does not affect the hookstate or any of the parameters that have been set for the device.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dev</td>
<td>specifies the valid Dialogic device handle obtained when a board or channel was opened using <strong>dx_open()</strong>.</td>
</tr>
</tbody>
</table>

---

**Cautions**

Once a device is closed, a process can no longer perform any action on that device using that device handle. Other handles for that device that exist in the same process or other processes will still be valid. The only process affected by **dx_close()** is the process that called the function.

**NOTES:**

1. The **dx_close()** function doesn’t affect any action occurring on a device, it only breaks the link between the calling process and the device by freeing the specified device handle. Other links through different device handles are still valid.

2. Never use the Windows NT **close()** function to close a Voice device; unpredictable results will occur.
dx_close() closes Dialogic devices

3. dx_close() will discard any outstanding events on that handle.

Example

#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
main()
{
    DX_CAP cap;
    int chdev;
    /* continue processing */
    if (dx_close (chdev) ==-1)

Errors

If this function returns -1 to indicate failure, check errno for one of the following reasons:

- EINVAL  • Invalid Argument
- EBADF  • Invalid file descriptor
- EINTR  • A signal was caught
clears all the fields in a DX_CAP structure

**dx_clrcap( )**

**Name:** void dx_clrcap(capp)

**Inputs:**

DX_CAP *capp

- pointer to Call Analysis Parameter Structure

**Returns:** None

**Includes:**
srllib.h
dxxxlib.h

**Category:** Structure Clearance

---

### Description

The **dx_clrcap( )** function clears all the fields in a DX_CAP structure by setting them to zero. **dx_clrcap( )** is a VOID function that returns no value. It is provided as a convenient way of clearing a DX_CAP structure.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>capp:</td>
<td>points to the DX_CAP structure. See Chapter 4. Voice Data Structures and Device Parameters for information about the DX_CAP structure.</td>
</tr>
</tbody>
</table>

### Cautions

The DX_CAP structure should be cleared and using **dx_clrcap( )** before the structure is used as an argument in a **dx_dial( )** function call. This will prevent parameters from being set unintentionally.

### Example

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

main()
{
    DX_CAP cap;
    int chdev;

    /* open the channel using dx_open */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
        /* process error */
    }
}  
```

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dx_clrccap( ) clears all the fields in a DX_CAP structure

}.

; /* set call analysis parameters before doing call analysis */
dx_clrccap(&cap);
cap.ca_nbrdna = 5; /* 5 rings before no answer */
;
; /* continue with call analysis */
;
}

- Errors

None.

- See Also
  
  - dx_dial()
  - DX_CAP (Chapter 4. Voice Data Structures and Device Parameters)
  - "Call Analysis" (Voice Features Guide for Windows NT)
**dx_clrdigbuf()**

*causes the digits present in the firmware digit buffer*

**Name:** int dx_clrdigbuf(chdev)

**Inputs:**
- int chdev
- valid Dialogic channel device handle

**Returns:**
- 0 if success
- -1 if failure

**Includes:**
- srllib.h
- dxxxlib.h

**Category:** Configuration

### Description

The `dx_clrdigbuf()` function causes the digits present in the firmware digit buffer of the channel specified by `chdev` to be flushed.

The function parameter is defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <code>dx_open()</code>.</td>
</tr>
</tbody>
</table>

### Cautions

None.

### Example

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

main()
{
    int chdev; /* channel descriptor */
    .
    .
    /* Open Channel */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
        /* process error */
    }

    /* Clear digit buffer */
```
dx_clrdigbuf() causes the digits present in the firmware digit buffer

    if (dx_clrdigbuf(chdev) == -1) {
        /* process error*/
    }
}

See the function references for dx_getdig(), dx_play(), and dx_rec() for more examples of how to use dx_clrdigbuf().

### Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR() and ATDV_ERRMSGP() to retrieve one of the following error reasons:

- **EDX_BADPARM** • Invalid Parameter
- **EDX_SYSTEM** • Windows NT system error - check 

errno
The `dx_clrsvcond()` function clears any speed or volume adjustment conditions that have been previously set with the `dx_setsvcond()` function or the convenience functions `dx_addspddig()` or `dx_addvoldig()`.

Each time you want to reset a single adjustment condition, you must reset all adjustment conditions, by first clearing them using this function, and then resetting the conditions using `dx_setsvcond()`, `dx_addspddig()` or `dx_addvoldig()`.

### Parameter Description

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
</table>
| chdev:    | specifies the valid channel device handle obtained when the channel was opened using `dx_open()`.

### Cautions

None.

### Example

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

int main()
{
    int dxxxdev;

    /* Open the Voice Channel Device and Enable a Handler */
    if ( ( dxxxdev = dx_open( "dxxxB1C1", NULL) ) == -1 ) {
```
dx_clrsvcond() clears any speed or volume adjustment conditions

```c
    perror("dxclrSVCl\n"); exit(1);

    /*
    * Clear all Speed and Volume Conditions
    */
    if ( dx_clrsvcond( dxxxdev ) == -1 ) {
        printf( "Unable to Clear the Speed/Volume\n" );
        printf( "Lasterror = %d  Err Msg = %s\n",
                ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
        dx_close( dxxxdev );
        exit(1);
    }

    /*
    * Continue Processing
    */
    /*
    */

    /*
    * Close the opened Voice Channel Device
    */
    if ( dx_close( dxxxdev ) != 0 ) {
        perror( "close\n" );
    }

    /* Terminate the Program */
    exit(0);
}
```

Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR() and ATDV_ERRMSGP() to retrieve one of the following error reasons:

- EDX_BADPARM: Invalid Parameter
- EDX_BADPROD: Function not supported on this board
- EDX_SYSTEM: Windows NT system error - check errno

See Also

- dx_setsvcond()
- dx_addspddig()
- dx_addvoldig()
- "Speed and Volume Modification Tables" (Voice Features Guide for Windows NT)
clears any speed or volume adjustment conditions \textit{dx_clrvcond( )}

- \textit{DX_SVCB (Chapter 4. Voice Data Structures and Device Parameters)}
**dx_clrtpt( )**  
*clears all DV_TPT fields*

<table>
<thead>
<tr>
<th>Name:</th>
<th>int dx_clrtpt(tptp,size)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>DV_TPT *tptp</td>
</tr>
<tr>
<td></td>
<td>• pointer to termination parameter table structure</td>
</tr>
<tr>
<td></td>
<td>int size</td>
</tr>
<tr>
<td></td>
<td>• number of entries to clear</td>
</tr>
<tr>
<td>Returns:</td>
<td>0 if success</td>
</tr>
<tr>
<td></td>
<td>-1 if failure</td>
</tr>
<tr>
<td>Includes:</td>
<td>srllib.h</td>
</tr>
<tr>
<td></td>
<td>dxxxlib.h</td>
</tr>
<tr>
<td>Category:</td>
<td>Structure Clearance</td>
</tr>
</tbody>
</table>

### Description

The **dx_clrtpt( )** function clears all DV_TPT fields except **tp_type** and **tp_nextp** in the number of DV_TPT structures indicated in the **size** parameter. **dx_clrtpt( )** is provided as a convenient way of clearing a DV_TPT structure, if this is required before initializing it for a new set of terminating conditions.

**NOTE:** The DV_TPT is defined in *srllib.h* since it can be used by other non-Voice devices. Valid Voice values for the DV_TPT are listed in *Appendix A*, and the DV_TPT structure is described in detail in the *Standard Runtime Library Programmer’s Guide*.

Prior to calling **dx_clrtpt( )**, you must set the **tp_type** field of DV_TPT as follows:

- **IO_CONT** • if the next DV_TPT is contiguous
- **IO_LINK** • if the next DV_TPT is linked
- **IO_EOT** • for the last DV_TPT

If **tp_type** is set to **IO_LINK**, you **MUST** set **tp_nextp** to point to the next DV_TPT in the chain. **dx_clrtpt( )** uses the information in **tp_type**, and in **tp_nextp** if **IO_LINK** is set, to access the next DV_TPT. By setting the **tp_type** and **tp_nextp** fields appropriately, **dx_clrtpt( )** can be used to clear a combination of contiguous and linked DV_TPT structures.

The function parameters are defined as follows:
**dx_clrtpt( )** clears all DV_TPT fields

### Parameter Description

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>tptp:</td>
<td>points to the first DV_TPT to be cleared. See Appendix A for information about the DV_TPT.</td>
</tr>
<tr>
<td>size:</td>
<td>indicates the number of DV_TPT structures to clear. If size is set to 0, the function will return a 0 to indicate success.</td>
</tr>
</tbody>
</table>

#### Cautions

**dx_clrtpt( )** uses the information present in tp_type and tp_nextp (if IO_LINK is set) to access the next DV_TPT in the chain. The last DV_TPT in the chain must have its tp_type field set to IO_EOT. If the DV_TPTs have to be reinitialized with a new set of conditions, **dx_clrtpt( )** must be called only *after* the links have been set up, as illustrated below.

#### Example

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

main()
{
    DV_TPT tpt1[2];
    DV_TPT tpt2[2];

    /* Set up the links in the DV_TPTs */
    tpt1[0].tp_type = IO_CONT;
    tpt1[1].tp_type = IO_LINK;
    tpt1[1].tp_nextp = &tpt2[0];

    tpt2[0].tp_type = IO_CONT;
    tpt2[1].tp_type = IO_EOT;
    /* set up the other DV_TPT fields as required for termination */
    ...
    /* play a voice file, get digits, etc. */
    ...
    /* clear out the DV_TPT structures if required */
    dx_clrtpt(&tpt1[0], 4);
    /* now set up the DV_TPT structures for the next play */
    ...
}
```
dx_clrtpt() clears all DV_TPT fields

■ Errors

The function will fail and return -1 if IO_EOT is encountered in the tp_type field before the number of DV_TPT structures specified in size have been cleared.

■ See Also

- DV_TPT (Chapter 4. Voice Data Structures and Device Parameters)
removes all user-defined tones

\texttt{dx\_deltones( )}

**Name:** \texttt{int dx\_deltones(chdev)}

**Inputs:**
- \texttt{int chdev} • valid Dialogic channel device handle

**Returns:**
- \texttt{0} • Success
- \texttt{-1} • Error return code

**Includes:**
- \texttt{srllib.h}
- \texttt{dxxxlib.h}

**Category:** Global Tone Detection

## Description

The \texttt{dx\_deltones( )} function removes all user-defined tones previously added to a channel with \texttt{dx\_addtone( )}. If no user-defined tones were previously enabled for this channel, this function has no effect.

**NOTE:** Calling this function deletes ALL user-defined tones defined by \texttt{dx\_bldtt( )}, \texttt{dx\_bdst( )}, \texttt{dx\_bdstd( )}, or \texttt{dx\_bldtstd( )}.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid Dialogic channel device handle obtained when the channel was opened using \texttt{dx_open( )}.</td>
</tr>
</tbody>
</table>

## Cautions

None.

## Example

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    int dxdev;
    /*
    * Open the Voice Channel Device and Enable a Handler
    */
    if ( ( dxdev = dx_open( "dxxxB1C1", NULL) ) == -1 ) {
        perror( "dxxxB1C1" );
        exit( 1 );
    }
}
```

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dx_deltones( ) removes all user-defined tones

/* Delete all Tone Templates */
if ( dx_deltones( dxxxdev ) == -1 ) {
    printf( "Unable to Delete all the Tone Templates\n" );
    printf( "Lasterror = %d  Err Msg = %s\n", ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
    dx_close( dxxxdev );
    exit( 1 );
}

/* Continue Processing */
if ( dx_close( dxxxdev ) != 0 ) {
    perror( "close" );
}
/* Terminate the Program */
exit( 0 );

Errors

If the function returns -1 to indicate failure, call ATDV_LASTERR( ) and ATDV_ERRMSGP( ) to return one of the following errors:

- EDX_BADPARM: Invalid parameter
- EDX_BADPROD: Function not supported on this board
- EDX_SYSTEM: Windows NT System error - check errno

See Also

Adding and Enabling User-defined Tones:
- dx_addtone( )
- dx_enbtone( )

Building Tones:
- dx_bldtt( )
- dx_bldtt( )
- dx_bldstcad( )
removes all user-defined tones \textit{dx\_deltones()} 

- \textit{dx\_blddtcad()}
dx_dial()  

dials an ASCIIZ string

Name:  int dx_dial(chdev,dialstrp,capp,mode)
Inputs:  int chdev  • valid Dialogic channel device handle
char *dialstrp  • pointer to the ASCIIZ dial string
DX_CAP *capp  • pointer to Call Analysis Parameter Structure
unsigned short mode  • asynchronous/synchronous setting and Call Analysis flag

Returns:  0 to indicate successful initiation (Asynchronous)
>=0 to indicate Call Analysis result if successful (Synchronous)
-1 if failure

Includes:  srllib.h
dxxxlib.h

Category:  I/O
Mode:  synchronous/asynchronous

Description

The dx_dial() function dials an ASCIIZ string on an open, idle channel and optionally enables Call Analysis to provide information about the call.

To determine the state of the channel during a dial and/or Call Analysis, use ATDX_STATE( ), which will return one of the following:

<table>
<thead>
<tr>
<th>State</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>CS_DIAL</td>
<td>dial state (with or without Call Analysis)</td>
</tr>
<tr>
<td>CS_CALL</td>
<td>Call Analysis state</td>
</tr>
</tbody>
</table>

NOTE: dx_dial() doesn’t affect the hook state.

dx_dial() without Call Analysis enabled cannot be terminated using dx_stopch( ), unlike most I/O functions.

The function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using dx_open( ).</td>
</tr>
</tbody>
</table>
dials an ASCIIZ string

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dialstrp:</td>
<td>points to the ASCII dial string. <strong>dialstrp</strong> must contain a null-terminated string of ASCII characters. Valid dialing and control characters are described Table 3.</td>
</tr>
</tbody>
</table>

Table 3 Valid Characters for Each Dialing Mode

<table>
<thead>
<tr>
<th>Pulse Digit</th>
<th>Description</th>
<th>DTMF Digit</th>
<th>Description</th>
<th>MF Digit</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>“0”-.“9”</td>
<td>“0”-.“9”</td>
<td>“0”-.“9”</td>
<td>“0”-.“9”</td>
<td>“0”-.“9”</td>
<td>“0”-.“9”</td>
</tr>
<tr>
<td>“#”</td>
<td>“#”</td>
<td>“#”</td>
<td>“#”</td>
<td>“#”</td>
<td>“#”</td>
</tr>
<tr>
<td>“*”</td>
<td>“*”</td>
<td>“*”</td>
<td>“*”</td>
<td>“*”</td>
<td>“*”</td>
</tr>
<tr>
<td>“#”</td>
<td>“#”</td>
<td>“#”</td>
<td>“#”</td>
<td>“#”</td>
<td>“#”</td>
</tr>
<tr>
<td>“a”</td>
<td>“a”</td>
<td>“a”</td>
<td>“a”</td>
<td>“a”</td>
<td>“a”</td>
</tr>
<tr>
<td>“b”</td>
<td>“b”</td>
<td>“b”</td>
<td>“b”</td>
<td>“b”</td>
<td>“b”</td>
</tr>
<tr>
<td>“c”</td>
<td>“c”</td>
<td>“c”</td>
<td>“c”</td>
<td>“c”</td>
<td>“c”</td>
</tr>
<tr>
<td>“d”</td>
<td>“d”</td>
<td>“d”</td>
<td>“d”</td>
<td>“d”</td>
<td>“d”</td>
</tr>
<tr>
<td>“,”</td>
<td>pause</td>
<td>“,”</td>
<td>pause</td>
<td>“,”</td>
<td>pause</td>
</tr>
<tr>
<td>“;”</td>
<td>flash</td>
<td>“;”</td>
<td>flash</td>
<td>“;”</td>
<td>flash</td>
</tr>
<tr>
<td>“I”</td>
<td>“I”</td>
<td>“I”</td>
<td>“I”</td>
<td>“I”</td>
<td>“I”</td>
</tr>
<tr>
<td>“X”</td>
<td>“X”</td>
<td>“X”</td>
<td>“X”</td>
<td>“X”</td>
<td>“X”</td>
</tr>
<tr>
<td>Change dial mode to:</td>
<td>Change dial mode to:</td>
<td>Change dial mode to:</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

The dialing mode is specified by a “T” (DTMF tone dialing), “P” (pulse dialing), or “M” (MF dialing) in the dial string. If “T”, “P”, or “M” is not specified in **dialstrp**, DTMF tone dialing is used.

**NOTE:** MF dialing is only available on systems with MF
dx_dial() dials an ASCII string capability such as the D/4xD board with MF support, or a D/21E, D/41E, D/41ESD, D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1, D/320SC board.

The pause character “,” and the flash character “&” are not available in MF dialing mode. To send these characters while sending a string of MF digits, switch to DTMF or pulse mode before sending “,” or “&”, then switch back to MF mode by sending an “M”. The following string demonstrates this use:

M*1234T,M5678a

capp: points to the Call Analysis Parameter Structure, DX_CAP. This structure is described in (Chapter 4. Voice Data Structures and Device Parameters).

To use the default Call Analysis parameters, specify NULL in capp and DX_CALLP in mode.

The D/40 board does not have the Call Analysis feature. When using dx_dial(), do not pass capp to D/40 channels; pass a NULL pointer and do not set mode to DX_CALLP.

mode specifies whether an ASCII string should be dialed with or without Call Analysis enabled, and whether the function should run asynchronously or synchronously. mode is a bit mask that can be set to a combination of the following values:

DX_CALLP • Enable Call Analysis.

EV_ASYNC • Run dx_dial() asynchronously.

EV_SYNC • Run dx_dial() synchronously. (default)

To run dx_dial() without Call Analysis, specify only EV_ASYNC or EV_SYNC.

NOTE: If dx_dial() is called on a channel that is onhook, the function will only dial digits. Call analysis will not occur.
dials an ASCIIZ string \textit{dx_dial( )}

\begin{itemize}
\item \textbf{Asynchronous Operation}
\end{itemize}

Set the \textbf{mode} field to EV\_ASYNC, using a bitwise OR. When running asynchronously, the function will return 0 to indicate it has initiated successfully, and will generate one of the following termination events to indicate completion:

- TDX\_DIAL \quad \bullet \quad \text{termination of dialing (without Call Analysis)}
- TDX\_CALLP \quad \bullet \quad \text{termination of dialing (with Call Analysis)}

If asynchronous \textit{dx_dial( )} terminates with a TDX\_DIAL event, use \texttt{ATDX\_TERMMSK( )} to determine the reason for termination. If \textit{dx_dial( )} terminates with a TDX\_CALLP event, use \texttt{ATDX\_CPTERM( )} to determine the reason for termination. These Call Analysis termination reasons are listed under the description of Call Analysis, below.

Use the SRL Event Management functions to handle the termination event. See Appendix A for more information about the Event Management functions.

\begin{itemize}
\item \textbf{Synchronous Operation}
\end{itemize}

By default, this function runs synchronously, and will return a 0 to indicate that it has completed successfully.

When synchronous dialing terminates, the function will return the Call Progress result (if Call Analysis is enabled) or 0 to indicate success (if Call Analysis isn’t enabled).

\begin{itemize}
\item \textbf{Call Analysis}
\end{itemize}

Call Analysis provides information about the call. It is enabled to run on the call after dialing completes by setting the \textbf{mode} field. The function can be set to run using default Call Analysis parameters, or by using the Call Analysis Parameter structure (DX\_CAP).

Call Analysis results can be retrieved using \texttt{ATDX\_CPTERM( )}.

If \textit{dx_dial( )} is running synchronously, the Call Analysis results will also be returned by the function.
**dx_dial()**

*dials an ASCIIZ string*

For more information about Call Analysis see the *Features Guide*.

Possible Call Analysis termination reasons are listed below:

- **CR_BUSY** • line was busy
- **CR_CEPT** • operator intercept
- **CR_CNCT** • call connected
- **CR_ERROR** • Call Analysis error
- **CR_FAXTONE** • fax machine or modem
- **CR_NOANS** • no answer
- **CR_NODIALTONE** • no dial tone
- **CR_NORB** • no ringback
- **CR_STOPD** • Call Analysis stopped due to **dx_stopch()**

If Call Analysis is enabled, additional information about the call can be obtained using the following Extended Attribute functions:

- **ATDX_ANSRSIZ()** • Returns duration of answer
- **ATDX_CPERROR()** • Returns Call Analysis error
- **ATDX_CPTERM()** • Returns last Call Analysis termination
- **ATDX_CRNID()** • Returns tone identifier
- **ATDX_DTNFAIL()** • Returns dial tone fail character
- **ATDX_FRQDUR()** • Returns duration of first frequency detected
- **ATDX_FRQDUR2()** • Returns duration of second frequency detected
- **ATDX_FRQDUR3()** • Returns duration of third frequency detected
- **ATDX_FRQHZ()** • Returns frequency detected in Hz
- **ATDX_FRQHZ2()** • Returns frequency of second detected tone
- **ATDX_FRQHZ3()** • Returns frequency of third detected tone
- **ATDX_LONGLOW()** • Returns duration of longer silence
- **ATDX_FRQOUT()** • Returns percent of frequency out of bounds
- **ATDX_SHORTLOW()** • Returns duration of shorter silence
- **ATDX_SIZEHI()** • Returns duration of non-silence

**Cautions**

1. If you attempt to dial a channel in MF mode and do not have MF capabilities on that channel, DTMF tone dialing is used.
2. Issuing a `dx_stopch()` on a channel that is dialing without Call Analysis enabled has no effect on the dial, and will return 0. The digits specified in the `dialstrp` parameter will still be dialed.

3. Issuing a `dx_stopch()` on a channel that is dialing with Call Analysis enabled will cause the dialing to complete, but Call Analysis will not be executed. The digits specified in the `dialstrp` parameter will be dialed. Any Call Analysis information collected prior to the stop will be returned by Extended Attribute functions.

4. This function must be issued when the channel is idle.

**Example 1: Call Analysis with user-specified parameters (Synchronous Mode)**

```c
/* Call Analysis with user-specified parameters and synchronous mode. */

#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    int cares, chdev;
    DX_CAP capp;

    /* open the channel using dx_open(). Obtain channel device descriptor in */
    /* chdev */
    if ((chdev = dx_open("dxxxB1C1", NULL)) == -1) {
        /* process error */
    }

    /* take the phone off-hook */
    if ((dx_sethook(chdev, DX_OFFHOOK, EV_SYNC)) == -1) {
        /* process error */
    }

    /* Clear DX_CAP structure */
    dx_clrcap(&capp);

    /* Set the DX_CAP structure as needed for call analysis. */
    /* Allow 3 rings before no answer. */
    capp.ca_nbrdna = 3;

    /* Perform the outbound dial with call analysis enabled. */
    if ((cares = dx_dial(chdev, "5551212", &capp, DX_CALLP | EV_SYNC)) == -1) {
        /* perform error routine */
    }
}  
```
dx_dial( )  

**dials an ASCIIZ string**

```c
switch (cares) {
    case CR_CNCT: /* Call Connected, get some additional info */
        printf("\nDuration of short low - %ld ms",ATDX_SHORTLOW(chdev)*10);
        printf("\nDuration of long low - %ld ms",ATDX_LONGLOW(chdev)*10);
        printf("\nDuration of answer - %ld ms",ATDX_ANSRSIZ(chdev)*10);
        break;
    case CR_CEPT: /* Operator Intercept detected */
        printf("\nFrequency detected - %ld Hz",ATDX_FRQHZ(chdev));
        printf("\n%% of Frequency out of bounds - %ld Hz",ATDX_FRQOUT(chdev));
        break;
    case CR_BUSY:
        
        /* carry out the next state */
        .
        .
    }
}
```

**Example 2: Call Analysis with default parameters (Synchronous Mode)**

```c
/* Call Analysis with default parameters and synchronous mode. */

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

main()
{
    int cares, chdev;
    DX_CAP capp;

    /* open the channel using dx_open(). Obtain channel device descriptor */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
        /* process error */
    }

    /* take the phone off-hook */
    if ((dx_setopt(chdev,DX_OFFHOOK,EV_SYNC)) == -1) {
        /* process error */
    }

    /* Perform the outbound dial with call analysis enabled and capp set to NULL */
    if (((cares = dx_dial(chdev, "5551212", (DX_CAP *)NULL, DX_CALLP|EV_SYNC)) == -1) {
        /* perform error routine */
    }

    /* Analyze the call analysis results as in Example 1 */
    .
    .
}
```

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Example 3: Call Analysis with default parameters (Asynchronous, Callback Mode)

/* Call Analysis with user-specified parameters and asynchronous, callback mode. */

#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

#define MAXCHAN 24

int dial_handler();

DX_CAP capp;

main()
{
    int i, chdev[MAXCHAN];
    char *chnamep;
    int srlmode;

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

    for (i=0; i<MAXCHAN; i++) {
        /* Set chnamep to the channel name, e.g., dxxxB1C1, dxxxB1C2,... */

        /* Open the device using dx_open(). chdev[i] has channel device descriptor. */
        if ((chdev[i] = dx_open(chnamep,NULL)) == -1) {
            /* process error */
        }

        /* Using sr_embhdlr(), set up handler function to handle call analysis completion events on this channel. */
        if (sr_embhdlr(chdev[i], TDX_CALLP, dial_handler) == -1) {
            /* process error */
        }

        /* Before issuing dx_dial(), place the phone off-hook. */

        /* Clear DX_CAP structure */
        dx_clrcap(&capp);

        /* Set the DX_CAP structure as needed for call analysis. */
        /* Allow 3 rings before no answer. */
        capp.ca_nbrdna = 3;
    }
}
dx_dial() dials an ASCIIZ string

/* Perform the outbound dial with call analysis enabled. */
if (dx_dial(chdev[i],"5551212",&cap_s,DX_CALLP|EV_ASYNC) == -1) {
        /* perform error routine */
}

/* Use sr_waitevt() to wait for the completion of call analysis.*/
* On receiving the completion event, TDX_CALLP, control is transferred
* to the handler function previously established using sr_enbhdlr().
*/
.
.

int dial_handler()
{
    int chdev;
    chdev = sr_getevtdev();
    switch (ATDX_CPTERM(chdev)) {
        case CR_CNCT: /* Call Connected, get some additional info */
            printf("\nDuration of short low - %ld ms",ATDX_SHORTLOW(chdev)*10);
            printf("\nDuration of long low  - %ld ms",ATDX_LONGLOW(chdev)*10);
            printf("\nDuration of answer    - %ld ms",ATDX_ANSRSIZ(chdev)*10);
            break;
        case CR_CEPT:     /* Operator Intercept detected */
            printf("\nFrequency detected - %ld Hz",ATDX_FRQHZ(chdev));
            printf("\n%% of Frequency out of bounds - %ld Hz",ATDX_FRQOUT(chdev));
            break;
        case CR_BUSY:     /* Kick off next function in the state machine model. */
            .
            .
            return 0;
    }

Example 4: PerfectCall Call Analysis (Synchronous Mode)

#include <stdio.h>

#include <srlib.h>
#include <dxlib.h>
#include <windows.h>

main()
{
    DX_CAP  cap_s;
    int     ddd, car;
    char    *chnam, *dialstrg;

    chnam = "dxxxB1C1";
dials an ASCII string

```c

char dialstrg = "L1234";
/* Open channel */
if ((ddd = dx_open( chnam, NULL )) == -1 ) {
    /* handle error */
}

/* Delete any previous tones */
if ( dx_deltones(ddd) < 0 ) {
    /* handle error */
}

/* Change Enhanced call progress default local dial tone */
if (dx_chgfreq( TID_DIAL_LCL, 425, 150, 0, 0 ) < 0) {
    /* handle error */
}

/* Change Enhanced call progress default busy cadence */
if (dx_chgdkdr( TID_BUSY1, 550, 400, 550, 400 ) < 0) {
    /* handle error */
}

if (dx_chgrepcnt( TID_BUSY1, 4 ) < 0) {
    /* handle error */
}

/* Now enable Enhanced call progress with above changed settings. */
if (dx_initcallp( ddd )) {
    /* handle error */
}

/* Set off Hook */
if (dx_sethook( ddd, DX_OFFHOOK, EV_SYNC )) == -1) {
    /* handle error */
}

/* Dial */
if ( (car = dx_dial( ddd, dialstrg, (DX_CAP *)&cap_s, DX_CALLP|EV_SYNC ))== -1) {
    /* handle error */
}

switch( car ) {
    case CR_NODIALTONE:
        printf(" Unable to get dial tone\n");
        Break;
    case CR_BUSY:
```
dx_dial( )

Dials an ASCIIZ string

printf(" %s engaged\n", dialstrg );
b

case CR_CNCT:
    printf(" Successful connection to %s\n", dialstrg );
    break;

default:
    break;

/*
 * Set on Hook
 */
if ((dx_setbook( ddd, DX_ONHOOK, EV_SYNC )) == -1) {
    /* handle error */
}
    dx_close( ddd );

---

Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR( ) and ATDV_ERRMSGP( ) to retrieve one of the following error reasons:

<table>
<thead>
<tr>
<th>Error Code</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_BADPARM</td>
<td>Invalid Parameter</td>
</tr>
<tr>
<td>EDX_BUSY</td>
<td>Channel is busy</td>
</tr>
<tr>
<td>EDX_SYSTEM</td>
<td>Windows NT system error - check errno</td>
</tr>
</tbody>
</table>

---

See Also

- dx_stopch( )

Retrieving termination reasons and events for dx_dial( ) with Call Analysis:

- Event Management functions (Standard Runtime Library Programmer’s Guide for Windows NT)
- ATDX_CPTERM( )

Retrieving termination reasons for dx_dial( ) without Call Analysis:

- ATDX_TERMMSK( )

---

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dials an ASCII string \texttt{dx_dial( )}

Call Analysis:

- \texttt{DX\_CAP} (Chapter 4. Voice Data Structures and Device Parameters)
- "Call Analysis" (Voice Features Guide for Windows NT)
- \texttt{ATDX\_ANSRSIZ( )}
- \texttt{ATDX\_CPERROR}
- \texttt{ATDX\_FRQDUR*( )}
- \texttt{ATDX\_FRQHZ*( )}
- \texttt{ATDX\_FRQOUT( )}
- \texttt{ATDX\_LONGLOW( )}
- \texttt{ATDX\_SHORTLOW( )}
- \texttt{ATDX\_SIZEHI( )}
**dx_distone()**

Disables detection of TONE ON

<table>
<thead>
<tr>
<th>Name:</th>
<th>int dx_distone(chdev,toneid,evt_mask)</th>
</tr>
</thead>
</table>
| Inputs: | int chdev • channel device  
int toneid • tone template identification  
int evt_mask • event mask |
| Returns: | =0 • Success  
-1 • Error return code |
| Category: | Global Tone Detection |

### Description

The `dx_distone()` function disables detection of TONE ON and/or TONE OFF for a user-defined tone on a channel. Detection capability for user-defined tones is enabled on a channel by default when `dx_addtone()` is called.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <code>dx_open()</code></td>
</tr>
</tbody>
</table>
| toneid:   | specifies the user-defined tone identifier for which detection is being disabled.  
To disable detection of all user-defined tones on the channel, set **toneid** to TONEALL. |
| evt_mask: | specifies whether to disable detection of the user-defined tone going on or going off. Set to one or both of the following using a bitwise-OR (|) operator.  
• DM_TONEON disable TONE ON detection  
• DM_TONEOFF disable TONE OFF detection  
**evt_mask** affects the enabled/disabled status of the tone template and will remain in effect until `dx_distone()` or `dx_enbtone()` is called again to reset it. |

### Example

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
```
#define TID_1 101

main()
{
  int dxxxdev;

  /* * Open the Voice Channel Device and Enable a Handler */
  if ( ( dxxxdev = dx_open( "dxxxB1C1", NULL ) ) == -1 ) {
    perror( "dxxxB1C1" );
    exit( 1 );
  }

  /* * Describe a Simple Dual Tone Frequency Tone of 950-1050 Hz and 475-525 Hz using leading edge detection. */
  if ( dx_blddt( TID_1, 1000, 50, 500, 25, TN_LEADING ) == -1 ) {
    printf( "Unable to build a Dual Tone Template\n" );
  }

  /* * Bind the Tone to the Channel */
  if ( dx_addtone( dxxxdev, NULL, 0 ) == -1 ) {
    printf( "Unable to Bind the Tone %d\n", TID_1 );
    printf( "Error Code = %d  Err Msg = %s\n", ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
    dx_close( dxxxdev );
    exit( 1 );
  }

  /* * Disable Detection of ToneId TID_1 */
  if ( dx_distone( dxxxdev, TID_1, DM_TONEON | DM_TONEOFF ) == -1 ) {
    printf( "Unable to Disable Detection of Tone %d\n", TID_1 );
    printf( "Error Code = %d  Err Msg = %s\n", ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
    dx_close( dxxxdev );
    exit( 1 );
  }

  /* * Continue Processing */

  /* * Close the opened Voice Channel Device */
  if ( dx_close( dxxxdev ) != 0 ) {
    perror( "close" );
  }

  /* Terminate the Program */
  exit( 0 );
}

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\textit{dx\_distone( )} \hspace{1cm} \textit{disables detection of TONE ON}

\section*{Errors}

If the function returns -1 to indicate failure, call \texttt{ATDX\_LASTERR( )} and \texttt{ATDV\_ERRMSGP( )} to return one of the following errors:

- \texttt{EDX\_BADPARM} \hspace{1cm} \textbullet{} Invalid parameter
- \texttt{EDX\_BADPROD} \hspace{1cm} \textbullet{} Function not supported on this board
- \texttt{EDX\_SYSTEM} \hspace{1cm} \textbullet{} Windows NT System error - check \texttt{errno}
- \texttt{EDX\_TNMSGSTATUS} \hspace{1cm} \textbullet{} Invalid message status setting
- \texttt{EDX\_TONEID} \hspace{1cm} \textbullet{} Bad tone ID

\section*{See Also}

\textbf{Global Tone Detection functions:}

- \textit{dx\_addtone( )}
- \textit{dx\_bldtt( ), dx\_bldst( ), dx\_bldttcad( ), dx\_bldstcad( )}
- \textit{dx\_enbtone( )}
- "Global Tone Detection" (Voice Features Guide for Windows NT)

\textbf{Event Retrieval:}

- \textit{dx\_getevt( )}
- \textit{DX\_CST} data structure
The `dx_enbtone()` function enables detection of TONE ON and/or TONE OFF for a user-defined tone on a channel. Detection capability for tones is enabled on a channel by default when `dx_addtone()` is called.

The description of `dx_addtone()` (earlier in this chapter) explains how to synchronously and asynchronously retrieve CST tone on and tone off events.

Use this function to enable a tone that has been disabled using `dx_distone()`.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
</table>
| chdev:    | specifies the valid channel device handle obtained when the channel was opened using `dx_open()`.
| toneid:   | specifies the user-defined tone identifier for which detection is being enabled. To enable detection of all user-defined tones on the channel, set `toneid` to `TONEALL`.
| evt_mask: | specifies whether to enable detection of the user-defined tone going on or going off. Set to one or both of the...
**dx_enbtone( )**

Enables detection of **TONE ON**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>following using a bitwise -OR (</td>
</tr>
<tr>
<td>• DM_TONEON</td>
<td>disable TONE ON detection</td>
</tr>
<tr>
<td>• DM_TONEOFF</td>
<td>disable TONE OFF detection</td>
</tr>
<tr>
<td>evt_mask</td>
<td>affects the enabled/disabled status of the tone template and will remain in effect until dx_enbtone( ) or dx_distone( ) is called again to reset it.</td>
</tr>
</tbody>
</table>

### Example

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

#define TID_1   101

main()
{
  int dxxxdev;

  /*
   * Open the Voice Channel Device and Enable a Handler
   */
  if ( ( dxxxdev = dx_open( "dxxxB1C1", NULL ) ) == -1 ) {
    perror( "dxxxB1C1" );
    exit( 1 );
  }

  /*
   * Describe a Simple Dual Tone Frequency Tone of 950-1050 Hz and 475-525 Hz using leading edge detection.
   */
  if ( dx_blddt( TID_1, 1000, 50, 500, 25, TN_LEADING ) == -1 ) {
    printf( "Unable to build a Dual Tone Template\n" );
  }

  /*
   * Bind the Tone to the Channel
   */

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```
enables detection of TONE ON

```c
if ( dx_addtone( dxxxdev, NULL, 0 ) == -1 ) {
    printf( "Unable to Bind the Tone %d\n", TID_1 );
    printf( "Lasterror = %d  Err Msg = %s\n",
            ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
    dx_close( dxxxdev );
    exit( 1 );
}

/*
 * Enable Detection of ToneId TID_1
 */
if ( dx_enbtone( dxxxdev, TID_1, DM_TONEON | DM_TONEOFF ) == -1 ) {
    printf( "Unable to Enable Detection of Tone %d\n", TID_1 );
    printf( "Lasterror = %d  Err Msg = %s\n",
            ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
    dx_close( dxxxdev );
    exit( 1 );
}

/*
 * Continue Processing
 *:
 */

/*
 * Close the opened Voice Channel Device
 */
if ( dx_close( dxxxdev ) != 0 ) {
    perror( "close" );
}

/* Terminate the Program */
exit( 0 );
```

## Cautions

None.

## Errors

If the function returns -1 to indicate failure, call ATDX_LASTERR() and ATDV_ERRMSGP() to return one of the following errors:

- **EDX_BADPARM**: Invalid parameter
- **EDX_BADPROD**: Function not supported on this board
- **EDX_SYSTEM**: Windows NT System error - check **errno**
- **EDX_TONEID**: Bad tone ID
- **EDX_TNMSGSTATUS**: Invalid message status setting

---

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dx_enbtone( ) enables detection of TONE ON

- See Also

Global Tone Detection:
- dx_addtone( )
- dx_bldtt(), dx_bldst(), dx_bldttcad(), dx_bldstcad()
- dx_distone( )
- "Global Tone Detection" (Voice Features Guide for Windows NT)

Event Retrieval:
- dx_getevt( )
- DX_CST data structure
**dx_fileclose()**

**Name:** int dx_fileclose(handle)

**Inputs:** int handle

- handle returned from dx_fileopen()

**Returns:**
- 0 if success
- -1 if failure

**Category:** File Management

---

**Description**

The dx_fileclose() function closes the file associated with the handle returned by the dx_fileopen() function. See the _close function in the Microsoft Visual C++ Run-Time Library Reference for more information.

**Cautions**

Use dx_fileclose() instead of _close to ensure the compatibility of applications with the libraries across various versions of Visual C++.

**Example**

```c
/* Play a voice file. Terminate on receiving 4 digits or at end of file */
#include <fcntl.h>
#include <srilib.h>
#include <dxxxlib.h>
#include <windows.h>
main()
{
  int chdev;
  DX_IOTT iott;
  DV_TPT tpt;
  DV_DIGIT dig;
  
  /* Open the device using dx_open(). Get channel device descriptor in */
  /* chdev. */
  if (chdev = dx_open("dxxxB1C1",NULL)) == -1) { /* process error */
    } /* set up DX_IOTT */
  iott.io_type = IO_DEV|IO_EOL;
  iott.io_bufp = 0;
  iott.io_offset = 0;
  iott.io_length = -1; /* play till end of file */
  if((iott.io_handle = dx_fileopen("prompt.vox",
    O_RDONLY|O_BINARY)) == -1) { /* process error */
    }
```
dx_fileclose()  

  closes the file associated with the handle

/* set up DV_TPT */
dx_clrtpt(&tpt,1);
tpt.tp_type   = IO_EOT;          /* only entry in the table */
tpt.tp_termno = DX_MAXDTMF;      /* Maximum digits */
tpt.tp_length = 4;               /* terminate on four digits */
tpt.tp_flags  = TF_MAXDTMF;      /* Use the default flags */
/* clear previously entered digits */
if (dx_clrdigbuf(chdev) == -1)  {  
  /* process error */
}
/* Now play the file */
if (dx_play(chdev,&iott,&tpt,EV_SYNC) == -1)  {  
  /* process error */
}
/* get digit using dx_getdig() and continue processing. */
.
.
if (dx_fileclose(iott.io_handle) == -1)  {  
  /* process error */
}

Errors

If this function returns -1 to indicate failure, errno is set to EBADF to indicate an invalid file-handle parameter.

See Also

- dx_fileopen()
- dx_fileseek()
- dx_fileread()
- dx_filewrite()
**dx_fileopen()**

<table>
<thead>
<tr>
<th>Name:</th>
<th>int dx_fileopen(filep, flags, pmode)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>const char *filep</td>
</tr>
<tr>
<td></td>
<td>int flags</td>
</tr>
<tr>
<td></td>
<td>int pmode</td>
</tr>
<tr>
<td>Returns:</td>
<td>file handle if success</td>
</tr>
<tr>
<td></td>
<td>-1 if failure</td>
</tr>
<tr>
<td>Category:</td>
<td>File Management</td>
</tr>
</tbody>
</table>

**Description**

The `dx_fileopen()` function opens the file specified by `filep` and prepares the file for reading and writing, as specified by `flags`. See the `_open` function in the Microsoft Visual C++ Run-Time Library Reference for more information.

**Cautions**

Use `dx_fileopen()` instead of `_open` to ensure the compatibility of applications with the libraries across various versions of Visual C++.

**Example**

```c
/* Play a voice file. Terminate on receiving 4 digits or at end of file*/
#include <fcntl.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
main()
{
    int chdev;
    DX_IOTT iott;
    DV_TPT  tpt;
    DV_DIGIT dig;
.
    /* Open the device using dx_open(). Get channel device descriptor in
     * chdev. */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
        /* process error */
    }
.
    /* set up DX_IOTT */
    iott.io_type = IO_DEV|IO_EOT;
    iott.io_bufp = 0;
    iott.io_offset = 0;
    iott.io_length = -1;  /* play till end of file */
    if((iott.io_handle = dx_fileopen("prompt.vox", O_RDONLY|O_BINARY)) == -1) {
        /* process error */
    }
}  
```
**dx_fileopen( )**

*opens the file specified by filep*

```c
/* set up DV_TPT */
dx_clrtpt(&tpt,1);
tpt.tp_type = IO_EOT; /* only entry in the table */
tpt.tp_termno = DX_MAXDTMF; /* Maximum digits */
tpt.tp_length = 4; /* terminate on four digits */
tpt.tp_flags = TF_MAXDTMF; /* Use the default flags */
/* clear previously entered digits */
if (dx_clrdigbuf(chdev) == -1) {
  /* process error */
}
/* Now play the file */
if (dx_play(chdev,&iot,&tpt,EV_SYNC) == -1) {
  /* process error */
}
/* get digit using dx_getdig( ) and continue processing. */
.
.
if (dx_fileclose(iott.io_handle) == -1) {
  /* process error */
}
```

### Errors

If this function returns -1 to indicate failure, *errno* is set to one of the following values:

- **EACCES**
  - Tried to open read only file for writing, file’s sharing mode does not allow specified operations, or given path is directory

- **EEXIST**
  - `_O_CREAT` and `_O_EXCL` flags specified, but filename already exists

- **EINVAL**
  - Invalid flags or *pmod* argument

- **EMFILE**
  - Invalid flags or *pmod* argument

- **ENOENT**
  - File or path not found

### See Also

- `dx_fileclose( )`
- `dx_fileseek( )`
- `dx_fileread( )`
- `dx_filewrite( )`

---

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### Name:

`int dx_fileread(handle, buffer, count)`

### Inputs:

- `int handle` • handle returned from `dx_fileopen()`
- `void *buffer` • storage location for data
- `unsigned int count` • maximum number of bytes

### Returns:

- number of bytes if success
- `-1` if failure

### Category:

File Management

---

## Description

The `dx_fileread()` function turns number of bytes read by application. The function will read the number of bytes from the file associated with the handle into the buffer. The number of bytes read may be less than the value of `count` if there are fewer than `count` bytes left in the file or if the file was opened in text mode. See the `_read` function in the *Microsoft Visual C++ Run-Time Library Reference* for more information.

## Cautions

Use `dx_fileread()` instead of `_read` to ensure the compatibility of applications with the libraries across various versions of Visual C++.

## Example

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

int cd;                 /* channel descriptor */
DX_UIO myio;            /* user definable I/O structure */
/* User defined I/O functions */
int my_read(fd,ptr,cnt)
int fd;
char * ptr;
unsigned cnt;
{
    printf("My read\n");
    return(dx_fileread(fd,ptr,cnt));
}
/*
 * my write function
 */
int my_write(fd,ptr,cnt)
```
dx_fileread() turns number of bytes read by application.

```c
int fd;
char * ptr;
unsigned cnt;
{
    printf("My write \n");
    return(dx_filewrite(fd,ptr,cnt));
}
/*
 * my seek function
 */
long my_seek(fd,offset,whence)
int fd;
long offset;
int whence;
{
    printf("My seek\n");
    return(dx_fileseek(fd,offset,whence));
}
void main(argc,argv)
int argc;
char *argv[];
{
    /* Other initialization */

    DX_UIO uioblk;
    /* Initialize the UIO structure */
    uioblk.u_read=my_read;
    uioblk.u_write=my_write;
    uioblk.u_seek=my_seek;
    /* Install my I/O routines */
    dx_setuio(devhandle,uioblk);
    vodat_fd = dx_fileopen("JUNK.VOX",O_RDWR|O_BINARY);
    /*This block uses standard I/O functions */
    iott->io_type = IO_DEV|IO_CONT
    iott->io_fhandle = vodat_fd;
    iott->io_offset = 0;
    iott->io_length = 20000;
    /*This block uses my I/O functions */
    iottp++;
    iottp->io_type = IO_DEV|IO_UIO|IO_CONT
    iottp->io_fhandle = vodat_fd;
    iottp->io_offset = 20002;
    iottp->io_length = 20000;
    /*This block uses standard I/O functions */
    iottp++;
    iottp->io_type = IO_DEV|IO_DEV
    iottp->io_fhandle = vodat_fd;
    iottp->io_offset = 0;
    iottp->io_length = 20000;
    /*This block uses my I/O functions */
    devhandle = dx_open("dxxxB1C1", 0);
    dx_sethook(devhandle, DX-HOOK, EV_SYNC)
    dx_string(devhandle, DX_OFFHOOK, EV_SYNC);
    dx_clrdigbuf;
    if(dx_rec(devhandle,iott,(DX_TPT*)NULL,RT_TONE|EV_SYNC) == -1) {
        perror("***");
        exit(1);
    }
    dx_clrdigbuf(devhandle);
    if(dx_play(devhandle,iott,(DX_TPT*)EV_SYNC) == -1 {
```

190-CD
turns number of bytes read by application.

dx_fileread()

```c
    perror("**");
    exit(1);
}
dx_close(devhandle);
```

### Errors

If this function returns -1 to indicate failure, `errno` is set to `EBADF` which indicates an invalid file-handle parameter, a closed file, or a locked file.

### See Also

- `dx_fileopen( )`
- `dx_fileclose( )`
- `dx_fileseek( )`
- `dx_filewrite( )`
**dx_fileseek( )** moves file pointer associated with handle

<table>
<thead>
<tr>
<th>Name:</th>
<th>long dx_fileseek(handle, offset, origin)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>int handle • handle returned from dx_fileopen()</td>
</tr>
<tr>
<td></td>
<td>long offset • number of bytes from the origin</td>
</tr>
<tr>
<td></td>
<td>int origin • initial position</td>
</tr>
<tr>
<td>Returns:</td>
<td>number of bytes read if success</td>
</tr>
<tr>
<td></td>
<td>-1 if failure</td>
</tr>
<tr>
<td>Category:</td>
<td>File Management</td>
</tr>
</tbody>
</table>

**Description**

The `dx_fileseek( )` function moves file pointer associated with handle to a new location that is `offset` bytes from `origin`. The function returns the offset, in bytes, of the new position from the beginning of the file. See the `_lseek` function in the *Microsoft Visual C++ Run-Time Library Reference* for more information.

**Cautions**

Use `dx_fileseek( )` instead of `_lseek` to ensure the compatibility of applications with the libraries across various versions of Visual C++.

**Example**

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

int cd;                 /* channel descriptor */
DX_UIO myio;            /* user definable I/O structure */

/* User defined I/O functions */
int my_read(fd,ptr,cnt)
   int fd;
   char * ptr;
   unsigned cnt;
   { printf("My read\n");
     return(dx_fileread(fd,ptr,cnt));
   }
/* * my write function */
int my_write(fd,ptr,cnt)
   int fd;
```

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char * ptr;
unsigned cnt;
{
printf("My write \n");
return(dx_filewrite(fd,ptr,cnt));
}
/*
 * my seek function
 */
long my_seek(fd,offset,whence)
int fd;
long offset;
int whence;
{
printf("My seek\n");
return(dx_fileseek(fd,offset,whence));
}
void main(argc,argv)
int argc;
char *argv[];
{
    /* Other initialization */

    DX_UIO uiblk;
    /* Initialize the UIO structure */
    uiblk.u_read=my_read;
    uiblk.u_write=my_write;
    uiblk.u_seek=my_seek;
    /* Install my I/O routines */
    vodat_fd = dx_fileopen("JUNK.VOX",O_RDWR|O_BINARY);
    /*This block uses standard I/O functions */
    iott->io_type = IO_DEV|IO_CONT
    iott->io_fhandle = vodat_fd;
    iott->io_offset = 0;
    iott->io_length = 20000;
    /*This block uses my I/O functions */
    iottp++;
    iottp->io_type = IO_DEV|IO_UIO|IO_CONT
    iottp->io_fhandle = vodat_fd;
    iottp->io_offset = 20001;
    iottp->io_length = 20000;
    /*This block uses standard I/O functions */
    iottp++
    iottp->io_type = IO_DEV|IO_CONT
    iottp->io_fhandle = vodat_fd;
    iottp->io_offset = 20002;
    iottp->io_length = 20000;
    /*This block uses my I/O functions */
    iott->io_type = IO_DEV|IO_UIO|IO_CONT
    iott->io_fhandle = vodat_fd;
    iott->io_offset = 10003;
    iott->io_length = 20000;
    devhandle = dx_open("dxxxB1C1", NULL);
    dx_sethook(devhandle,DX_ONHOOK,EV_SYNC)
    dx_wtring(devhandle,1,DX_OFFHOOK,EV_SYNC);
    dx_clrdbgbuf;
    if(dx_rec(devhandle,iott,(DX_TPT*)NULL,RM_TONE|EV_SYNC) == -1) {
        perror("**");
        exit(1);
    }
    dx_clrdbgbuf(devhandle);
    if(dx_play(devhandle,iott,(DX_TPT*)EV_SYNC) == -1 {
        perror("**");
    }

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dx_fileseek( )  moves file pointer associated with handle

exit(1);
}
dx_close(devhandle);

■ Errors

If this function returns -1 to indicate failure, errno is set to the following values:

- EBADF  • Invalid file-handle parameter, a closed file, or a locked file.
- EINVAL  • Value for origin is invalid or the position specified by offset is before the beginning of the file

On devices incapable of seeking, the return value is undefined.

■ See Also

- dx_fileopen( )
- dx_fileclose( )
- dx_fileread( )
- dx_filewrite( )
writes count bytes from buffer into file associated with handle  \texttt{dx\_fwrite( )}

<table>
<thead>
<tr>
<th>Name:</th>
<th>int \texttt{dx_fwrite(handle, buffer, count)}</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>int handle</td>
</tr>
<tr>
<td></td>
<td>• handle returned from \texttt{dx_fileopen( )}</td>
</tr>
<tr>
<td></td>
<td>void *buffer</td>
</tr>
<tr>
<td></td>
<td>• data to be written</td>
</tr>
<tr>
<td></td>
<td>unsigned int count</td>
</tr>
<tr>
<td></td>
<td>• number of bytes</td>
</tr>
<tr>
<td>Returns:</td>
<td>number of bytes if success</td>
</tr>
<tr>
<td></td>
<td>-1 if failure</td>
</tr>
<tr>
<td>Category:</td>
<td>File Management</td>
</tr>
</tbody>
</table>

\section*{Description}

The \texttt{dx\_fwrite( )} function writes count bytes from buffer into file associated with handle. The write operation begins at the current position of the file pointer (if any) associated with the given file. If the file was opened for appending, the operation begins at the current end of the file. After the write operation, the file pointer is increased by the number of bytes actually written. See the \texttt{\_write} function in the \textit{Microsoft Visual C++ Run-Time Library Reference} for more information.

\section*{Cautions}

Use \texttt{dx\_fwrite( )} instead of \texttt{\_write} to ensure the compatibility of applications with the libraries across various versions of Visual C++.

\section*{Example}

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

int cd; /* channel descriptor */
DX_UIO myio; /* user definable I/O structure */

/* User defined I/O functions */
int my_read(fd,ptr,cnt)
int fd;
int ptr;
unsigned cnt;
{
    printf("My read\n");
    return(dx_fileread(fd,ptr,cnt));
}

/* my write function

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dx_filewrite()  writes count bytes from buffer into file associated with handle

int my_write(fd,ptr,cnt)
  int fd;
  char * ptr;
  unsigned cnt;
{
  printf("My write \n");
  return(dx_filewrite(fd,ptr,cnt));
}

/*
 * my seek function
*/
long my_seek(fd,offset,whence)
  int fd;
  long offset;
  int whence;
{
  printf("My seek\n");
  return(dx_fileseek(fd,offset,whence));
}

void main(argc,argv)
  int argc;
  char *argv[];
{
  /* Other initialization */
  DX_UIO uioblk;
  /* Initialize the UIO structure */
  uioblk.u_read=my_read;
  uioblk.u_write=my_write;
  uioblk.u_seek=my_seek;
  /* Install my I/O routines */
  dx_setuio(devhandle,uioblk);
  vodat_fd = dx_fileopen("JUNK.VOX",O_RDWR|O_BINARY);
  /*This block uses standard I/O functions */
  iott->io_type = IO_DEV|IO_CONT
  iott->io_fhandle = vodat_fd;
  iott->io_offset = 0;
  iott->io_length = 20000;
  /*This block uses my I/O functions */
  iott->io_fhandle = vodat_fd;
  iott->io_offset = 20001;
  iott->io_length = 20000;
  /*This block uses standard I/O functions */
  iott->io_fhandle = vodat_fd;
  iott->io_offset = 20002;
  iott->io_length = 20000;
  /*This block uses my I/O functions */
  iott->io_fhandle = vodat_fd;
  iott->io_offset = 10003;
  iott->io_length = 20000;
  devhandle = dx_open("dxxxB1C1", NULL);
  dx_sethook(devhandle, DX_ONHOOK,EV_SYNC)
  dx_wtring(devhandle,1,DX_OFFHOOK,EV_SYNC);
  dx_clrdigbuf;
  if(dx_rec(devhandle,iott,(DX_TPT*)NULL,RM_TONE|EV_SYNC) == -1) {
    perror("'");
    exit(1);
  }
writes count bytes from buffer into file associated with handle  \texttt{dx\_fwrite( )} \\

\begin{verbatim}
\texttt{dx\_clrdigbuf(devhandle);} \\
\texttt{if(dx\_play(devhandle,iott,(DX\_TPT*)EV\_SYNC) == -1 { \\
\texttt{perror(""}; \\
\texttt{exit(1);} \\
\texttt{}) \\
\texttt{dx\_close(devhandle);} \\
\end{verbatim}

\section*{Errors}

If this function returns -1 to indicate failure, errno is set to the following values:

- \texttt{EBADF}  
  - File handle is invalid or the file is not opened for writing
- \texttt{ENOSPC}  
  - Not enough space left on the device for the operation

\section*{See Also}

- \texttt{dx\_fileopen( )}
- \texttt{dx\_fileclose( )}
- \texttt{dx\_fileseek( )}
- \texttt{dx\_fileread( )}
**dx_getcursv( )** returns the specified channel’s current speed and volume adjustments active on a channel. For example, use **dx_getcursv( )** to determine the speed and volume level set interactively by a listener using DTMF digits during a play. (DTMF digits are set as play adjustment conditions using the **dx_setsvcond( )** function, or by one of the convenience functions **dx_addspddig( )** and **dx_addvoldig( )**)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained by a call to <strong>dx_open( )</strong>.</td>
</tr>
<tr>
<td>curvolp:</td>
<td>points to an integer that represents the current absolute volume setting for the channel. This value will lie between -30dB and +10dB.</td>
</tr>
<tr>
<td>curspeedp:</td>
<td>points to an integer that represents the current absolute speed setting for the channel. This value will be between -50% and +50%.</td>
</tr>
</tbody>
</table>

**Cautions**

None.
returns the specified channel's current speed \( \text{dx\_getcursv}(\) \)

Example

```
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

/*
 * Global Variables
 */
main()
{
  int dxxxdev;
  int curspeed, curvolume;

  /* Open the Voice Channel Device and Enable a Handler */
  if ( ( dxxxdev = dx_open( "dxxxB1C1", NULL) ) == -1 ) {
    perror( "dxxxB1C1" );
    exit( 1 );
  }

  /* Get the Current Volume and Speed Settings */
  if ( \( \text{dx\_getcursv}( \text{dxxxdev}, \&\text{curvolume}, \&\text{curspeed} ) \) == -1 ) {
    printf( "Unable to Get the Current Speed and\n" );
    printf( "Volume Settings:\n"");
    printf( "Lasterror = %d  Err Msg = %s\n", 
           ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
    dx_close( dxxxdev );
    exit( 1 );
  } else {
    printf( "Volume = %d   Speed = %d\n", curvolume, curspeed );
  }

  /* Continue Processing */

  /* Close the opened Voice Channel Device */
  if ( dx_close( dxxxdev ) != 0 ) {
    perror( "close" );
  }

  /* Terminate the Program */
  exit( 0 );
}
```
dx_getcursv( ) returns the specified channel's current speed

- Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR( ) and ATDV_ERRMSGP( ) to retrieve one of the following error reasons:

- EDX_BADPARM: Invalid Parameter
- EDX_BADPROD: Function not supported on this board
- EDX_SYSTEM: Windows NT system error - check errno

- See Also

Related to Speed and Volume:

- dx_adjsv( )
- dx_addspddig( )
- dx_addvoldig( )
- dx_setsvvm( )
- dx_getvvm( )
- dx_setsvcond( )
- dx_clrsvcond( )
- "Speed and Volume Modification Tables" (Voice Features Guide for Windows NT)
- DX_SVMT structure (Chapter 4. Voice Data Structures and Device Parameters)
**initiates the collection of digits**

**dx_getdig()**

Name: int dx_getdig(chdev, tpt, digitp, mode)

Inputs:
- int chdev: valid Dialogic channel device handle
- DV_TPT *tpt: pointer to Termination Parameter Table Structure
- DV_DIGIT *digitp: pointer to User Digit Buffer Structure
- unsigned short mode: asynchronous/synchronous setting

Returns: 0 to indicate successful initiation (asynchronous) number of digits (+1 for NULL) if successful (synchronous) -1 if failure

Includes: srllib.h dxxxlib.h

Category: I/O Mode: synchronous/asynchronous

---

**Description**

The **dx_getdig()** function initiates the collection of digits from an open channel’s digit buffer. Upon termination of the function, the collected digits are written in ASCII format into the local buffer, which is arranged as a **DV_DIGIT** structure.

The type of digits collected depends on the digit detection mode set by the **dx_setdigtyp()** function (for standard Voice board digits) or by the **dx_addtone()** function (for user-defined digits).

See the function descriptions for **dx_setdigtyp()**, **dx_addtone()** in this chapter and the description of the **DV_DIGIT** structure in Chapter 4, **Voice Data Structures and Device Parameters** for more information.

Set termination conditions using the DV_TPT structure. This structure is pointed to by the **tpt** parameter described below.

**Asynchronous Operation**

To run this function asynchronously set the **mode** field to EV_ASYNC. When running asynchronously, this function will return 0 to indicate it has initiated successfully, and will generate a termination event (see below) to indicate
*dx_getdig( )* 

**initiates the collection of digits**

Completion. Use the SRL Event Management functions to handle the termination event. See Appendix A for more information about the Event Management functions.

Termination of asynchronous digit collection is indicated by a TDX_GETDIG event. After *dx_getdig( )* terminates, use the ATDX_TERMMSK() function to determine the reason for termination.

---

**Synchronous Operation**

By default, this function runs synchronously. Termination of synchronous digit collection is indicated by a return value greater than 0 that represents the number of digits received (+1 for NULL). Use ATDX_TERMMSK() to determine the reason for termination.

The function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>chdev:</strong></td>
<td>specifies the valid channel device handle obtained when the channel was opened using <em>dx_open( )</em></td>
</tr>
<tr>
<td><strong>tptp:</strong></td>
<td>points to the Termination Parameter Table Structure, DV_TPT, which specifies termination conditions for this function. See Appendix A for more information about the DV_TPT. Termination conditions are listed below:</td>
</tr>
<tr>
<td>DX_DIGTYPE</td>
<td>User-defined digits</td>
</tr>
<tr>
<td>DX_MAXDTMF</td>
<td>Maximum number of digits received</td>
</tr>
<tr>
<td>DX_MAXSIL</td>
<td>Maximum silence</td>
</tr>
<tr>
<td>DX_MAXNOSIL</td>
<td>Maximum non-silence</td>
</tr>
<tr>
<td>DX_LCOFF</td>
<td>Loop current off</td>
</tr>
<tr>
<td>DX_IDDTIME</td>
<td>Inter-digit delay</td>
</tr>
<tr>
<td>DX_MAXTIME</td>
<td>Function time</td>
</tr>
<tr>
<td>DX_DIGMASK</td>
<td>Digit mask termination</td>
</tr>
<tr>
<td>DX_PMOFF</td>
<td>Pattern match silence off</td>
</tr>
<tr>
<td>DX_PMON</td>
<td>Pattern match silence on</td>
</tr>
<tr>
<td>DX_TONE</td>
<td>Tone-off or Tone-on detection</td>
</tr>
</tbody>
</table>

**digitp:** points to the User’s Digit Buffer Structure, where collected
**initiates the collection of digits**  
*dx_getdig()*

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>digits and their types are stored in arrays. The digit types in <strong>DV_Digit</strong> can be one of the following:</td>
<td></td>
</tr>
<tr>
<td>DG_DTMF</td>
<td>• DTMF digit</td>
</tr>
<tr>
<td>DG_LPD</td>
<td>• Loop Pulse digit</td>
</tr>
<tr>
<td>DG_MF</td>
<td>• MF digit</td>
</tr>
<tr>
<td>DG_USER1</td>
<td>• User-defined digit</td>
</tr>
<tr>
<td>DG_USER2</td>
<td>• User-defined digit</td>
</tr>
<tr>
<td>DG_USER3</td>
<td>• User-defined digit</td>
</tr>
<tr>
<td>DG_USER4</td>
<td>• User-defined digit</td>
</tr>
<tr>
<td>DG_USER5</td>
<td>• User-defined digit</td>
</tr>
</tbody>
</table>

See Chapter 4. Voice Data Structures and Device Parameters for information about the **DV_Digit** structure.

See `dx_addtone()` for information about creating user-defined digits.

**mode:**
specifies whether to run `dx_getdig()` asynchronously or synchronously. Specify one of the following:

- **EV_ASYNC:** Run `dx_getdig()` asynchronously.
- **EV_SYNC:** Run `dx_getdig()` synchronously (default).

The channel’s digit buffer contains up to 31 digits, collected on a First-In First-Out (FIFO) basis. Since the digits remain in the channel’s digit buffer until they are overwritten or cleared using `dx_clrdigbuf()`, the digits in the channel’s buffer may have been received prior to this function call. **DG_MAXDIGS** is the define for the maximum number of digits that can be returned by a single call to `dx_getdig()`.

**NOTE:** By default, after the 31st digit, all subsequent digits will be discarded. You can use the `dx_setdigbuf()` function with the mode parameter set to **DX_DIGCYCLIC**, which will cause all incoming digits to overwrite the oldest digit in the buffer. See the `dx_setdigbuf()` function.

### Cautions

1. Some MF digits use approximately the same frequencies as DTMF digits (see Appendix C). Because there is a frequency overlap, if you have the incorrect kind of detection enabled, MF digits may be mistaken for DTMF digits, and vice versa. To ensure that digits are correctly detected,
**dx_getdig()** initiates the collection of digits

only one kind of detection should be enabled at any time. To set MF digit detection, use the **dx_setdigtyp()** function.

2. A digit that is set to adjust play-speed or play-volume (using **dx_setsvcond()**) will not be passed to **dx_getdig()**, and will not be used as a terminating condition. If a digit is defined to adjust play and to terminate play, then the play adjustment will take priority.

3. When operating asynchronously, ensure that the digit buffer stays in scope for the duration of the function.

4. The channel must be idle, or the function will return an EDX_BUSY error.

5. Speed and volume control are supported on the D/21D, D/21E, D/41D, D/41E, D/41ESC, D/81A, D/121B, D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1 and D/320SC boards only. Do not use the Speed and Volume control functions to control speed on the D/120, D/121, or D/121A boards.

6. If the function is operating synchronously and there are no digits in the buffer, the return value from this function will be 1, which indicates the NULL terminator.

---

**Example 1: Using dx_getdig() in synchronous mode**

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    DV_TPT tpt[3];
    DV_DIGIT digp;
    int chdev, numdigs, cnt;
    /* open the channel with dx_open(). Obtain channel device descriptor * in chdev */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
        /* process error */
    }
    /* initiate the call */
    /* Set up the DV_TPT and get the digits */
    dx_clrpt(tpt,3);
    tpt[0].tp_type  = IO_CONT; /* Maximum number of digits */
    tpt[0].tp_termno = DX_MAXDTMF; /* Maximum number of digits */
    tpt[0].tp_length = 4; /* terminate on 4 digits */
    tpt[0].tp_flags  = TP_MAXDTMF; /* terminate if already in buf. */
    tpt[1].tp_type  = IO_CONT; /* LC off termination */
    tpt[1].tp_termno = DX_LCOFF; /* LC off termination */
    tpt[1].tp_length = 3; /* Use 30 ms (10 ms resolution */
    tpt[1].tp_length = 3;
    /* Use 30 ms (10 ms resolution */
```
Example 2: Using `dx_getdig()` in asynchronous mode

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
#define MAXCHAN 24
int digit_handler();
DV_TPT stpt[3];
DV_DIGIT disp[256];
main()
{
    int i, chdev[MAXCHAN];
    char *chnamep;
    int srlmode;
    /* Set SRL to run in polled mode. */
    srlmode = SR_POLLMODE;
    if (sr_setparm(SRL_DEVICE, SR_MODEID, (void *)&srlmode) == -1) {
        /* process error */
    }
    for (i=0; i<MAXCHAN; i++) {
        /* Set chnamep to the channel name - e.g., dxxxB1C1 */
        /* open the channel with `dx_open()` . Obtain channel device */
        /* descriptor in chdev[i] */
        if ((chdev[i] = dx_open(chnamep, NULL)) == -1) {
            /* process error */
        }
        /* Using `sr_enbhdlr()` , set up handler function to handle `dx_getdig()` */
        /* completion events on this channel. */
        if (sr_enbhdlr(chdev[i], TDX_GETDIG, digit_handler) == -1) {
            /* process error */
        }
        /* initiate the call */
        /* Set up the DV_TPT and get the digits */
    }
}
```
dx_getdig() initiates the collection of digits

dx_clrtpt(tpt,3);
tpt[0].tp_type = IO_CONT;
tpt[0].tp_termno = DX_MAXDTMF; /* Maximum number of digits */
tpt[0].tp_length = 4; /* terminate on 4 digits */
tpt[0].tp_flags = TF_MAXDTMF; /* terminate if already in buf*/
tpt[1].tp_type = IO_CONT;
tpt[1].tp_termno = DX_LCOFF; /* LC off termination */
tpt[1].tp_length = 3; /* Use 30 ms (10 ms resolution */
   * timer */
tpt[1].tp_flags = TF_LCOFF|TF_10MS; /* level triggered, clear
   * history, 10 ms resolution */
tpt[2].tp_type = IO_EOT;
tpt[2].tp_termno = DX_MAXTIME; /* Function Time */
tpt[2].tp_length = 100; /* 10 seconds (100 ms resolution */
   * timer */
tpt[2].tp_flags = TF_MAXTIME; /* Edge triggered */
if (dx_clrdigbuf(chdev[i]) == -1) {
    /* process error */
}
if (dx_getdig(chdev[i], tpt, &digp[chdev[i]], EV_ASYNC) == -1) {
    /* process error */
}
/* Use sr_waitevt() to wait for the completion of dx_getdig().
 * On receiving the completion event, TDX_GETDIG, control is transferred
 * to the handler function previously established using sr_enbhdlr().
 */
int digit_handler()
{
    int chfd;
    int cnt, numdigs;
    chfd = sr_getevtdev();
    numdigs = strlen(digp[chfd].dg_value);
    for(cnt=0; cnt < numdigs; cnt++) {
        printf("Digit received = %c, digit type = %d",
               digp[chfd].dg_value[cnt], digp[chfd].dg_type[cnt]);
    }
    /* Kick off next function in the state machine model. */
    .
    return 0;
}

Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR() and
ATDV_ERRMSGP() to retrieve one of the following error reasons:

EDX_BADPARM          •  Invalid Parameter
EDX_BADTPT           •  Invalid DV_TPT entry
EDX_BUSY             •  Channel busy
EDX_SYSTEM           •  Windows NT system error - check errno

206-CD
initiates the collection of digits  \hspace*{1.5em} dx_getdig()

- See Also

**Setting User-Defined Digits:**
- dx_addtone( )
- dx_setdigtyp( )

**Collecting Digits:**
- \textit{DV\_DIGIT}
- dx_sethook( )
The `dx_getevt()` function is used to synchronously monitor channels for possible call status transition events in conjunction with `dx_setevtmsk()`, `dx_getevt()` blocks and returns control to the program after one of the events set by `dx_setevtmsk()` occurs on the channel specified in the `chdev` parameter. The `DX_EBLK` structure contains the event that ended the blocking.

The function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <code>dx_open()</code></td>
</tr>
<tr>
<td>eblkp:</td>
<td>points to the Event Block Structure <code>DX_EBLK</code>, which will contain the event that ended the blocking.</td>
</tr>
<tr>
<td>timeout:</td>
<td>specifies the maximum amount of time in seconds to wait for an event to occur. timeout can have one of the following values:</td>
</tr>
<tr>
<td></td>
<td># of seconds: maximum length of time <code>dx_getevt()</code> will wait for an event. When the time specified has elapsed, the function will terminate and return an error.</td>
</tr>
<tr>
<td></td>
<td>-1: <code>dx_getevt()</code> will block until an event occurs; it will not time out.</td>
</tr>
<tr>
<td></td>
<td>0: The function will return -1 immediately if no event is present.</td>
</tr>
</tbody>
</table>

```c
int dx_getevt(chdev,eblkp,timeout)
```
used to synchronously monitor channels

dx_getevt()

### Parameter Description

**NOTE:** When the time specified in **timeout** expires, **dx_getevt()** will terminate and return an error. The Standard Attribute function **ATDV_LASTERR()** can be used to determine the cause of the error, which in this case is EDX_TIMEOUT.

### Cautions

We recommend enabling only one process per channel. The event that **dx_getevt()** is waiting for may change if another process sets a different event for that channel. See **dx_setevtmnk()** for more details.

### Example

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
  int chdev;         /* channel descriptor */
  int timeout;       /* timeout for function */
  DX_EBLK eblk;      /* Event Block Structure */

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

  /* Set RINGS or WINK as events to wait on */
  if (dx_setevtmnk(chdev,DM_RINGS|DM_WINK) == -1) {
    /* process error */
  }

  /* Set timeout to 5 seconds */
  timeout = 5;
  if (dx_getevt(chdev, &eblk, timeout) == -1) {
    /* process error */
    if (ATDV_LASTERR(chdev) == EDX_TIMEOUT) {    /* check if timed out */
      printf("Timed out waiting for event.\n");
    } else {
      /* further error processing */
    }
  }

  switch (eblk.ev_event) {
    case DE_RINGS:
      printf("Ring event occurred.\n");
      break;
    case DE_WINK:
      printf("Wink event occurred.\n");
      break;
  }
}
```


`dx_getevt( )` used to synchronously monitor channels

```c
    printf("Wink event occurred.\n");
    break;
}
```

## Errors

If this function returns -1 to indicate failure, use `ATDV_LASTERR( )` and `ATDV_ERRMSGP( )` to retrieve one of the following error reasons:

- **EDX_BADPARM**  •  Invalid Parameter
- **EDX_SYSTEM**   •  Windows NT system error - check `errno`
- **EDX_TIMEOUT**  •  Timeout time limit is reached

## See Also

- `dx_setevtsk( )`
- `DX_EBLK (Chapter 4. Voice Data Structures and Device Parameters)`

210-CD
obtains the current parameter settings  

---

**Name:** int dx_getparm(dev,parm,valuep)

**Inputs:**
- int dev • valid Dialogic channel or board device handle
- unsigned long parm • parameter type to get value of
- void *valuep • pointer to variable for returning parameter value

**Returns:**
- 0 if success
- -1 if failure

**Includes:**
- srllib.h
- dxxxlib.h

**Category:** Configuration

---

### Description

The `dx_getparm()` function obtains the current parameter settings for an open device. `dx_getparm()` can only obtain the value of one parameter at a time. The channel must be idle (i.e., no I/O function running) when calling `dx_getparm()`.

The function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dev</td>
<td>specifies the valid Dialogic device handle obtained when a board or channel was opened using <code>dx_open()</code>.</td>
</tr>
</tbody>
</table>
| parm      | specifies the define for the parameter type whose value is to be returned in the variable pointed to by `valuep`.  
Board and channel parameter defines, defaults and descriptions are listed in Section 5.2. Clearing Voice Structures |
| valuep    | points to the variable where the value of the parameter specified in `parm` should be returned.  
**NOTE:** You must use a void* cast on the returned parameter value, as demonstrated in the example that follows. |

`valuep` should point to a variable large enough to hold the value of the parameter. Refer to `dxxxlib.h` for parameters sizes. The size of a parameter is encoded in the define for the parameter. The defines for parameter sizes are PM_SHORT, PM_BYTE, PM_INT, PM_LONG,
**dx_getparm( )**

obtains the current parameter settings

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>PM_FLSTR (fixed length string), and PM_VLSTR (variable length string)</td>
<td>See dxxxlib.h for DXBD_ and DXCH_defines.</td>
</tr>
</tbody>
</table>

Most parameters are of type short.

### Cautions

We highly recommend that you clear the variable the parameter value is returned to prior to calling `dx_getparm( )`, as illustrated in the example below. The variable whose address is passed to should be of a size sufficient to hold the value of the parameter. See the description section of this function for more information.

### Example

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    int bddev;
    unsigned short parmval;

    /* open the board using dx_open( ). Obtain board device descriptor in * bddev */
    if ((bddev = dx_open("dxxxB1",NULL)) == -1) {
        /* process error */
    }

    parmval = 0; /* CLEAR parmval */

    /* get the number of channels on the board. DXBD_CHNUM is of type * unsigned short as specified by the PM_SHORT define in the definition * for DXBD_CHNUM in dxxxlib.h. The size of the variable parmval is * sufficient to hold the value of DXBD_CHNUM. */
    if (dx_getparm(bddev, DXBD_CHNUM, (void *)&parmval) == -1) {
        /* process error */
    }

    printf("\nNumber of channels on board = %d",parmval);
    .
}

212-CD
obtains the current parameter settings

dx_getparm()

Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR() and ATDV_ERRMSGP() to retrieve one of the following error reasons:

- EDX_BADPARM: Invalid Parameter
- EDX_SYSTEM: Windows NT system error - check errno
- EDX_BUSY: Channel is busy (when channel device handle is specified) or first channel is busy (when board device handle is specified)

See Also

- dx_setparm()
dx_getsvmt() returns contents of Speed or Volume Modification Table

**Name:** int dx_getsvmt(chdev, tabletype, svmtp )

**Inputs:**
- int chdev
  - valid Dialogic channel device handle
- unsigned short tabletype
  - table to retrieve (speed or volume)
- DX_SVMT * svmtp
  - pointer to DX_SVMT structure

**Returns:**
- 0 if success
- -1 if failure

**Includes:**
srllib.h
dxxxlib.h

**Category:** Speed and Volume

**Description**

The `dx_getsvmt()` function returns contents of Speed or Volume Modification Table to the DX_SVMT structure.

For a full description of the Speed and Volume Modification Tables see the *Voice Features Guide for Windows NT* see a description in section 4.1.6. DX_SVMT - speed/volume modification table structure of the DX_SVMT structure.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained by a call to <code>dx_open()</code> .</td>
</tr>
<tr>
<td>tabletype:</td>
<td>specifies whether to retrieve the Speed or the Volume Modification Table.</td>
</tr>
<tr>
<td></td>
<td>SV_SPEEDTBL Retrieve the Speed Modification Table values</td>
</tr>
<tr>
<td></td>
<td>SV_VOLUMETBL Retrieve the Volume Modification Table values</td>
</tr>
<tr>
<td>svmtp:</td>
<td>points to the DX_SVMT structure that contains the Speed/Volume Modification Table entries.</td>
</tr>
</tbody>
</table>

**Cautions**

None.

214-CD
returns contents of Speed or Volume Modification Table

```
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

/*
* Global Variables
*/

main()
{
  DX_SVMT svmt;
  int  dxxxdev, index;

  /*
  * Open the Voice Channel Device and Enable a Handler
  */
  if ( ( dxxxdev = dx_open( "dxxxB1C1", NULL ) ) == -1 ) {
    perror( "dxxxB1C1" );
    exit( 1 );
  }

  /*
  * Get the Current Volume Modification Table
  */
  memset( &svmt, 0, sizeof( DX_SVMT ) );
  if ( dx_getsvmt( dxxxdev, SV_VOLUMETBL, &svmt ) == -1 ){
    printf( "Unable to Get the Current Volume"
    );
    printf( " Modification Table\n"
    );
    printf( " Lasterror = %d  Err Msg = %s\n",
        ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
    dx_close( dxxxdev );
    exit( 1 );
  }

  else {
    printf( "Volume Modification Table is:\n"
    );
    for ( index = 0; index < 10; index++ ) {
      printf( "decrease\[ %d \] = %d\n",
                index, svmt.decrease[ index ] );
    }

    printf( "origin = %d\n",
            svmt.origin );

    for ( index = 0; index < 10; index++ ) {
      printf( "increase\[ %d \] = %d\n",
                index, svmt.increase[ index ] );
    }
  }
}

/*
* Continue Processing
*/
```

215-CD
dx_getsvmt( ) returns contents of Speed or Volume Modification Table

* Close the opened Voice Channel Device
* if ( dx_close( dxxxdev ) != 0 ) {
    perror( "close" );
}

/* Terminate the Program */
exit( 0 );

Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR( ) and ATDV_ERRMSGP( ) to retrieve one of the following error reasons:

- EDX_BADPARM • Invalid Parameter
- EDX_BADPROD • Function not supported on this board
- EDX_SPDVOL • Must Specify either SV_SPEEDTBL or SV_VOLUMETBL
- EDX_SYSTEM • Windows NT system error - check errno

See Also

- dx_addspddig( )
- dx_addvoldig( )
- dx_adjsv( )
- dx_clrsvcond( )
- dx_getcursv( )
- dx_setsvcond( )
- dx_setsvmt( )
- "Speed and Volume Modification Tables" (Voice Features Guide for Windows NT)
- DX_SVMT (Chapter 4. Voice Data Structures and Device Parameters)
The `dx_initcallp()` function initializes and activates PerfectCall Call Analysis on the channel identified by `chdev`. In addition, this function adds all tones used in Call Analysis to the channel’s Global Tone Detection (GTD) templates.

To use PerfectCall Call Analysis, `dx_initcallp()` must be called prior to using `dx_dial()` on the specified channel. If `dx_dial()` is called before initializing the channel with `dx_initcallp()`, then Call Analysis will operate in Basic mode only for that channel.

PerfectCall Call Analysis allows the application to detect three different types of dial tone, two busy signals, ringback, and two fax or modem tones on the channel. It is also capable of distinguishing between a live voice and an answering machine when a call is connected. Parameters for these capabilities are downloaded to the channel when `dx_initcallp()` is called.

The Voice Driver comes equipped with useful default definitions for each of the signals mentioned above. The application can change these definitions through the `dx_chgdur()`, `dx_chgfreq()`, and `dx_chgrepcnt()` functions. The `dx_initcallp()` function takes whatever definitions are currently in force and uses these definitions to initialize the specified channel.

Once a channel is initialized with the current tone definitions, these definitions cannot be changed for that channel without deleting all tones (`dx_deltones()` ) and re-initializing with another call to `dx_initcallp()`. `dx_deltones` also disables PerfectCall Call Analysis. Note, however, that `dx_deltones()` will erase all user-defined tones from the channel (including any Global Tone Detection information), and not just the PerfectCall Call Analysis tones.
dx_initcallp() initializes and activates PerfectCall Call Analysis

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the channel device handle.</td>
</tr>
</tbody>
</table>

**Example**

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

main()
{
    DX_CAP cap_s;
    int ddd, car;
    char *chnam, *dialstrg;

    chnam = "dxxxB1C1";
    dialstrg = "l1234";

    /*
     * Open channel
     */
    if ((ddd = dx_open( chnam, NULL )) == -1 ) {
        /* handle error */
    }

    /*
     * Delete any previous tones
     */
    if ( dx_deltones(ddd) < 0 ) {
        /* handle error */
    }

    /*
     * Change Enhanced call progress default local dial tone
     */
    if (dx_chgfreq( TID_DIAL_LCL, 425, 150, 0, 0 ) < 0) {
        /* handle error */
    }

    /*
     * Change Enhanced call progress default busy cadence
     */
    if (dx_chgdur( TID_BUSY1, 550, 400, 550, 400 ) < 0) {
        /* handle error */
    }

    if (dx_chgrepcnt( TID_BUSY1, 4 ) < 0) {
        /* handle error */
    }

    /*
     * Now enable Enhanced call progress with above changed settings.
     */
```

218-CD
initializes and activates PerfectCall Call Analysis  

`dx_initcallp()`

```c
if (dx_initcallp( ddd )) {  
    /* handle error */
}

/*
   * Set off Hook
   */
if ((dx_sethook( ddd, DX_OFFHOOK, EV_SYNC )) == -1) {  
    /* handle error */
}

/*
   * Dial
   */
if ((car = dx_dial( ddd, dialstrg,(DX_CAP *)&cap_s, DX_CALLP|EV_SYNC))==-1) {  
    /* handle error */
}

switch( car ) {  
    case CR_NODIALTONE:  
        printf(" Unable to get dial tone\n");  
        break;

    case CR_BUSY:  
        printf(" %s engaged\n", dialstrg );  
        break;

    case CR_CNCT:  
        printf(" Successful connection to %s\n", dialstrg );  
        break;

    default:  
        break;
}

/*
   * Set on Hook
   */
if ((dx_sethook( ddd, DX_ONHOOK, EV_SYNC )) == -1) {  
    /* handle error */
}

dx_close( ddd );
```

### Cautions

The channel must be idle.

### See Also

- `dx_chgdur()`
- `dx_chgfreq()`
- `dx_chgerepcnt()`
dx_initcallp() initializes and activates PerfectCall Call Analysis

- dx_deltones()
The `dx_open()` function opens a Voice device and returns a unique Dialogic device handle to identify the device. All subsequent references to the opened device must be made using the handle until the device is closed. A device can be opened more than once by any number of processes.

**NOTE:** The device handle returned by this function is **Dialogic defined**. It is not a standard Windows NT file descriptor. Any attempts to use Windows NT operating system commands such as `read()`, `write()`, or `ioctl()` will produce unexpected results.

In applications that spawn child processes off a parent process, the device handle is not inheritable by the child process. Make sure devices are opened in the child process.

The function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>namep:</td>
<td>points to an ASCIIZ string that contains the name of the valid Dialogic device. These valid devices can be either boards or channels.</td>
</tr>
<tr>
<td>oflags:</td>
<td>is reserved for future use. This parameter should be set to NULL.</td>
</tr>
</tbody>
</table>

**Cautions**

Do not use the Windows NT `open()` function to open a Voice device. Unpredictable results will occur.
dx_open( )

**opens a Voice device**

### Example

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

main()
{
    int chdev;    /* channel descriptor */
    .
    .
    /* Open Channel */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
        /* process error */
    .
    .
    }
}
```

### Errors

If this function returns -1 to indicate failure, check errno for one of the following reasons:

- **EINVAL** • Invalid Argument
- **EBADF** • Invalid file descriptor
- **EINTR** • A signal was caught
- **EIO** • Error during a Windows NT STREAMS open

### See Also

- **dx_close( )**
**Name:** int dx_play(chdev, iottp, tptp, mode)

**Inputs:**

- int chdev • valid Dialogic channel device handle
- DX_IOTT *iottp • pointer to I/O Transfer Table Structure
- DV_TPT *tptp • pointer to Termination Parameter Table Structure
- unsigned short mode • asynchronous/synchronous playing mode bit mask for this play session

**Returns:**

- 0 if success
- -1 if failure

**Includes:**

- srllib.h
- dxxxlib.h

**Category:** I/O

**Mode:** synchronous/asynchronous

---

### Description

The `dx_play()` function plays recorded voice data or transfers Analog Display Services Interface (ADSI) data on a specified channel. The voice data may come from any combination of data files, memory, or custom devices.

The order of play and the location of the voice data is specified in the `DX_IOTT` structure pointed to by `iottp`. The `DX_IOTT` structure is described in *Chapter 4, Voice Data Structures and Device Parameters*.

**NOTE:** For a single file synchronous play, `dx_playf()` is more convenient because you do not have to set up a DX_IOTT structure. See the `dx_playf()` function description for more information.

### Asynchronous Operation

To run this function asynchronously set the `mode` field to EV_ASYNC. When running asynchronously, this function will return 0 to indicate it has initiated successfully, and will generate a termination event (see below) to indicate completion.
*dx_play( )* plays recorded voice data

Termination conditions for play are set using the DV_TPT structure. Play continues until all data specified in *DX_IOTT* has been played, or until one of the conditions specified in DV_TPT is satisfied.

When *dx_play( )* terminates, the current channel’s status information, including the reason for termination, can be accessed using Extended Attribute functions.

Termination of asynchronous play is indicated by a TDX_PLAY event.

After *dx_play( )* terminates, use the *ATDX_TERMMSK( )* function to determine the reason for termination.

Use the SRL Event Management functions to handle the termination event. See Appendix A for more information about the Event Management functions.

**NOTE:** The DX_IOTT structure must remain in scope for the duration of the function if running asynchronously.

### Synchronous Operation

By default, this function runs synchronously, and will return a 0 to indicate that it has completed successfully.

Termination conditions for play are set using the DV_TPT structure. Play continues until all data specified in *DX_IOTT* has been played, or until one of the conditions specified in DV_TPT is satisfied.

Termination of synchronous play is indicated by a return value of 0. After *dx_play( )* terminates, use the *ATDX_TERMMSK( )* function to determine the reason for termination.

### Analog Display Services Interface (ADSI) Protocol

The Analog Display Services Interface (ADSI) protocol is used to transmit data to a display-based telephone that is connected to an analog loop start line. An ADSI alert tone is used to verify that Dialogic hardware is connected to an ADSI telephone and to alert the telephone that ADSI data will be transferred.

**NOTE:** Check with your telephone manufacturer to verify that your telephone is a true ADSI-compliant device.
Each time a new call is initiated on a channel, send the alert tone to alert the telephone that ADSI data will be transferred.

The ADSI alert tone can be sent and acknowledged, and ADSI data can be transferred using the `dx_setparm()` and `dx_play()` or `dx_playf()` functions. This is accomplished by setting the voice channel parameter DXCH_DTINITSET to DM_A in the `dx_setparm()` function and executing the `dx_play()` or `dx_playf()` function with the PM_ADSIALERT define ORed in the `mode` parameter.

If the acknowledgment digit is not received from the telephone within 500 ms following the end of the alert tone, the function will return a 0 but the termination mask returned by `ATDX_TERMMSK()` will be TM_MAXTIME to indicate an ADSI protocol error.

**NOTE:** The function will return a -1 if a failure is due to a general play error.

If the handshaking and transmission are successful, the function terminates normally with a TM_EOD (End of data reached on playback) termination mask returned by `ATDX_TERMMSK()` to indicate that the operation is complete.

To transfer ADSI data without an alert tone, use the `dx_clrdigbuf()` or `dx_getdig()` function to ensure that there are no pending digits. Transfer ADSI data using the `dx_play()` or `dx_playf()` function with the PM_ADSI define ORed in the `mode` parameter.

If the transmission is successful, the function terminates normally with a TM_EOD (End of data reached on playback) termination mask returned by `ATDX_TERMMSK()` to indicate that the operation is complete.

The application is responsible for determining whether the message count acknowledgement matches the number of messages that were transmitted and for retransmitting any messages. Use the `dx_getdig()` function with DV_TPT `tp_termno` set to DX_DIGTYPE to receive the DTMF string 'adx' where 'x' is the message count acknowledgement digit (1 - 5).

**NOTE:** The ADSI data must conform to interface requirements described in Bellcore Technical Reference TR-NWT-000030, *Voiceband Data Transmission Interface Generic Requirements*.

For information about message requirements (how the data should be displayed on the Customer Premise Equipment), see Bellcore Technical Reference TR-NWT-001273, *Generic Requirements for an SPCS to...*
The \texttt{dx\_play()} function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>\texttt{chdev:}</td>
<td>specifies the valid channel device handle obtained when the channel was opened using \texttt{dx_open()}.</td>
</tr>
<tr>
<td>\texttt{iottp:}</td>
<td>points to the I/O Transfer Table Structure, \texttt{DX_IOTT}, which sets the order in which and media from which the voice data will be played. See 5.1. Always Check Return Code in Voice Programming for information about the \texttt{DX_IOTT} structure.</td>
</tr>
<tr>
<td>\texttt{tptp:}</td>
<td>points to the Termination Parameter Table Structure, \texttt{DV_TPT}, which specifies termination conditions for \texttt{dx_play()} playing. Valid \texttt{DV_TPT} terminating conditions are listed below:</td>
</tr>
<tr>
<td>\texttt{DX_DIGTYPE}</td>
<td>User-defined digit occurred</td>
</tr>
<tr>
<td>\texttt{DX_MAXDTMF}</td>
<td>Maximum number of digits received</td>
</tr>
<tr>
<td>\texttt{DX_MAXSIL}</td>
<td>Maximum silence</td>
</tr>
<tr>
<td>\texttt{DX_MAXNOSIL}</td>
<td>Maximum non-silence</td>
</tr>
<tr>
<td>\texttt{DX_LCOFF}</td>
<td>Loop current off</td>
</tr>
<tr>
<td>\texttt{DX_IDDTIME}</td>
<td>Inter-digit delay</td>
</tr>
<tr>
<td>\texttt{DX_MAXTIME}</td>
<td>Function time</td>
</tr>
<tr>
<td>\texttt{DX_DIGMASK}</td>
<td>Digit mask termination</td>
</tr>
<tr>
<td>\texttt{DX_PMOFF}</td>
<td>Pattern match silence off</td>
</tr>
<tr>
<td>\texttt{DX_PMON}</td>
<td>Pattern match silence on</td>
</tr>
<tr>
<td>\texttt{DX_TONE}</td>
<td>Tone-off or Tone-on detection</td>
</tr>
</tbody>
</table>

See Appendix A, which describes the Standard Runtime

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plays recorded voice data  

\textit{dx\_play( )}

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Library, for information about this structure.</td>
</tr>
</tbody>
</table>

**NOTE:** In addition to DV\_TPT terminations, the function can fail due to maximum byte count, \texttt{dx\_stopch( )}, or end of file. See \texttt{ATDX\_TERMMSK( )} for a full list of termination reasons.

**mode:** defines the play mode and asynchronous/synchronous mode. One or more of the play mode parameters listed below may be selected in the bit mask for play mode combinations (see Table 4).

Choose one only:

- **EV\_ASYNC:** Run \texttt{dx\_play( )} asynchronously.
- **EV\_SYNC:** Run \texttt{dx\_play( )} synchronously (default).

Choose one or more:

- **MD\_ADPCM:** Play using Adaptive Differential Pulse Code Modulation encoding algorithm (4 bits per sample). Playing with ADPCM is the default setting.
- **MD\_PCM:** Play using Pulse Code Modulation encoding algorithm (8 bits per sample).
- **PM\_ALAW:** Play using A-Law.
- **PM\_TONE:** Transmit a tone before initiating play. If this mode is not selected, no tone will be transmitted. No tone transmitted is the default setting.
- **PM\_SR6:** Play using 6KHz sampling rate (6,000 samples per second).
- **PM\_SR8:** Play using 8KHz sampling rate (8,000 samples per second).
parameter. PM_ADSIALERT should be ORed with the EV_SYNC or EV_ASYNC parameter in the mode parameter.

PM_ADSI: Play using the ADSI protocol without an alert tone preceding play. If ADSI protocol mode is selected, it is not necessary to select any other play mode parameters. If ADSI data will be transferred, PM_ADSI should be ORed with the EV_SYNC or EV_ASYNC parameter in the mode parameter.

NOTES: 1. The rate specified in the last play function will apply to the next play function, unless the rate was changed in the parameter DXCH_PLAYDRATE using dx_setparm( ).

2. Specifying PM_SR6 or PM_SR8 using dx_play() changes the setting of the parameter DXCH_PLAYDRATE. DXCH_PLAYDRATE can also be set and queried using dx_setparm() and dx_getparm(). The default setting for DXCH_PLAYDRATE is 6KHz.

3. Make sure data is played using the same encoding algorithm and sampling rate used when the data was recorded.

4. MD_PCM can be used on D/12x or D/81A board.

5. The D/21E, D/41E, D/41ESC, D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1 and D/320SC boards enable the user to select either A-Law or mu-Law encoding of data. The default on the board is set to mu-Law and returns to mu-Law after each play. The A-Law parameters must be passed each time the play function is called. Enable A-Law playback by OR’ing the new play mode,
Table 4 shows play mode selections when transmitting or not transmitting a tone before initiating play. The first column of the table lists the two play features (tone or no tone), and the first row lists each type of encoding algorithm (ADPCM or PCM) and data-storage rate for each algorithm/sampling rate combination in parenthesis (24 Kbps, 32 Kbps, 48 Kbps, or 64 Kbps).

Select the desired play feature in the first column of the table and look across that row until the column containing the desired encoding algorithm and data-storage rate is reached. The play modes that must be entered in the mode bit mask are provided where the feature row and encoding algorithm/data-storage rate column intersect. Parameters listed in { } are default settings and do not have to be specified.

NOTE: If PM_ADSI play mode is selected (not shown in Table 4), the ADSI protocol will be used to transfer ADSI data and it is not necessary to select any other play mode parameters. PM_ADSI should be ORed with the EV_SYNC or EV_ASYNC parameter in the mode parameter.

### Table 4. Play Mode Selections

<table>
<thead>
<tr>
<th>Feature(s)</th>
<th>ADPCM (24 Kbps)</th>
<th>ADPCM (32 Kbps)</th>
<th>PCM (48 Kbps)</th>
<th>PCM (64 Kbps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>• Tone</td>
<td>PM_TONE</td>
<td>PM_TONE</td>
<td>PM_TONE</td>
<td>PM_TONE</td>
</tr>
<tr>
<td></td>
<td>PM_SR6</td>
<td>PM_SR8</td>
<td>PM_ALAW*</td>
<td>PM_SR6</td>
</tr>
<tr>
<td></td>
<td>{ MD_ADPCM     }</td>
<td>{ MD_ADPCM     }</td>
<td>MD_PCM</td>
<td>MD_PCM</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>• No Tone</td>
<td>PM_SR6</td>
<td>PM_SR8</td>
<td>PM_SR6</td>
<td>PM_SR8</td>
</tr>
<tr>
<td></td>
<td>{ MD_ADPCM     }</td>
<td>{ MD_ADPCM     }</td>
<td>MD_PCM</td>
<td>MD_PCM</td>
</tr>
</tbody>
</table>

{ } = Default modes.
* = Select if file was encoded using A-Law (supported by D/41ESC, D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1 and D/320SC boards only).

NOTE: `dx_play()` will run synchronously if you do not specify EV_ASYNC, or if
**dx_play( )**

*plays recorded voice data*

you specify EV_SYNC (default).

---

**Cautions**

Whenever **dx_play( )** is called, its speed and volume is based on the most recent adjustment made using **dx_adjsv( )** or **dx_setsvcond( )**.

---

**Example 1: Using dx_play( ) in synchronous mode.**

```c
/* Play a voice file. Terminate on receiving 4 digits or at end of file*/
#include <fcntl.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
main()
{
    int chdev;
    DX_IOTT iott;
    DV_TPT tpt;
    DV_DIGIT dig;
    //
    /* Open the device using dx_open( ). Get channel device descriptor in
     * chdev. */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
        /* process error */
    }
    /* set up DX_IOTT */
    iott.io_type = IO_DEV|IO_EOT;
    iott.io_bufp = 0;
    iott.io_offset = 0;
    iott.io_length = -1; /* play till end of file */
    if((iott.io_fhandle = dx_fileopen("prompt.vox", O_RDONLY|O_BINARY))
        == -1) {
        /* process error */
    }
    /* set up DV_TPT */
    dx_clrtpt(&tpt,1);
    tpt.tp_type = IO_EOT; /* only entry in the table */
    tpt.tp_termno = DX_MAXDTMF; /* Maximum digits */
    tpt.tp_length = 4; /* terminate on four digits */
    tpt.tp_flags = TF_MAXDTMF; /* Use the default flags */
    /* clear previously entered digits */
    if (dx_clrdigbuf(chdev) == -1) { /* process error */
    }
    /* Now play the file */
    if (dx_play(chdev,&iott,&tpt,EV_SYNC) == -1) {
        /* process error */
    }
    /* get digit using dx_getdig( ) and continue processing */
    .
}```

---

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Example 2: Using `dx_play()` in asynchronous mode.

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
#define MAXCHAN 24
int play_handler();
DX_IOTT prompt[MAXCHAN];
DV_TPT tpt;
DV_DIGIT dig;
main()
{
    int chdev[MAXCHAN], index, index1;
    char *chname;
    int i, srlmode, voxfd;
    /* Set SRL to run in polled mode. */
    srlmode = SR_POLLMODE;
    if (sr_setparm(SRL_DEVICE, SR_MODEID, (void *)&srlmode) == -1) {
        /* process error */
    }
    /* Initialize all the DX_IOTT structures for each individual prompt */
    /* Open the vox file to play; the file descriptor will be used */
    /* by all channels. */
    if ((voxfd = dx_fileopen("prompt.vox", O_RDONLY|O_BINARY)) == -1) {
        /* process error */
    }
    /* For each channel, open the device using dx_open(), set up a DX_IOTT */
    /* structure for each channel, and issue dx_play() in asynchronous mode. */
    for (i=0; i<MAXCHAN; i++) {
        /* Set chname to the channel name, e.g., dxxxB1C1, dxxxB1C2, ... */
        /* Open the device using dx_open(). chdev[i] has channel device */
        /* descriptor. */
        if ((chdev[i] = dx_open(chname,NULL)) == -1) {
            /* process error */
        }
        /* Use sr_enbhdlr() to set up handler function to handle play */
        /* completion events on this channel. */
        if (sr_enbhdlr(chdev[i], TDX_PLAY, play_handler) == -1) {
            /* process error */
        }
        /* Set the DV_TPT structures up for MAXDTMF. Play until one digit is */
        /* pressed or the file is played */
        dx_clrtpt(&tpt,1);
        tpt.tp_type = IO_EOT;       /* only entry in the table */
        tpt.tp_termno = DX_MAXDTMF; /* Maximum digits */
        tpt.tp_length = 1;          /* terminate on the first digit */
        tpt.tp_flags = TF_MAXDTMF;  /* Use the default flags */
        prompt[i].io_type = IO_DEV|IO_EOT; /* play from file */
        prompt[i].io_bufp = 0;
        prompt[i].offset = 0;
        prompt[i].io_length = -1;    /* play till end of file */
        prompt[i].io_nextp = NULL;
        prompt[i].io_fhandle = voxfd;
        /* play the data */
        if (dx_play(chdev[i], &prompt[i], &tpt, DV_ASYNC) == -1) {
            /* process error */
        }
    }
}
```
dx_play( ) plays recorded voice data

Example 3: Defining and playing an alert tone, receiving acknowledgement of the alert tone, and using dx_play( ) to transfer ADSI data.

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

int main(argc,argv)  
   int argc;  
   char* argv[];  
{
   int chfd;  
   char charname[12];  
   parm = SR_POLLMODE;  
   sr_setparm(SRL_DEVICE, SR_MODEID, &parm);  
   /*
   * Open the channel using the command line arguments as input
   */
   sprintf(charname, "%sC", argv[1], argv[2]);
   if (( chfd = dx_open(charname, NULL)) == -1)  
     printf("Board open failed on device %s\n", charname);
   exit(1);
  }  
  printf("Devices open and waiting ...\n");  
  /*
  * Take the phone off-hook to talk to the ADSI phone
  * This assumes we are connected through a Skutch Box.
  */
  if (dx_sethook( chfd, DX_OFFHOOK, EV_SYNC) == -1)  
    printf("sethook failed\n");
```
```
while (1) {
    sleep(5);
    dx_clrdigbuf( chfd );
    printf("Digit buffer cleared ..\n");
    /* * Generate the alert tone */
    iott.io_type = IO_DEV | IO_EOT;
    iott.io_fhandle = dx_fileopen("message.asc",O_RDONLY);
    iott.io_length = -1;
    parm = DM_D;
    if (dx_setparm (chfd, DXCM_DTINITSET, (void *)parm) ==-1){
        printf ("dx_setparm on DTINITSET failed\n");
        exit(1);
    }
    if (dx_play(chfd,iott, (DV_TPT *)NULL, PM_ADSIALERT|EV_SYNC) ==-1) {
        printf("dx_play on the ADSI file failed\n");
        exit(1);
    }
    dx_close(chfd);
    exit(0);
}
```

### Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR() and ATDV_ERRMSGP() to retrieve one of the following error reasons:

- **EDX_BADPARM** • Invalid Parameter
- **EDX_BADIOTT** • Invalid DX_IOTT entry
- **EDX_BADTPT** • Invalid DX_TPT entry
- **EDX_BUSY** • Busy executing I/O function
- **EDX_SYSTEM** • Windows NT system error - check `errno`

### See Also

**Related Functions:**

- `dx_play()`
- `dx_rec()`
- `dx_recf()`
- `dx_setparm()`, `dx_getparm()`

**Setting Speed and Volume:**

- `dx_adjsv()`
- `dx_setsvcond()`
dx_play() plays recorded voice data

Setting Order and Location for Voice Data:

• DX_JOTT (Chapter 4. Voice Data Structures and Device Parameters)

Retrieving and Handling Play Termination Events:

• Event Management functions (Standard Runtime Library Programmer’s Guide for Windows NT and Appendix A of this guide)
• ATDX_TERMMSK( )
• DV_TPT (Appendix A)
**synchronously plays voice data**  

**dx_playf()**

### Name:

int dx_playf(chdev,fnamep,tptp,mode)

### Inputs:

- int chdev  
  - valid Dialogic channel device handle
- char *fnamep  
  - pointer to name of file to play
- DV_TPT *tptp  
  - pointer to Termination Parameter Table Structure
- unsigned short mode  
  - playing mode bit mask for this play session

### Returns:

0 if success  
-1 if failure

### Includes:

srllib.h  
dxxlib.h

### Category:

Convenience

---

### Description

dx_playf() is a convenience function that synchronously plays voice data or transfers ADSI data (using the ADSI protocol) from a single file.

dx_playf() operates the same as synchronous dx_play() if the DX_IOTT structure specified a single file entry. dx_playf() is provided as a convenient way to play back data or transfer ADSI data from a single file without having to specify a DX_IOTT structure for only one file. The dx_playf() function opens and closes the file specified by fnamep while the dx_play() function uses a DX_IOTT structure that requires the application to open and close the file.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>fnamep:</td>
<td>points to the file from which voice data will be played.</td>
</tr>
</tbody>
</table>

For information about other function arguments and transferring ADSI data, see dx_play().

---

### Source Code

```c
/***************************************************************************
*        NAME: int dx_playf(chdev,fnamep,tptp,mode)
* DESCRIPTION: This function opens and plays a named file.
* INPUTS: chdev - channel descriptor
*         tptp - pointer to the termination control block
*         fnamep - pointer to file name
***************************************************************************/
```
dx_play()  
synchronously plays voice data

* OUTPUTS: Data is played.
* RETURNS: 0 - success -1 - failure
* CALLS: open() dx_play() close()
* CAUTIONS: none.

*************************************************************************/
int dx_playf(devd, filep, tptp, mode)
{
    DX_IOTT iott;
    int rval;

    /* If Async then return Error
     * Reason: IOTT's must be in scope for the duration of the play
     */
    if ( mode & EV_ASYNC ) {
        return( -1 );
    }

    /* Open the File */
    if ((iott.io_fhandle = dx_fileopen(filep,O_RDONLY,O_BINARY) == -1) { return -1; }

    /* Use dx_play() to do the Play */
    iott.io_type = IO_EOT | IO_DEV;
    iott.io_offset = (unsigned long)0;
    iott.io_length = -1;
    rval = dx_play(devd,&iott,tptp,mode);

    if (dx_fileclose(iott.io_fhandle) == -1) { return -1; }

    return rval;
}

■ Example

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

main()
{
    int chdev;
    DV_TPT tpt[2];

    /* Open the channel using dx_open(). Get channel device descriptor in * chdev.
    */
    if ((chdev = dx_open("dxd8cl1",NULL)) == -1) { /* process error */
    }

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synchronously plays voice data

dx_playf()

/*
 * Set up the DV_TPT structures for MAXDTMF. Play until one digit is
 * pressed or the file has completed play
 */
dx_clrtpt(tpt,1);
tpt[0].tp_type = IO_EOT;      /* only entry in the table */
tpt[0].tp_termno = DX_MAXDTMF; /* Maximum digits */
tpt[0].tp_length = 1;           /* terminate on the first digit */
tpt[0].tp_flags = TF_MAXDTMF;  /* Use the default flags */

if (dx_playf(chdev,"weather.vox",tpt,EV_SYNC) == -1) {
    /* process error */
}

Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR( ) and
ATDV_ERRMSGP( ) to retrieve one of the following error reasons:

- EDX_BADPARM • Invalid Parameter
- EDX_BADIOTT • Invalid DX_IOTT entry
- EDX_BADTPT • Invalid DX_TPT entry
- EDX_BUSY • Busy executing I/O function
- EDX_SYSTEM • Windows NT system error - check errno

See Also

Related Functions:
  • dx_rec( )
  • dx_recf( )
  • dx_setparm( ), dx_getparm( )

Setting Speed and Volume:
  • dx_adjsv( )
  • dx_setsvcond( )

Setting and Handling Play Termination:
  • ATDX_TERMMSK( )
  • DV_TPT (Appendix A)
dx_playiottdata()  plays back recorded voice data from multiple sources

Name:  short dx_playiottdata(chdev, iottp, tptp, xpbp, mode)

Inputs:  
- int chdev  • valid Dialogic channel device handle
- DX_IOTT *iottp  • pointer to I/O transfer table
- DV_TPT *tptp  • pointer to termination parameter block
- DX_XPB *xpbp  • pointer to I/O transfer parameter block
- unsigned short mode  • play mode

Returns:  
- 0 if success
- -1 if failure

Includes:  
- srllib.h
- dxxxlib.h

Category:  I/O function

Mode:  synchronous or asynchronous

Description

The dx_playiottdata() function plays back recorded voice data from multiple sources on a channel. The file format for the files to be played is specified in the wFileFormat field of the DX_XPB. Other fields in the DX_XPB describe the data format. For files that include data format information (i.e. WAVE files), these other fields are ignored.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev</td>
<td>channel device descriptor.</td>
</tr>
<tr>
<td>iottp</td>
<td>the voice data may come from any combination of data files, memory, or custom devices. The order of playback and the location of the voice data is specified in an array of DX_IOTT structures pointed to by iottp</td>
</tr>
<tr>
<td>tptp</td>
<td>pointer to Termination parameter table</td>
</tr>
<tr>
<td>xpbp</td>
<td>pointer to I/O transfer parameter block</td>
</tr>
<tr>
<td>mode</td>
<td>specifies the record mode:</td>
</tr>
<tr>
<td></td>
<td>PM_TONE</td>
</tr>
<tr>
<td></td>
<td>EV_SYNCH</td>
</tr>
<tr>
<td></td>
<td>EVASYNCH</td>
</tr>
</tbody>
</table>
Cautions

1. All files specified in the DX_IOTT table must be of the same file format type and match the file format indicated in DX_XPB.

2. All files specified in the DX_IOTT table must contain data of the type described in DX_XPB.

3. When playing or recording VOX files, the data format is specified in DX_XPB rather than through the mode argument of this function.

4. When set to play WAVE files, all other fields in the DX_XPB are ignored.

5. When set to play WAVE files, this function will fail if an unsupported data format is attempted to be played. The supported data forms are:
   - 6, 8, and 11KHz linear 8-bit PCM (WAVE_FORMAT_PCM)
   - 6, 8, and 11KHz mu-law 8-bit PCM (WAVE_FORMAT_MULAW)
   - 6, 8, and 11KHz a-law 8-bit PCM (WAVE_FORMAT_ALAW)
   - 6 and 8KHz 4-bit Oki ADPCM (WAVE_FORMAT_DIALOGIC_OKI_ADPCM)

Example

```c
#include "srilib.h"
#include "dxoolib.h"

int chdev;     /* channel descriptor */
int fd;         /* file descriptor for file to be played */
DX_IOTT iott;   /* I/O transfer table */
DV_TPT tpt;     /* termination parameter table */
DX_XPB xpb;     /* I/O transfer parameter block */

/* Open channel */
if ((chdev = dx_open("dxxxB1C1",0)) == -1) {
    printf("Cannot open channel\n");
    printf("errno = %d\n",errno);
    exit(1);
}
/* Set to terminate play on 1 digit */
tpt.tp_type   = IO_EOT;
tpt.tp_termno = DX_MAXDTMF;
tpt.tp_length = 1;
tpt.tp_flags  = TF_MAXDTMF;
/* Open VOX file to play */
if ((fd = dx_fileopen("HELLO.VOX",O_RDONLY|O_BINARY)) == -1) {
    printf("File open error\n");
    printf("errno = %d\n",errno);
    exit(2);
}
/* Set up DX_IOTT */
```

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dx_playiottdata( )  plays back recorded voice data from multiple sources

iott.io_fhandle = fd;
iott.io_bufp = 0;
iott.io_offset = 0;
iott.io_length = -1;
iott.io_typ = IO_DEV | IO_EOT;
/*
 * Specify VOX file format for ADPCM at 8KHz
 */
xpb.wFileFormat = FILE_FORMAT_VOX;
xpb.wDataFormat = DATA_FORMAT_DIALOGIC_ADPCM;
xpb.nSamplesPerSec = DRT_8KHZ;
xpb.nBitsPerSample = 4;

/* Wait forever for phone to ring and go offhook */
if (dx_wtring(chdev,1,DX_OFFHOOK,-1) == -1) {
    printf("Error waiting for ring - %s\n", ATDV_LASTERR(chdev));
    exit(3);
}
/* Start playback */
if (dx_playiottdata(chdev,&iott,&tpt,&xpb,EV_SYNC)==-1) {
    printf("Error playing file - %s\n", ATDV_ERRMSGP(chdev));
    exit(4);
}

■ Errors

In asynchronous mode, function returns immediately and a TDX_PLAY event is queued upon completion. Check ATDX_LASTTERM() for the termination reason. If a failure occurs, then a TDX_ERROR event will be queued. Use ATDV_LASTERR() to determine the reason for error.

In synchronous mode, if this function returns -1 to indicate failure, one of the following reasons will be contained by ATDV_LASTERR( ):

<table>
<thead>
<tr>
<th>Equate</th>
<th>Returned When</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_BUSY</td>
<td>Channel is busy</td>
</tr>
<tr>
<td>EDX_XBPSPARM</td>
<td>Invalid DX_XPB setting</td>
</tr>
<tr>
<td>EDX_BAliOTT</td>
<td>Invalid DX_IOTT setting</td>
</tr>
<tr>
<td>EDX_SYSTEM</td>
<td>System I/O errors</td>
</tr>
<tr>
<td>EDX_BADWAVFILE</td>
<td>Invalid WAV file</td>
</tr>
<tr>
<td>EDX_SH_BACMD</td>
<td>Unsupported command or WAV file format</td>
</tr>
</tbody>
</table>

■ See Also

• dx_playwav()
• dx_playvox()
**Description**

The `dx_playtone()` function plays tone defined by TN_GEN template, which defines the frequency amplitude and duration of a single or dual frequency tone to be played.

**NOTE:** The `dx_playtone()` function is necessary for supporting the R2MF protocol in an application. See `r2_playbsig()` for information.

**Asynchronous Operation**

To run this function asynchronously set the `mode` field to EV_ASYNC. When running asynchronously, this function will return 0 to indicate it has initiated successfully, and will generate a termination event (see below) to indicate completion.

Set termination conditions using the DV_TPT structure. This structure is pointed to by the `tptp` parameter described below.

Termination of this function is indicated by a TDX_PLAYTONE event.

Use the SRL Event Management functions to handle the termination event. See Appendix A for more information about the Event Management functions.

After `dx_playtone()` terminates, use the `ATDX_TERMMSK()` function to determine the reason for termination.
dx_playtone() plays tone defined by TN_GEN template

### Synchronous Operation

By default, this function runs synchronously, and will return a 0 to indicate that it has completed successfully.

Set termination conditions using the DV_TPT structure. This structure is pointed to by the ttp parameter described below.

Termination of synchronous play is indicated by a return value of 0.

After dx_playtone() terminates, use the ATDX_TERMMSK() function to determine the reason for termination.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using dx_open().</td>
</tr>
<tr>
<td>tngenp:</td>
<td>points to the TN_GEN template structure, which defines the frequency, amplitude and duration of a single or dual frequency tone. See Chapter 4. Voice Data Structures and Device Parameters for a full description of this template. dx_bidtngen() can be used to set up the structure.</td>
</tr>
<tr>
<td>ttp:</td>
<td>points to the DV_TPT data structure, which specifies one of the following terminating conditions for this function:</td>
</tr>
<tr>
<td></td>
<td>DX_DIGTYPE  • Digit termination for user-defined tone</td>
</tr>
<tr>
<td></td>
<td>DX_MAXDTMF  • Maximum number of digits received</td>
</tr>
<tr>
<td></td>
<td>DX_MAXSIL   • Maximum silence</td>
</tr>
<tr>
<td></td>
<td>DX_MAXNOSIL • Maximum non-silence</td>
</tr>
<tr>
<td></td>
<td>DX_LCOFF    • Loop current off</td>
</tr>
<tr>
<td></td>
<td>DX_IDDTIME  • Inter-digit delay</td>
</tr>
<tr>
<td></td>
<td>DX_MAXTIME  • Function time</td>
</tr>
<tr>
<td></td>
<td>DX_DIGMASK  • Digit mask termination</td>
</tr>
<tr>
<td></td>
<td>DX_PMOFF    • Pattern match silence off</td>
</tr>
<tr>
<td></td>
<td>DX_PMON     • Pattern match silence on</td>
</tr>
<tr>
<td></td>
<td>DX_TONE     • Tone-off or Tone-on detection</td>
</tr>
</tbody>
</table>
**Parameter**  
**mode:** Specifies whether to run this function asynchronously or synchronously. Set to one of the following:

- **EV_ASYNC:** Run `dx_playtone()` asynchronously.
- **EV_SYNC:** Run `dx_playtone()` synchronously (default).

**Cautions**

1. The channel must be idle when calling this function.
2. If the tone generation template contains an invalid `tg_dflag`, or the specified amplitude or frequency is outside the valid range, `dx_playtone()` will generate a TDX_ERROR event if asynchronous, or -1 if synchronous.

**Example**

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
#define TID_1   101

main()
{
    TN_GEN   tngen;
    DV_TPT   tpt[ 5 ];
    int      dxxxdev;

    /* Open the Voice Channel Device and Enable a Handler */
    if ( ( dxxxdev = dx_open( "dxxxB1C1", NULL) ) == -1 ) {
        perror( "dxxxB1C1" );
        exit( 1 );
    }

    /* */
}  
243-CD"
* Describe a Simple Dual Tone Frequency Tone of 950-1050 Hz and 475-525 Hz using leading edge detection.
* 
if ( dx_blddt( TID_1, 1000, 50, 500, 25, TN_LEADING ) == -1 ) {
    printf( "Unable to build a Dual Tone Template\n" );
}

/* Bind the Tone to the Channel */
if ( dx_addtone( dxxxdev, NULL, 0 ) == -1 ) {
    printf( "Unable to Bind the Tone %d\n", TID_1 );
    ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev );
    dx_close( dxxxdev );
    exit( 1 );
}

/* Enable Detection of ToneId TID_1 */
if ( dx_enbtone( dxxxdev, TID_1, DM_TONEON | DM_TONEOFF ) == -1 ) {
    printf( "Unable to Enable Detection of Tone %d\n", TID_1 );
    ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev );
    dx_close( dxxxdev );
    exit( 1 );
}

/* Build a Tone Generation Template. 
* This template has Frequency1 = 1140, 
* Frequency2 = 1020, amplitude at -10dB for 
* both frequencies and duration of 100 * 10 msecs. 
*/
dx_bldtngen( &tngen, 1140, 1020, -10, -10, 100 );

/* Set up the Terminating Conditions */
tpt[0].tp_type = IO_CONT;
tpt[0].tp_termno = DX_TONE;
tpt[0].tp_length = TID_1;
tpt[0].tp_flags = TF_TONE;
tpt[0].tp_data = DX_TONEON;

tpt[1].tp_type = IO_CONT;
tpt[1].tp_termno = DX_TONE;
tpt[1].tp_length = TID_1;
tpt[1].tp_flags = TF_TONE;
tpt[1].tp_data = DX_TONEOFF;

tpt[2].tp_type = IO_EOT;
tpt[2].tp_termno = DX_MAXTIME;
tpt[2].tp_length = 6000;
tpt[2].tp_flags = TF_MAXTIME;

if ( dx_playtone( dxxxdev, &tngen, tpt, EV_SYNC ) == -1 ){
    printf( "Unable to Play the Tone\n" );
    ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev );
}
---

### plays tone defined by TN_GEN template

```c
    if ( dx_close( dxxxdev ) != 0 ) {
        perror( "close" );
    }

    /*
    * Close the opened Voice Channel Device
    */
    if ( dx_close( dxxxdev ) != 0 ) {
        perror( "close" );
    }

    /* Terminate the Program */
    exit( 1 );
```

### Errors

If this function returns -1 to indicate failure, use `ATDV_LASTERR( )` and `ATDV_ERRMSGP( )` to retrieve one of the following error reasons:

- **EDX_BADPARM** • Invalid parameter
- **EDX_BADPROD** • Function not supported on this board
- **EDX_BADTPT** • Invalid DV_TPT entry
- **EDX_BUSY** • Busy executing I/O function
- **EDX_AMPLGEN** • Invalid amplitude value in `TN_GEN` structure
- **EDX_FREQGEN** • Invalid frequency component in `TN_GEN` structure
- **EDX_FLAGGEN** • Invalid tn_dflag field in `TN_GEN` structure
- **EDX_SYSTEM** • Windows NT system error - check `errno`

### See Also

Related to Tone Generation:

- `dx_bldtngen( )`
- `TN_GEN (Chapter 4. Voice Data Structures and Device Parameters)`
- "Global Tone Generation" (Voice Features Guide for Windows NT)

R2MF functions:
dx_playtone( )  

plays tone defined by TN_GEN template

- r2_creatfsig( )
- r2_playbsig( )

Handling and Retrieving dx_playtone( ) Termination Events:

- Event Management functions (Standard Runtime Library Programmer’s Guide for Windows NT and Appendix A)
- DV_TPT (Appendix A)
- ATDX_TERMMSK( )
plays voice data stored in a single VOX file \hspace{1cm} dx_playvox() \\

**Name:** SHORT dx_playvox(chdev, filenamep, tptp, xpbp, mode) \\

**Inputs:** 
- int chdev \hspace{1cm} • valid Dialogic channel device handle 
- char *filenamep \hspace{1cm} • pointer to name of file to play 
- DV_TPT *tptp \hspace{1cm} • pointer to termination parameter block 
- DX_XPB *xpbp \hspace{1cm} • pointer to I/O transfer parameter block 
- unsigned short mode \hspace{1cm} • play mode 

**Returns:** 
- 0 if successful 
- -1 if failure 

**Includes:** dxxxlib.h 

**Category:** Convenience function 

**Mode:** synchronous 

--- 

**Description**

The \texttt{dx_playvox()} convenience function plays voice data stored in a single VOX file. If \texttt{xpbp} is set to NULL, it will interpret the data as 6KHz linear ADPCM. 

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev</td>
<td>Channel device descriptor</td>
</tr>
<tr>
<td>tcbp</td>
<td>Pointer to termination parameter table</td>
</tr>
<tr>
<td>filenamep</td>
<td>Pointer to name of file to play</td>
</tr>
<tr>
<td>xpbp</td>
<td>Pointer to I/O transfer parameter block (See the \textit{DX_XPB} data structure)</td>
</tr>
<tr>
<td>mode</td>
<td>specifies the play mode: \texttt{PM_TONE} play 200 ms audible tone \texttt{EV_SYNC} synchronous operation (must be specified)</td>
</tr>
</tbody>
</table>

**NOTE:** Both \texttt{PM_TONE} and \texttt{EV_SYNC} can be specified by ORing the two values.
dx_playvox() plays voice data stored in a single VOX file

■ Cautions

When playing or recording VOX files, the data format is specified in DX_XPB rather than through the dl_stprm() function.

■ Example

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

int chdev; /* channel descriptor */
DV_TPT tpt; /* termination parameter table */.

/* Open channel */
if ((chdev = dx_open("dxxxB1C1", 0)) == -1) {
    printf("Cannot open channel\n");
    printf("errno = %d\n", errno);
    exit(1);
}

/* Set to terminate play on 1 digit */
tpt.tp_type   = IO_EOT;
tpt.tp_termno = DX_MAXDTMF;
tpt.tp_length = 1;
tpt.tp_flags  = TF_MAXDTMF;
/* Wait forever for phone to ring and go offhook */
if (dx_wtring(chdev, 1, DX_OFFHOOK, -1) == -1) {
    printf("Error waiting for ring - %s\n", ATDV_LASTERR(chdev));
    exit(3);
}

/* Start 6KHz ADPCM playback */
if (dx_playvox(chdev, &tpt, "HELLO.VOX", NULL, 0) == -1) {
    printf("Error playing file - %s\n", ATDV_ERRMSGP(chdev));
    exit(4);
}
```

■ Errors

If this function returns -1 to indicate failure, one of the following reasons will be contained by ATDV_LASTERR():

<table>
<thead>
<tr>
<th>Equate</th>
<th>Returned When</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_BUSY</td>
<td>Channel is busy</td>
</tr>
<tr>
<td>EDX_XPBPARM</td>
<td>Invalid DX_XPB setting</td>
</tr>
<tr>
<td>EDX_BADIOTT</td>
<td>Invalid DX_IOTT setting</td>
</tr>
<tr>
<td>EDX_SYSTEM</td>
<td>System I/O errors</td>
</tr>
<tr>
<td>EDX_BADWAVFILE</td>
<td>Invalid WAV file</td>
</tr>
</tbody>
</table>

248-CD
plays voice data stored in a single VOX file \hspace{1cm} \textit{dx\_playvox( )}

<table>
<thead>
<tr>
<th>Equate</th>
<th>Returned When</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_SH_BADCMD</td>
<td>Unsupported command or WAV file format</td>
</tr>
</tbody>
</table>

- See Also
  - \textit{dx\_playiotldata( )}
  - \textit{dx\_playwav( )}
dx_playwav( )  

plays voice data stored in a single WAVE file

<table>
<thead>
<tr>
<th>Name:</th>
<th>SHORT  dx_playwav(chdev, filenamep, tptp, mode)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>int chdev</td>
</tr>
<tr>
<td></td>
<td>• valid Dialogic channel device handle</td>
</tr>
<tr>
<td></td>
<td>char *filenamep</td>
</tr>
<tr>
<td></td>
<td>• pointer to name of file to play</td>
</tr>
<tr>
<td></td>
<td>DV_TPT*tptp</td>
</tr>
<tr>
<td></td>
<td>• pointer to termination parameter block</td>
</tr>
<tr>
<td></td>
<td>unsigned short mode</td>
</tr>
<tr>
<td></td>
<td>• play mode</td>
</tr>
<tr>
<td>Returns:</td>
<td>0 if successful</td>
</tr>
<tr>
<td></td>
<td>-1 if failure</td>
</tr>
<tr>
<td>Includes:</td>
<td>srllib.h</td>
</tr>
<tr>
<td></td>
<td>dxxlib.h</td>
</tr>
<tr>
<td>Category:</td>
<td>Convenience function</td>
</tr>
<tr>
<td>Mode:</td>
<td>synchronous</td>
</tr>
</tbody>
</table>

**Description**

The dx_playwav( ) convenience function plays voice data stored in a single WAVE file. This function calls dx_playiotdata( ).

The function does not specify a DX_XPB structure because the WAVE file contains the necessary format information.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev</td>
<td>Channel device descriptor</td>
</tr>
<tr>
<td>tcbp</td>
<td>Pointer to termination parameter table</td>
</tr>
<tr>
<td>filenamep</td>
<td>Pointer to name of file to play</td>
</tr>
<tr>
<td>mode</td>
<td>specifies the play mode:</td>
</tr>
<tr>
<td></td>
<td>PM_TONE</td>
</tr>
<tr>
<td></td>
<td>EV_SYNC</td>
</tr>
<tr>
<td></td>
<td>(must be specified)</td>
</tr>
</tbody>
</table>

**NOTE:** Both PM_TONE and EV_SYNC can be specified by ORing the two values.

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**Cautions**

This function fails when an unsupported data waveform attempts to play. The supported waveforms are:

- 6, 8, and 11KHz linear 8-bit PCM (WAVE_FORMAT_PCM)
- 6, 8, and 11KHz mu-law 8-bit PCM (WAVE_FORMAT_MULAW)
- 6, 8, and 11KHz a-law 8-bit PCM (WAVE_FORMAT_ALAW)
- 6 and 8KHz 4-bit Oki ADPCM (WAVE_FORMAT_DIALOOGIC_OKI_ADPCM)

**Example**

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

int chdev; /* channel descriptor */
DV_TPT tpt; /* termination parameter table */
.
.
.
/* Open channel */
if ((chdev = dx_open("dxxxB1C1",0)) == -1) {
    printf("Cannot open channel\n");
    printf("errno = %d\n",errno);
    exit(1);
}
/* Set to terminate play on 1 digit */
tpt.tp_type   = IO_EOT;
tpt.tp_termno = DX_MAXDTMF;
tpt.tp_length = 1;
tpt.tp_flags  = TF_MAXDTMF;
/* Wait forever for phone to ring and go offhook */
if (dx_wtring(chdev,1,DX_OFFHOOK,-1) == -1) {
    printf("Error waiting for ring - \tsn", ATDV_LASTERR(chdev));
    exit(3);
}
/* Start playback */
if (dx_playwav(chdev,&tpt,"HELLO.WAV",EV_SYNC) == -1) {
    printf("Error playing file - \tsn", ATDV_ERRMSGP(chdev));
    exit(4);
}
```

**Errors**

If this function returns -1 to indicate failure, one of the following reasons will be contained by `ATDV_LASTERR()`:
dx_playwav()  
plays voice data stored in a single WAVE file

<table>
<thead>
<tr>
<th>Equate</th>
<th>Returned When</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_BUSY</td>
<td>Channel is busy</td>
</tr>
<tr>
<td>EDX_XPBPARM</td>
<td>Invalid DX_XPB setting</td>
</tr>
<tr>
<td>EDX_BADIOTT</td>
<td>Invalid DX_IOTT setting</td>
</tr>
<tr>
<td>EDX_SYSTEM</td>
<td>System I/O errors</td>
</tr>
<tr>
<td>EDX_BADWAVFILE</td>
<td>Invalid WAV file</td>
</tr>
<tr>
<td>EDX_SH_BADCMD</td>
<td>Unsupported command or WAV file format</td>
</tr>
</tbody>
</table>

See Also
- dx_playiotdata()
- dx_playvox()
Name: int dx_rec(chdev,iottp,tptp,mode)
Inputs:
- int chdev: valid Dialogic channel device handle
- DX_IOTT *iottp: pointer to I/O Descriptor Table
- DV_TPT *tptp: pointer to Termination Parameter Table Structure
- unsigned short mode: asynchronous/synchronous setting and recording mode bit mask for this record session

Returns: 0 if successful
-1 if failure
Includes: srllib.h
dxxxlib.h
Category: I/O
Mode: synchronous/asyncronous

Description

The dx_rec( ) function records voice data from a single channel. The data may be recorded to a combination of data files, memory, or custom devices.

The order in which voice data is recorded is specified in the DX_IOTT structure. The DX_IOTT structure must remain in scope for the duration of the function if running asynchronously.

After dx_rec( ) is called, recording continues until dx_stopch( ) is called, the data requirements specified in the DX_IOTT are fulfilled, or until one of the conditions for termination in the DV_TPT is satisfied. When dx_rec( ) terminates, the current channel’s status information, including the reason for termination, can be accessed using Extended Attribute functions.

NOTE: For a single file synchronous record, dx_recf( ) is more convenient because you do not have to set up a DX_IOTT structure. See the function description of dx_recf( ) for information.

Asynchronous Operation

To run this function asynchronously set the mode field to EV_ASYNC. When running asynchronously, this function will return 0 to indicate it has initiated
\textit{dx\_rec( )} records voice data from a single channel successfully, and will generate a termination event (see below) to indicate completion.

Set termination conditions using the DV\_TPT structure. This structure is pointed to by the \texttt{tptp} parameter described below.

Termination of asynchronous recording is indicated by a TDX\_RECORD event.

Use the SRL Event Management functions to handle the termination event. See Appendix A for more information about the Event Management functions.

After \texttt{dx\_rec( )} terminates, use the \texttt{ATDX\_TERMMSK( )} function to determine the reason for termination.

\section*{Synchronous Operation}

By default, this function runs synchronously, and will return a 0 to indicate that it has completed successfully.

Set termination conditions using the DV\_TPT structure. This structure is pointed to by the \texttt{tptp} parameter described below. After \texttt{dx\_rec( )} terminates, use the \texttt{ATDX\_TERMMSK( )} function to determine the reason for termination.

The function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using \texttt{dx_open( )}.</td>
</tr>
<tr>
<td>iottp:</td>
<td>points to the I/O Transfer Table Structure, \textit{DX_IOTT}, which specifies the order in which and media onto which the voice data will be recorded. This structure is defined in \textit{Chapter 4. Voice Data Structures and Device Parameters} and must remain in scope for the duration of the function if using asynchronously.</td>
</tr>
</tbody>
</table>
| tptp:     | points to the Termination Parameter Table Structure, DV\_TPT, which specifies termination conditions for recording. Valid termination conditions for this function are listed below: 
\begin{itemize}
  \item DX\_DIGTYPE
\end{itemize} |

\textit{254-CD}
records voice data from a single channel  

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DX_MAXDTMF</td>
<td>Maximum number of digits received</td>
</tr>
<tr>
<td>DX_MAXSIL</td>
<td>Maximum silence</td>
</tr>
<tr>
<td>DX_MAXNOSIL</td>
<td>Maximum non-silence</td>
</tr>
<tr>
<td>DX_LCOFF</td>
<td>Loop current off</td>
</tr>
<tr>
<td>DX_IDDTIME</td>
<td>Inter-digit delay</td>
</tr>
<tr>
<td>DX_MAXTIME</td>
<td>Function time</td>
</tr>
<tr>
<td>DX_DIGMASK</td>
<td>Digit mask termination</td>
</tr>
<tr>
<td>DX_PMOFF</td>
<td>Pattern match silence off</td>
</tr>
<tr>
<td>DX_PMON</td>
<td>Pattern match silence on</td>
</tr>
<tr>
<td>DX_TONE</td>
<td>Tone-off or Tone-on detection</td>
</tr>
</tbody>
</table>

See Appendix A, which describes the Standard Runtime Library, for information about this structure.

**NOTE:** In addition to DV_TPT terminations, the function can fail due to maximum byte count, `dx_stopch()`, or end of file. See `ATDX_TERMMSK()` for a full list of termination reasons.

**mode:** defines the recording mode. One or more of the values listed below may be selected in the bit mask (see Table 5 for record mode combinations).

Choose one only:

- **EV_ASYNC**: Run `dx_rec()` asynchronously.
- **EV_SYNC**: Run `dx_rec()` synchronously (default).

Choose one or more:

- **MD_ADPCM**: Record using Adaptive Differential Pulse Code Modulation encoding algorithm (4 bits per sample). Recording with ADPCM is the default setting.
**dx_rec( )**  
records voice data from a single channel  

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>MD_PCM:</td>
<td>Record using Pulse Code Modulation encoding algorithm (8 bits per sample).</td>
</tr>
<tr>
<td>MD_GAIN:</td>
<td>Record with Automatic Gain Control (AGC). Recording with AGC is the default setting.</td>
</tr>
<tr>
<td>MD_NOGAIN:</td>
<td>Record without AGC.</td>
</tr>
<tr>
<td>RM_ALAW:</td>
<td>Record using A-Law.</td>
</tr>
<tr>
<td>RM_TONE:</td>
<td>Transmit a tone before initiating record. If this mode is not selected, no tone will be transmitted (the default setting).</td>
</tr>
<tr>
<td>RM_SR6:</td>
<td>Record using 6KHz sampling rate (6,000 samples per second). This is the default setting.</td>
</tr>
<tr>
<td>RM_SR8:</td>
<td>Record using 8KHz sampling rate (8,000 samples per second).</td>
</tr>
</tbody>
</table>

**NOTES:**  
1. The rate specified in the last record function will apply to the next record function, unless the rate was changed in the parameter DXCH_RECRDRATE using dx_setparm().  
2. Specifying RM_SR6 or RM_SR8 in mode changes the setting of the parameter DXCH_RECRDRATE. DXCH_RECRDRATE can also be set and queried using dx_setparm() and dx_getparm(). The default setting for DXCH_RECRDRATE is 6KHz.  
3. If both MD_ADPCM and MD_PCM are set, MD_PCM will take precedence. If both MD_GAIN and MD_NOGAIN are set, MD_NOGAIN will take precedence. If both RM_TONE and NULL are set, RM_TONE takes precedence. If both RM_SR6 and RM_SR8 are set, RM_SR6 will take precedence.  
4. MD_PCM and MD_NOGAIN can be used on D/12x or D/81A boards.
records voice data from a single channel  

\[ \text{dx\_rec()} \]

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>5.</td>
<td>When playing pre-recorded data, make sure it is played using the same encoding algorithm and sampling rate used when the data was recorded.</td>
</tr>
<tr>
<td>6.</td>
<td>\text{dx_rec()} will run synchronously if you do not specify \text{EV_ASYNC}, or if you specify \text{EV_SYNC} (default).</td>
</tr>
<tr>
<td>7.</td>
<td>The D/21E, D/41E, D/41ESC, D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1 and D/320SC boards enable the user to select either A-Law or mu-Law encoding of data. The default on the board is set to mu-Law and returns to mu-Law after each record. The A-Law parameters must be passed each time the record function is called. Enable A-Law record by OR’ing the new record, \text{RM_ALAW}.</td>
</tr>
</tbody>
</table>

Table 5 shows recording mode selections. The first column of the table lists all possible combinations of record features, and the first row lists each type of encoding algorithm (ADPCM or PCM) and the data-storage rate for each algorithm/sampling rate combination in parenthesis (24 Kbps, 32 Kbps, 48 Kbps, or 64 Kbps).

Select the desired record feature in the first column of the table and move across that row until the column containing the desired encoding algorithm and data-storage rate is reached. The record modes that must be entered in \text{dx\_rec()} are provided where the features row, and encoding algorithm/data-storage rate column intersect. Parameters listed in \{ \} are default settings and do not have to be specified.
Table 5. Record Mode Selections

<table>
<thead>
<tr>
<th>Feature</th>
<th>ADPCM (24 Kbps)</th>
<th>ADPCM (32 Kbps)</th>
<th>PCM (48 Kbps)</th>
<th>PCM (64 Kbps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>• AGC</td>
<td>RM_SR6 {MD_ADPCM}</td>
<td>RM_SR8 {MD_ADPCM}</td>
<td>RM_SR8 {MD_PCM}</td>
<td>RM_SR8 {MD_PCM}</td>
</tr>
<tr>
<td>• No Tone</td>
<td>{MD_GAIN}</td>
<td>{MD_GAIN}</td>
<td>{MD_GAIN}</td>
<td>{MD_GAIN}</td>
</tr>
<tr>
<td>• No AGC</td>
<td>MD_NOGAIN</td>
<td>MD_NOGAIN</td>
<td>MD_NOGAIN</td>
<td>MD_NOGAIN</td>
</tr>
<tr>
<td>• No Tone</td>
<td>RM_SR6 {MD_ADPCM}</td>
<td>RM_SR8 {MD_ADPCM}</td>
<td>RM_SR6 {MD_PCM}</td>
<td>RM_SR6 {MD_PCM}</td>
</tr>
<tr>
<td>• AGC</td>
<td>RM_TONE RM_SR6 {MD_ADPCM}</td>
<td>RM_TONE RM_SR8 {MD_ADPCM}</td>
<td>RM_TONE RM_SR6 {MD_PCM}</td>
<td>RM_TONE RM_SR6 {MD_PCM}</td>
</tr>
<tr>
<td>• No AGC</td>
<td>MD_NOGAIN</td>
<td>MD_NOGAIN</td>
<td>MD_NOGAIN</td>
<td>MD_NOGAIN</td>
</tr>
<tr>
<td>• Tone</td>
<td>RM_SR6 {MD_ADPCM}</td>
<td>RM_SR8 {MD_ADPCM}</td>
<td>RM_SR6 {MD_PCM}</td>
<td>RM_SR6 {MD_PCM}</td>
</tr>
<tr>
<td>• No AGC</td>
<td>MD_NOGAIN</td>
<td>MD_NOGAIN</td>
<td>MD_NOGAIN</td>
<td>MD_NOGAIN</td>
</tr>
<tr>
<td>• Tone</td>
<td>RM_SR6 {MD_ADPCM}</td>
<td>RM_SR8 {MD_ADPCM}</td>
<td>RM_SR6 {MD_PCM}</td>
<td>RM_SR6 {MD_PCM}</td>
</tr>
</tbody>
</table>

{ } = Default modes.
* = Select if A-Law encoding is required (supported on D/41ESC, D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1 and D/320SC boards only).

NOTE: dx_rec( ) will run synchronously if you do not specify EV_ASYNC, or if you specify EV_SYNC (default).

■ Cautions

None.
Example 1: Using dx_rec( ) in synchronous mode

```c
#include <fcntl.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
#define MAXLEN 10000
main()
{
    DV_TPT tpt;
    DX_IOTT iott[2];
    int chdev;
    char basebufp[MAXLEN];
    /*
    * open the channel using dx_open( )
    */
    if ((chdev = dx_open("dxxxB1C1", NULL)) == -1) {
        /* process error */
    }
    /*
    * Set up the DV_TPT structures for MAXDTMF
    */
    dx_clrtpt(&tpt, 1);
    tpt.tp_type = IO_EOT; /* last entry in the table */
    tpt.tp_termno = DX_MAXDTMF; /* Maximum digits */
    tpt.tp_length = 1; /* terminate on the first digit */
    tpt.tp_flags = TF_MAXDTMF; /* Use the default flags */
    /*
    * Set up the DX_IOTT. The application records the voice data to memory
    * allocated by the user.
    */
    iott[0].io_type = IO_MEM | IO_CONT; /* Record to memory */
    iott[0].io_bufp = basebufp; /* Set up pointer to buffer */
    iott[0].io_offset = 0; /* Start at beginning of buffer */
    iott[0].io_length = MAXLEN; /* Record 10,000 bytes of voice data */
    iott[1].io_type = IO_DEV | IO_EOT; /* Record to file, last DX_IOTT
    * entry */
    iott[1].io_bufp = 0; /* Set up pointer to buffer */
    iott[1].io_offset = 0; /* Start at beginning of buffer */
    iott[1].io_length = MAXLEN; /* Record 10,000 bytes of voice
    * data */
    if (iott[1].io_fhandle = dx_fileopen("file.vox",
        O_RDWR | O_CREAT | O_TRUNC | O_BINARY, 0666)) == -1) {
        /* process error */
    }
    /* clear previously entered digits */
    if (dx_clrdigbuf(chdev) == -1) {
        /* process error */
    }
    if (dx_rec(chdev, &iott[0], &tpt, RM_TONE | EV_SYNC) == -1) {
        /* process error */
    }
    /* Analyze the data recorded */
    .
    .
}
```

Example 2: Using dx_rec( ) in asynchronous mode

```c
#include <stdio.h>
#include <fcntl.h>
```
dx_rec( )

records voice data from a single channel

#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
#define MAXLEN 10000
#define MAXCHAN 24
int record_handler();
DV_TPT tpt;
DX_IOTT iott[MAXCHAN];
int chdev[MAXCHAN];
char basebufp[MAXCHAN][MAXLEN];
main()
{
    int i, srlmode;
    char *chname;
    /* Set SRL to run in polled mode. */
    srlmode = SR_POLLMODE;
    if (sr_setparm(SRLDEVICE, SR_MODEID, (void *)&srlmode) == -1) {
        /* process error */
    }
    /* Start asynchronous dx_rec() on all the channels. */
    for (i=0; i<MAXCHAN; i++) {
        /* Set chname to the channel name, e.g., dxxxB1C1, dxxxB1C2,... */
        /* open the channel using dx_open( ) */
        if ((chdev[i] = dx_open(chname, NULL)) == -1) {
            /* process error */
        }
        /* Using sr_enbhdlr(), set up handler function to handle record
           completion events on this channel. */
        if (sr_enbhdlr(chdev[i], TDX_RECORD, record_handler) == -1) {
            /* process error */
        }
        /* Set up the DV_TPT structures for MAXDTMF */
        dx_clrtpt((tpt, 1);
        tpt.tp_type = IO_EOT;    /* last entry in the table */
        tpt.tp_termno = DX_MAXDTMF;  /* Maximum digits */
        tpt.tp_length = 1;       /* terminate on the first digit */
        tpt.tp_flags = TF_MAXDTMF; /* Use the default flags */
        /* Set up the DX_IOTT. The application records the voice data to memory
           allocated by the user. */
        iott[i].io_type = IO_MEM | IO_EOT; /* Record to memory, last DX_IOTT */
        iott[i].io_bufp = basebufp[i];  /* Set up pointer to buffer */
        iott[i].io_offset = 0;         /* Start at beginning of buffer */
        iott[i].io_length = MAXLEN;    /* Record 10,000 bytes voice data */
        /* clear previously entered digits */
        if (dx_clrdigbuf(chdev) == -1) {
            /* process error */
        }
        /* Start asynchronous dx_rec() on the channel */
        if (dx_rec(chdev[i], &iott[i], tpt, RM_TONE | EV_ASYNC) == -1) {
            /* process error */
        }
    }
    /* Use sr_waitev to wait for the completion of dx_rec(). */
    /* On receiving the completion event, TDX_RECORD, control is transferred
     * to a handler function previously established using sr_enbhdlr(). */

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**dx_rec( )**

*records voice data from a single channel*

```c
int record_handler()
{
    long term;
    /* Use ATDX_TERMMSK() to get the reason for termination. */
    term = ATDX_TERMMSK(sr_getevtdev());
    if (term & TM_MAXDTMF) {
        printf("record terminated on receiving DTMF digit(s)\n");
    } else if (term & TM_NORMTERM) {
        printf("normal termination of dx_rec()\n");
    } else {
        printf("Unknown termination reason: %x\n", term);
    }
    /* Kick off next function in the state machine model. */
    return 0;
}
```

### Errors

If this function returns -1 to indicate failure, use **ATDV_LASTERR( )** and **ATDV_ERRMSGP( )** to retrieve one of the following error reasons:

- **EDX_BADDEV** Invalid Device Descriptor
- **EDX_BADPARM** Invalid Parameter
- **EDX_BADIOTT** Invalid DX_IOTT entry
- **EDX_BADTPT** Invalid DX_TPT entry
- **EDX_BUSY** Busy executing I/O function
- **EDX_SYSTEM** Windows NT system error - check errno

### See Also

**Related Functions:**

- **dx_recf( )**
- **dx_play( )**
- **dx_playf( )**
- **dx_setparm( ), dx_getparm( )**

**Setting Order and Location for Voice Data:**

- **DX_IOTT** (Chapter 4. Voice Data Structures and Device Parameters)

---

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dx_rec() records voice data from a single channel

Retrieving and Handling Record Termination Events:

- Event Management functions (*Standard Runtime Library Programmer’s Guide for Windows NT* and Appendix A of this guide)
- ATDX_TERMMSK()
- DV_TPT (Appendix A)
permits voice data to be recorded

dx_recf()

Name: int dx_recf(chdev,fnamep,tptp,mode)

Inputs: int chdev • valid Dialogic channel device handle
char *fnamep • pointer to file to record to
DV_TPT *tptp • pointer to Termination Parameter Table Structure
unsigned short mode • recording mode bit mask for this record session

Returns: 0 if success
-1 if failure

Includes: srlib.h
dxxxlib.h

Category: Convenience
Mode: synchronous/Asynchronous

Description

The dx_recf( ) function permits voice data to be recorded from a channel to a single file. dx_recf( ) performs the same as synchronous dx_rec( ) does with a DX_IOTT structure that specified a single file. dx_recf( ) is provided as a convenient method for recording to one file without having to specify a DX_IOTT structure. dx_recf( ) opens and closes the file pointed to by fnamep while dx_rec( ) uses a DX_IOTT structure that requires the application to open the file.

Parameter Description

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>fnamep:</td>
<td>points to the file from which voice data will be recorded.</td>
</tr>
</tbody>
</table>

For information about other function arguments and other function information, see dx_rec( ).

Source Code

/***************************************************************
 * NAME: int dx_recf(devd,filep,tptp,mode)
 * DESCRIPTION: Record data to a file
 * INPUTS: devd - channel descriptor
 * tptp - TPT pointer
 * filep - ASCII string for name of file to read into
 * mode - tone initiation flag
 * OUTPUTS: Data stored in file, status in CSB pointed to by csbp
 ***************************************************************/

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dx_recf() permits voice data to be recorded

* RETURNS: 0 or -1 on error
* CALLS: open(), dx_rec(), close()
* CAUTIONS: none.
*******************************************************************************/

```c
int dx_recf(devd, filep, tptp, mode)
{
    int rval;
    DX_IOTT iott;
    /* If Async then return Error
     * Reason: IOTT's must be in scope for the duration of the record
     */
    if (mode & EV_ASYNC) {
        return(-1);
    }
    /* Open the File */
    if ((iott.io_fhandle = dx_fileopen(filep, (O_WRONLY|O_CREAT|O_TRUNC), 0666)) == -1) {
        return(-1);
    }
    /* Use dx_rec() to do the record */
    iott.io_type = IO_EOT | IO_DEV;
    iott.io_offset = (long)0;
    iott.io_length = -1;
    rval = dx_rec(devd, &iott, tptp, mode);
    if (dx_fileclose(iott.io_fhandle) == -1) {
        return(-1);
    }
    return rval;
}
```

**Example**

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

main()
{
    int chdev;
    long termtype;
    DV_TPT tpt[2];

    /* Open the channel using dx_open(). Get channel device descriptor in */
    /* chdev */
    if ((chdev = dx_open("dxxd1C1", NULL)) == -1) { /* process error */

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permits voice data to be recorded

dx_recf()

/* Set the DX_TPT structures up for MAXDTMF and MAXSIL */
dx_clrtpt(tpt,2);
tpt[0].tp_type  = IO_CONT;  /* Maximum digits */
tpt[0].tp_termno = DX_MAXDTMF;  /* terminate on the first digit */
tpt[0].tp_length = 1;  /* Use the default flags */

/* If the initial silence period before the first non-silence period
 * exceeds 4 seconds then terminate. If a silence period after the
 * first non-silence period exceeds 2 seconds then terminate. */
tpt[1].tp_type  = IO_EOT;  /* last entry in the table */
tpt[1].tp_termno = DX_MAXSIL;  /* Maximum silence */
tpt[1].tp_length = 20;  /* terminate on 2 seconds of
 * continuous silence */
tpt[1].tp_flags = TF_MAXSIL|TF_SETINIT;  /* Use the default flags and
 * initial silence flag */
tpt[1].tp_data = 40;  /* Allow 4 seconds of initial
 * silence */

if (dx_recf(chdev,"weather.vox",tpt,RM_TONE) == -1) {
    /* process error */
}

if (termtype & TM_MAXDTMF)  {
    /* process DTMF termination */
}

Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR() and
ATDV_ERRMSGP() to retrieve one of the following error reasons:

EDX_BADPARM  Invalid Parameter
EDX_BADIOTT  Invalid DX_IOTT entry
EDX_BADTPT  Invalid DX_TPT entry
EDX_BUSY  Busy executing I/O function
EDX_SYSTEM  Windows NT system error - check errno

See Also

Related Functions:

- dx_rec()
- dx_play()
\textit{dx_rect()} permits voice data to be recorded

- \textit{dx_play()}
- \textit{dx_setparm(), dx_getparm()}

Setting and Handling Record Termination:

- \textit{ATDX_TERMMSK()}
- \textit{DV_TPT (Appendix A)}
The `dx_reciotdata()` function records voice data to multiple destinations, a combination of data files, memory, or custom devices.

### Parameter Description

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev</td>
<td>channel device descriptor.</td>
</tr>
<tr>
<td>iottp</td>
<td>Pointer to <code>DX_IOTT</code> table that specifies the order and media onto which the voice data will be recorded.</td>
</tr>
<tr>
<td>tptp</td>
<td>pointer to Termination Parameter Table structure</td>
</tr>
<tr>
<td>xpbp</td>
<td>pointer to I/O transfer parameter block</td>
</tr>
</tbody>
</table>
| mode      | specifies the record mode:  
|           | `PM_TONE`  play 200 ms audible tone  
|           | `EV_SYNCH` synchronous mode  
|           | `EV_ASYNCH` asynchronous mode |
dx_reciottdata( ) records voice data to multiple destinations.

- Cautions
  1. All files specified in the DX_JOTT table will be of the file format described in DX_XPB.
  2. All files recorded to will have the data encoding and rate as described in DX_XPB.
  3. When playing or recording VOX files, the data format is specified in DX_XPB rather than through the dl_stprm() function.

- Example

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

int chdev;        /* channel descriptor */
int fd;           /* file descriptor for file to be played */
DX_IOTT iott;     /* I/O transfer table */
DX_TPT tpt;       /* termination parameter table */
DX_XPB xpb;       /* I/O transfer parameter block */
.
.
/* Open channel */
if ((chdev = dx_open("dxxxB1C1",0)) == -1) {
    printf("Cannot open channel\n");
    printf("errno = %d\n",errno);
    exit(1);
}
/* Set to terminate play on 1 digit */
tpt.tp_type   = IO_EOT;
tpt.tp_termno = DX_MAXDTMF;
tpt.tp_length = 1;
tpt.tp_flags  = TF_MAXDTMF;
/* Open file */
if ((fd = dx_fileopen("MESSAGE.VOX",O_RDWR|O_BINARY)) == -1) {
    printf("File open error\n");
    exit(2);
}
/* Set up DX_IOTT */
iott.io_fhandle = fd;
iott.io_bufp    = 0;
iott.io_offset  = 0;
iott.io_length  = -1;
iott.io_typ     = IO_DEV | IO_EOT;
/*
 * Specify VOX file format for PCM at 8KHz.
 *
 xpb.wFileFormat = FILE_FORMAT_VOX;
 xpb.wDataFormat = DATA_FORMAT_PCM;
 xpb.nSamplesPerSec = DRT_8KHZ;
 xpb.nBitsPerSample = 8;
/*
 * Wait forever for phone to ring and go offhook */
if (dx_wtring(chdev,1,DX_OFFHOOK,-1) == -1) {
    printf("Error waiting for ring - %s\n", ATDV_LASTERR(chdev));
}
```

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records voice data to multiple destinations,  \hspace{1cm} \texttt{dx_reciottdata()} \\

```
exit(3);
}
/* Play intro message */
if (dx_playwav(chdev, &tpt, "HELLO.WAV", EV_SYNC) == -1) {
    printf("Error playing file - %s\n", ATDV_ERRMSGP(chdev));
    exit(4);
}
/* Start recording */
if (dx_reciottdata(chdev, &iott, &tpt, &xpb, PM_TONE|EV_SYNC) == -1) {
    printf("Error recording file - %s\n", ATDV_ERRMSGP(chdev));
    exit(4);
}

\section{Errors}

\textbf{In asynchronous mode}, function returns immediately and a TDX_RECORD event is queued upon completion. Check \texttt{ATDX_LASTTERM()} for the termination reason. If a failure occurs, then a TDX_ERROR event will be queued. Use \texttt{ATDV_LASTERR()} to determine the reason for error.

\textbf{In synchronous mode}, if this function returns -1 to indicate failure, one of the following reasons will be contained by \texttt{ATDV_LASTERR()}:

<table>
<thead>
<tr>
<th>Equate</th>
<th>Returned When</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_BUSY</td>
<td>Channel is busy</td>
</tr>
<tr>
<td>EDX_XPBParm</td>
<td>Invalid \texttt{DX_XPB} setting</td>
</tr>
<tr>
<td>EDX_BADIOt</td>
<td>Invalid \texttt{DX_IOTT} setting</td>
</tr>
<tr>
<td>EDX_SYSTEM</td>
<td>System I/O errors</td>
</tr>
<tr>
<td>EDX_BADWAVFILE</td>
<td>Invalid WAV file</td>
</tr>
<tr>
<td>EDX_SH_BADCMD</td>
<td>Unsupported command or WAV file format</td>
</tr>
</tbody>
</table>

\section{See Also}

- \texttt{dx_reciowav()} \\
- \texttt{dx_recvox()}

\textbf{269-CD}
**dx_recvox()**  
*records voice data to a single VOX file*

**Inputs:**
- `int chdev` • valid Dialogic channel device handle
- `char *filenamep` • pointer to name of file to record to
- `DV_TPT*tptp` • pointer to Termination Parameter Table
- `DX_XPB *xpbp` • pointer to I/O Transfer Parameter Block
- `unsigned short mode` • play mode

**Returns:**
- 0 if successful
- -1 if failure

**Includes:**
- srllib.h
- dxxxlib.h

**Category:** Convenience function

**Mode:** synchronous

### Description

The **dx_recvox()** convenience function records voice data to a single VOX file. If `xpbp` is set to NULL, it will interpret the data as 6KHz linear ADPCM.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev</td>
<td>Channel device descriptor</td>
</tr>
<tr>
<td>tcbp</td>
<td>Pointer to Termination Parameter Table</td>
</tr>
<tr>
<td>filenamep</td>
<td>Pointer to name of file to record to</td>
</tr>
<tr>
<td>xpbp</td>
<td>Pointer to I/O Transfer Parameter Block (See the <code>DX_XPB</code> data structure)</td>
</tr>
<tr>
<td>mode</td>
<td>specifies the play mode: PM_TONE: play 200 ms audible tone EV_SYNC: synchronous operation (must be specified)</td>
</tr>
</tbody>
</table>

**NOTE:** Both PM_TONE and EV_SYNC can be specified by ORing the two values.
records voice data to a single VOX file  

dx_recovox()

### Cautions

When playing or recording VOX files, the data format is specified in `DX_XPB` rather than through the mode parameter of `dx_recovox()`.

### Example

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

int chdev;  /* channel descriptor */
DV_TPT tpt;  /* termination parameter table */
DX_XPB xpb;  /* I/O transfer parameter block */
.
.
./* Open channel */
if ((chdev = dx_open("dxxxB1C1",0)) == -1) {
    printf("Cannot open channel\n");
    printf("errno = %d\n",errno);
    exit(1);
}
/* Set to terminate play on 1 digit */
tpt.tp_type   = IO_EOT;
tpt.tp_termno = DX_MAXDTMF;
tpt.tp_length = 1;
tpt.tp_flags  = TF_MAXDTMF;
/* Wait forever for phone to ring and go offhook */
if (dx_wtring(chdev,1,DX_OFFHOOK,-1) == -1) {
    printf("Error waiting for ring - %s\n", ATDV_LASTERR(chdev));
    exit(3);
}
/* Start playback */
if (dx_playwav(chdev,&tpt,"HELLO.WAV",EV_SYNC) == -1) {
    printf("Error playing file - %s\n", ATDV_ERRMSGP(chdev));
    exit(4);
}
/* clear digit buffer */
dx_clrdigbuf(chdev);
/* Start 6KHz ADPCM recording */
if (dx_recovox(chdev,"MESSAGE.VOX", &tpt, NULL, PM_TONE|EV_SYNC) == -1) {
    printf("Error recording file - %s\n", ATDV_ERRMSGP(chdev));
    exit(4);
}
```

### Errors

If this function returns -1 to indicate failure, one of the following reasons will be contained by `ATDV_LASTERR()`:

<table>
<thead>
<tr>
<th>Equate</th>
<th>Returned When</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_BUSY</td>
<td>Channel is busy</td>
</tr>
</tbody>
</table>

271-CD
**dx_recvox( )**  
*records voice data to a single VOX file*

<table>
<thead>
<tr>
<th>Equate</th>
<th>Returned When</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_XPBPARAM</td>
<td>Invalid <code>DX_XPB</code> setting</td>
</tr>
<tr>
<td>EDX_BADIOITT</td>
<td>Invalid <code>DX_IOTT</code> setting</td>
</tr>
<tr>
<td>EDX_SYSTEM</td>
<td>System I/O errors</td>
</tr>
<tr>
<td>EDX_SH_BADCMD</td>
<td>Unsupported command or VOX file format</td>
</tr>
</tbody>
</table>

- See Also
  - `dx_reciotdata( )`
  - `dx_recwav( )`
records voice data to a single WAVE file

**dx_recwav()**

**Name:** SHORT dx_recwav(chdev, filenamep, tptp, xpbp, mode)

**Inputs:**
- int chdev
  • valid Dialogic channel device handle
- char *filenamep
  • pointer to name of file to record to
- DV_TPT *tptp
  • pointer to termination parameter block
- DX_XPB *xpbp
  • pointer to I/O Transfer Block
- unsigned short mode
  • record mode

**Returns:**
- 0 if successful
- -1 if failure

**Includes:**
srllib.h
dxxxlib.h

**Category:** Convenience function

**Mode:** synchronous

---

**Description**

The **dx_recwav()** convenience function records voice data to a single WAVE file. If **xpbp** is set to NULL, the function will record in 11 KHz linear 8-bit PCM. This function calls **dx_reciotdata()**.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev</td>
<td>channel device descriptor</td>
</tr>
<tr>
<td>tcbp</td>
<td>pointer to termination parameter table</td>
</tr>
<tr>
<td>filenamep</td>
<td>pointer to name of file to play</td>
</tr>
<tr>
<td>xpbp</td>
<td>pointer to I/O Transfer Parameter Block</td>
</tr>
</tbody>
</table>
| mode      | specifies the play mode:  
PM_TONE    | play 200 ms audible tone  
EV_SYNC    | synchronous operation (must be specified)  

**NOTE:** Both PM_TONE and EV_SYNC can be specified by ORing the two values.
dx_recwav() records voice data to a single WAVE file

Cautions

None.

Example

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

int chdev;     /* channel descriptor */
DV_TPT tpt;    /* termination parameter table */
DX_XPB xpb;    /* I/O transfer parameter block */

/* Open channel */
if ((chdev = dx_open("dxxxB1C1",0)) == -1) {
    printf("Cannot open channel\n");
    printf("errno = %d\n",errno);
    exit(1);
}

/* Set to terminate play on 1 digit */
tpt.tp_type = IO_EOT;
tpt.tp_termno = DX_MAXDTMF;
tpt.tp_length = 1;
tpt.tp_flags = TF_MAXDTMF;
/* Wait forever for phone to ring and go offhook */
if (dx_wtring(chdev,1,DX_OFFHOOK,-1) == -1) {
    printf("Error waiting for ring - %s\n", ATDV_LASTERR(chdev));
    exit(3);
}

/* Start playback */
if (dx_playwav(chdev,&tpt,"HELLO.WAV",EV_SYNC) == -1) {
    printf("Error playing file - %s\n", ATDV_ERRMSGP(chdev));
    exit(4);
}

/* clear digit buffer */
dx_clrdigbuf(chdev);
/* Start 11KHz PCM recording */
if (dx_recwav(chdev,"MESSAGE.WAV", &tpt, NULL, NULL, PM_TONE|EV_SYNC) == -1) {
    printf("Error recording file - %s\n", ATDV_ERRMSGP(chdev));
    exit(4);
}
```

Errors

If this function returns -1 to indicate failure, one of the following reasons will be contained by ATDV_LASTERR() :

<table>
<thead>
<tr>
<th>Equate</th>
<th>Returned When</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_BUSY</td>
<td>Channel is busy</td>
</tr>
</tbody>
</table>

274-CD
records voice data to a single WAVE file \( \text{dx\_recwav}() \)

<table>
<thead>
<tr>
<th>Equate</th>
<th>Returned When</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_XPBPARM</td>
<td>Invalid ( DX_XPB ) setting</td>
</tr>
<tr>
<td>EDX_BAD_IOTT</td>
<td>Invalid ( DX_IOTT ) setting</td>
</tr>
<tr>
<td>EDX_SYSTEM</td>
<td>System I/O errors</td>
</tr>
<tr>
<td>EDX_BAD_WAVFILE</td>
<td>Invalid WAV file</td>
</tr>
<tr>
<td>EDX_SH_BADCMD</td>
<td>Unsupported command or WAV file format</td>
</tr>
</tbody>
</table>

See Also

- \( \text{dx\_reciottdata}( ) \)
- \( \text{dx\_recvox}( ) \)
**dx_setdigbuf()**

 sets the digit buffering mode

<table>
<thead>
<tr>
<th>Name</th>
<th>int dx_setdigbuf(chdev,mode)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>int chdev</td>
</tr>
<tr>
<td></td>
<td>int mode</td>
</tr>
<tr>
<td>Returns:</td>
<td>0 if successful</td>
</tr>
<tr>
<td></td>
<td>-1 if failure</td>
</tr>
<tr>
<td>Includes:</td>
<td>srllib.h</td>
</tr>
<tr>
<td></td>
<td>dxxxlib.h</td>
</tr>
<tr>
<td>Category:</td>
<td>I/O</td>
</tr>
<tr>
<td>Mode:</td>
<td>synchronous</td>
</tr>
</tbody>
</table>

## Description

The **dx_setdigbuf()** function sets the digit buffering mode that will be used by the Voice Driver. Once the digit buffer is full (31 digits), the application may select whether subsequent digits will be ignored or will overwrite the oldest digits in the queue.

The function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid Dialogic channel device handle obtained by a call to <strong>dx_open()</strong>.</td>
</tr>
<tr>
<td>mode:</td>
<td>specifies the type of digit buffering that will be used. Mode can be:</td>
</tr>
<tr>
<td></td>
<td>• <strong>DX_DIGTRUNC</strong>  Incoming digits will be ignored if the digit buffer is full (default).</td>
</tr>
<tr>
<td></td>
<td>• <strong>DX_DIGCYCLIC</strong> Incoming digits will overwrite the oldest digits in the buffer if the buffer is full.</td>
</tr>
</tbody>
</table>

## Cautions

None.

---

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sets the digit buffering mode \texttt{dx_setdigbuf()}

\section*{Example}

\begin{verbatim}
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

int chfd;

int init_digbuf()
{
    /* open the device using dx_open, chfd has the device handle */

    /* Set up digit buffering to be Cyclic. When digit
    * queue overflows oldest digit will be overwritten
    */
    if (dx_setdigbuf(chfd, DX_DIGCYCLIC) == -1) {
        printf("Error during setdigbuf \%s\n", ATDV_ERRMSGP(chfd));
        return(1);
    }
    return(0);
}
\end{verbatim}

\section*{Errors}

If this function returns -1 to indicate failure, use \texttt{ATDV_LASTERR() \ and \ ATDV_ERRMSGP()} to retrieve one of the following error reasons:

- \texttt{EDX_BADPARM} \hspace{1cm} \textbullet \ Invalid Parameter
- \texttt{EDX_SYSTEM} \hspace{1cm} \textbullet \ Windows NT system error - check \texttt{errno}
- \texttt{EDX_TIMEOUT} \hspace{1cm} \textbullet \ Timeout limit is reached
**dx_setdigtyp( )**

**Name:** int dx_setdigtyp(chdev, dmask)

**Inputs:**
- int chdev
  - valid Dialogic channel device handle
- unsigned short dmask
  - type of digit the channel will detect

**Returns:**
- 0 if successful
- -1 if failure

**Includes:**
- srllib.h
- dxxxlib.h

**Category:** Configuration

### Description

The `dx_setdigtyp( )` function controls the types of digits the Voice channel detects.

**NOTE:** This function only applies to the standard Voice board digits (i.e., DTMF, MF, LPD). To set user-defined digits, use the `dx_addtone( )` function.

`dx_setdigtyp( )` does not clear the previously detected digits in the digit buffer.

The function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <code>dx_open( )</code>.</td>
</tr>
<tr>
<td>dmask:</td>
<td>sets the type of digits the channel will detect. More than one type of digit detection can be enabled in single function call, as shown in the function example. <strong>dmask</strong> can have one, or a combination of several, of the following values:</td>
</tr>
<tr>
<td></td>
<td>DM_DTMF: enable DTMF digits; detection (default)</td>
</tr>
<tr>
<td></td>
<td>DM_LP: enable loop pulse detection</td>
</tr>
<tr>
<td></td>
<td>DM_AP: enable audio pulse digits detection</td>
</tr>
<tr>
<td></td>
<td>DM_MF: enable MF digit detection</td>
</tr>
<tr>
<td></td>
<td>DM_DPD: enable dial pulse digit (DPD) detection</td>
</tr>
</tbody>
</table>
controls the types of digits  

**dx_setdigtyp( )**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DM_DPD2</td>
<td>enable zero train DPD detection</td>
</tr>
</tbody>
</table>

To disable digit detection, set `dmask` to NULL.

**NOTES:**
1. MF detection can only be enabled on systems with MF capability, such as D/4xD boards with MF support.
2. The digit detection type specified in `dmask` will remain valid after the channel has been closed and reopened.

`dx_setdigtyp( )` disables any digit detection enabled in a previous call to `dx_setdigtyp( )`.

**Cautions**

1. Some MF digits use approximately the same frequencies as DTMF digits (see Appendix C). Because there is a frequency overlap, if you have the incorrect kind of detection enabled, MF digits may be mistaken for DTMF digits, and vice versa. To ensure that digits are correctly detected, DTMF and MF detection should not be enabled at the same time.

**Example**

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

main()
{
    int chdev;
    /* Open Voice channel */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
        /* process error */
    }

    /* Set channel to detect DTMF and loop pulse digits */
    if (dx_setdigtyp(chdev, DM_DTMF|DM_LPD) == -1) {
        /* error routine */
    }
}
```
dx_setdigtyp() controls the types of digits

Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR() and ATDV_ERRMSGP() to retrieve one of the following error reasons:

- EDX_BADPARM • Invalid Parameter
- EDX_SYSTEM • Windows NT system error - check errno

See Also

Specifying user-defined digits:

- dx_addtone( )
**enables detection of Call Status Transition (CST) event**

**dx_setevtmsk()**

**Name:** int dx_setevtmsk(chdev, mask)

**Inputs:**
- int chdev
  - valid Dialogic channel device handle
- unsigned int mask
  - event mask of events to wait for

**Returns:**
- 0 if successful
- -1 if failure

**Includes:**
- srllib.h
- dxxxlib.h

**Category:** Call Status Transition Event

---

**Description**

The **dx_setevtmsk()** function enables detection of Call Status Transition (CST) event or group of events. This function can be used by synchronous or asynchronous applications waiting for a CST event.

**NOTE:** This function enables detection for all CST events except user-defined tone detection. See **dx_addtone()** and **dx_enbtone()** for information.

The function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <strong>dx_open()</strong>.</td>
</tr>
<tr>
<td>mask:</td>
<td>specifies the event(s) <strong>dx_getevt()</strong> will wait for. mask can have one or more of the following values:</td>
</tr>
<tr>
<td>DM_LCOFF</td>
<td>wait for loop current to be off</td>
</tr>
<tr>
<td>DM_LCON</td>
<td>wait for loop current to be on</td>
</tr>
<tr>
<td>DM_RINGS</td>
<td>wait for rings</td>
</tr>
<tr>
<td>DM_RNGOFF</td>
<td>wait for ring to drop (hang-up)</td>
</tr>
<tr>
<td>DM_SILOF</td>
<td>wait for non-silence</td>
</tr>
<tr>
<td>DM_SILON</td>
<td>wait for silence</td>
</tr>
<tr>
<td>DM_WINK</td>
<td>wait for wink to occur on an E&amp;M line</td>
</tr>
<tr>
<td>DM_DIGITS</td>
<td>turns off digit reporting and digits flag set by DM_DIGITS</td>
</tr>
<tr>
<td>DM_LCREV</td>
<td>wait for flow of current to reverse (D/41ESC and D/160SC-LS boards)</td>
</tr>
</tbody>
</table>
dx_setevtmsk() enables detection of Call Status Transition (CST) event

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>only)</td>
<td></td>
</tr>
</tbody>
</table>

When the event mask is set with DM_DIGITS, a digits flag is set that causes individual digit events to queue until this flag is turned off by the DM_DIGOFF equate. Setting the event mask for DM_DIGITS and then subsequently resetting the event mask without DM_DIGITS does not disable the queueing of digit events. Digit events will remain in the queue until collected by dx_getevt(). This queue is not affected by dx_getdig() calls. The digits flag is set by:

```c
/* Set event mask to collect digits */
if (dx_setevtmsk(chdev, DM_DIGITS) == -1) {
```

To turn off the digits flag and stop queueing digits:

```c
dx_setevtmsk(DM_DIGOFF);
```

To poll for multiple events, perform an OR operation on the bit masks of the events you want to wait for. The first enabled CST event to occur will be returned. On the D/21E, D/41E, or D/160S-LS board, when the DM-LCREV bit is OR’ed, a new event message DE_LCREV is queued when the flow of current over the line is reversed.

For configurations using the DID/120 Direct Inward Dialing board, you may need to check whether the caller has hung-up after the DID/120 board received the DNIS digits. DID systems often get the digits for a call back without ever going off hook. You can check for hang up as follows:

```c
In asynchronous mode: look for a ring off event using dx_setevtmsk(DM_RNGOFF) and dx_getevt() to get the event. After the event is received, issue a dx_stopch() to stop the current multitasking function.
```

Alternatively, you can poll using ATDX_LINEST() to determine when ring is not present (returns RLS_RING).
enables detection of Call Status Transition (CST) event \( dx_{\text{setevtmsk}}() \)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>and then issue a ( dx_{\text{stopch}}() ) to stop the current multitasking function.</td>
</tr>
</tbody>
</table>

The following table outlines the synchronous or asynchronous handling of CST events:

<table>
<thead>
<tr>
<th>Synchronous</th>
<th>Asynchronous</th>
</tr>
</thead>
<tbody>
<tr>
<td>1. Call ( dx_{\text{setevtmsk}}() ) to enable CST event(s)</td>
<td>Call ( dx_{\text{setevtmsk}}() ) to enable CST event(s)</td>
</tr>
<tr>
<td>2. Call ( dx_{\text{getevt}}() ) to wait for CST event(s). Events are returned to the ( DX_{\text{EBLK}} ) structure.</td>
<td>Use SRL to asynchronously wait for TDX_CST event(s).</td>
</tr>
<tr>
<td>3. Use ( sr_{\text{getevtdatap}}() ) to retrieve ( DX_{\text{CST}} ) structure.</td>
<td></td>
</tr>
</tbody>
</table>

**NOTE:** If DM_WINK is not specified in the \texttt{mask} parameter, and DM_RINGS is specified, a wink will be interpreted as an incoming call depending on the setting of the DXBD_R_ON parameter.

**Cautions**

1. Events are preserved between \( dx_{\text{getevt}}() \) function calls. The event that was set remains the same, until another call to \( dx_{\text{setevtmsk}}() \) or \( dx_{\text{wtring}}() \) changes it. See \( dx_{\text{wtring}}() \) for more information on how it changes the event mask.

2. If you call this function on a busy device, and specify DM_DIGITS as the \texttt{mask} argument, the function will fail.

**Example 1: Using \( dx_{\text{setevtmsk}}() \) to wait for ring events - synchronous processing**

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
```
dx_setevtmsk() enables detection of Call Status Transition (CST) event

```c
main()
{
    int chdev;
    DX_EBLK eblk;
    /* open a channel with chdev as descriptor */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) { /* process error */
        /* Set event mask to receive ring events */
        if (dx_setevtmsk(chdev, DM_RINGS) == -1) { /* error setting event */
            /* check for ring event, timeout set to 20 seconds */
            if (dx_getevt(chdev,&eblk,20) == -1) { /* error timeout */
                if(eblk.ev_event==DE_RINGS) { printf("Ring event occurred\n");
                    /* Example 2: Using dx_setevtmsk() to handle call status transition events - asynchronous processing */
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

#define MAXCHAN 24

int cst_handler();

main()
{
    int chdev[MAXCHAN];
    char *cname;
    int i, srlmode;

    /* Set SRL to run in polled mode. */
    srlmode = SR_POLLMODE;
    if (sr_setparm(SRL_DEVICE, SR_MODEID, (void *)&srlmode) == -1) { /* process error */
        for (i=0; i<MAXCHAN; i++) { /* Set chname to the channel name, e.g., dxxxB1C1, dxxxB1C2,... */
```
enables detection of Call Status Transition (CST) event \( \text{dx_setevtmsk()} \)

```c
/* Open the device using \text{dx_open(\)}. \ chdev[i] has channel device * descriptor. */
if ((chdev[i] = dx_open(chname,NULL)) == -1) {
    /* process error */
}

/* Use \text{dx_setevtmsk()} to enable call status transition events on this channel. */
if (dx_setevtmsk(chdev[i],
    DM_LCOFF|DM_LCON|DM_RINGS|DM_SILOFF|DM_SILON|DM_WINK) == -1) { /* process error */
}

/* Using \text{sr_enbhdlr()}, set up handler function to handle call status * transition events on this channel. */
if (sr_enbhdlr(chdev[i], TDX_CST, cst_handler) == -1) { /* process error */
}

/* Use \text{sr_waitevt} to wait for call status transition event. * On receiving the transition event, TDX_CST, control is transferred * to the handler function previously established using \text{sr_enbhdlr()}. */
.
.
}

int cst_handler()
{
    DX_CST *cstp;

    /* \text{sr_getevtdatap()} points to the event that caused the call status * transition. */
    *cstp = (DX_CST *)sr_getevtdatap();

    switch (cstp->cst_event) {
    case DE_RINGS:
        printf("Ring event occurred on channel %s\n", ATDX_NAMEP(sr_getevtdev()));
        break;
    case DE_WINK:
        printf("Wink event occurred on channel %s\n", ATDX_NAMEP(sr_getevtdev()));
        break;
    case DE_LCON:
        printf("Loop current ON event occurred on channel %s\n", ATDX_NAMEP(sr_getevtdev()));
        break;
    case DE_LCOFF:
        .
    .
    }

    /* Kick off next function in the state machine model. */
    .
    .
```

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dx_setevtm sk( ) enables detection of Call Status Transition (CST) event

    return 0;

■ Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR( ) and
ATDV_ERRMSGP( ) to retrieve one of the following error reasons:

    EDX_BADPARM    • Invalid Parameter
    EDX_SYSTEM     • Windows NT system error - check errno

■ See Also

CST Event Handling and Retrieval:

    • dx_getevt( ) - synchronous operation
    • sr_getevtdatap( ) - asynchronous operation (Standard Runtime Library
      Programmer’s Guide for Windows NT)
    • DX_CST data structure

Enabling User-Defined Tone Detection:

    • dx_addtone( )
sets up the amplitudes

dx_setgtdamp()

Name: void dx_setgtdamp(gtd minampl1, gtd maxampl1, gtd minampl2, gtd maxampl2)

Inputs: short int gtd_minampl1 • Minimum amplitude of the first frequency
short int gtd_maxampl1 • Maximum amplitude of the first frequency
short int gtd_minampl2 • Minimum amplitude of the second frequency
short int gtd_maxampl2 • Maximum amplitude of the second frequency

Returns: void

Includes: srllib.h
dxxxlib.h

Category: GTD Function

Description

The dx_setgtdamp() function sets up the amplitudes to be used by the general tone detection. This function must be called before calling dx_bld...( ) functions and dx_addtone( ). Once called, the values set will take effect for all dx_bld...( ) function calls.

If this function is not called, then the MINERG firmware parameters that were downloaded remain at the following settings: -42dBm for minimum amplitude and 0dBm for maximum amplitude.

Default Value | Description
---|---
GT_MIN_DEF | Default value in dB for minimum GTD amplitude that can be entered for gtd_minamp1* parameters.
GT_MAX_DEF | Default value in dB for maximum GTD amplitude that can be entered for gtd_maxamp1* parameters.

gtd minampl1: | specifies the minimum amplitude in dB of tone 1.
dx_setgtdamp()  

sets up the amplitudes

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>gtd maxamp1:</td>
<td>specifies the maximum amplitude in dB of tone 1.</td>
</tr>
<tr>
<td>gtd minamp2:</td>
<td>specifies the minimum amplitude in dB of tone 2.</td>
</tr>
<tr>
<td>gtd maxamp2:</td>
<td>specifies the maximum amplitude in dB of tone 2.</td>
</tr>
</tbody>
</table>

### Cautions

If this function is called, then the amplitudes set will take effect for all tones added afterwards. To reset the amplitudes back to the defaults, then call this function with the defines GT_MIN_DEF and GT_MAX_DEF for minimum and maximum defaults.

### Example

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
#include "voxlib.h"     /* Dialogic voice library header file */

#define TID 1;          /* Tone ID */

/*
 * Set amplitude for GTD;
 * freq1 -30dBm to 0 dBm
 * freq2 -30dBm to 0 dBm
 */
dx_setgtdamp(-30,0,-30,0);

/*
 * Build temporary simple dual tone frequency tone of
 * 950-1050 Hz and 475-525 Hz, using trailing edge detection, and
 * -30dBm to 0dBm.
 * if (dx_blddt(TID, 1000, 50, 500, 25, TN LEADING) ==-1) {
 * printf("Error building temporary tone: %d\n",dx_errno);
 * exit(3);
 * }
 */
```
sets up the amplitudes \( dx \_setgtdamp() \)

- **Errors**

  None.
dx_sethook( ) provides control of the hookswitch status

Name: int dx_sethook(chdev, hookstate, mode)
Inputs: int chdev • valid Dialogic channel device handle
int hookstate • hook state (on-hook or off-hook)
unsigned short mode • asynchronous/synchronous

Returns: 0 if successful
-1 if failure

Includes: srllib.h
dxxxlib.h

Category: Configuration
Mode: asynchronous/synchronous

Description

The dx_sethook( ) function provides control of the hookswitch status of the specified channel. A hookswitch state may be either on-hook or off-hook.

NOTE: Do not call this function for a digital T-1 SCbus configuration that includes a D/240SC, D/240SC-T1, DTI/241SC, or DTI/301SC board. Transparent signaling for SCbus digital interface devices is not supported in System Release 4.1SC.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dx_sethook( )</td>
<td>clears loop current and silence history from the channel’s buffers.</td>
</tr>
</tbody>
</table>

Asynchronous Operation

To run dx_sethook( ) asynchronously, set the mode field to EV_ASYNC. The function will return 0 to indicate it has initiated successfully, and will generate a termination event to indicate completion. Use the SRL Event Management functions to handle the termination event.

If running asynchronously, termination is indicated by a TDX_SETHOOK event. The cst_event field in the data structure will specify one of the following:

- DX_ONHOOK if the hookstate has been set to on-hook
- DX_OFFHOOK if the hookstate has been set to off-hook
provides control of the hookswitch status

\texttt{dx\_sethook( )} \\

(Use the Event Management function \texttt{sr\_getevtdatap( )} to return a pointer to the \textit{DX\_CST} structure). See Appendix A for more information about the Event Management functions.

\texttt{ATDX\_HOOKST( )} will also return the type of hookstate event.

- **Synchronous Operation**

By default, this function runs synchronously.

If running synchronously (default) \texttt{dx\_sethook( )} will return 0 when complete.

The function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>\texttt{chdev:}</td>
<td>specifies the valid channel device handle obtained when the channel was opened using \texttt{dx_open( )}.</td>
</tr>
<tr>
<td>\texttt{hookstate:}</td>
<td>forces the hookstate of the specified channel to on-hook or off-hook. The following values can be specified:</td>
</tr>
<tr>
<td>\texttt{DX_ONHOOK:}</td>
<td>set to on-hook state</td>
</tr>
<tr>
<td>\texttt{DX_OFFHOOK:}</td>
<td>set to off-hook state</td>
</tr>
<tr>
<td>\texttt{mode:}</td>
<td>specifies whether to run \texttt{dx_sethook( )} asynchronously or synchronously. Specify one of the following:</td>
</tr>
<tr>
<td>\texttt{EV_ASYNC:}</td>
<td>Run \texttt{dx_sethook( )} asynchronously.</td>
</tr>
<tr>
<td>\texttt{EV_SYNC:}</td>
<td>Run \texttt{dx_sethook( )} synchronously (default).</td>
</tr>
</tbody>
</table>

- **Cautions**

None.

- **Example 1: Using \texttt{dx\_sethook( )} in synchronous mode**

```c
#include <srllib.h>
#include <dxoolib.h>
```
dx_sethook( )

provides control of the hookswitch status

```c
#include <windows.h>

main()
{
    int chdev;
    /* open a channel with chdev as descriptor */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
        /* process error */
    }

    /* put the channel on-hook */
    if (dx_sethook(chdev,DX_ONHOOK,EV_SYNC) == -1) {
        /* error setting hook state */
    }

    /* take the channel off-hook */
    if (dx_sethook(chdev,DX_OFFHOOK,EV_SYNC) == -1) {
        /* error setting hook state */
    }
}
```

Example 2: Using dx_sethook( ) in asynchronous mode

```c
#include <stdio.h>
#include <rllib.h>
#include <dxxxlib.h>
#include <windows.h>

#define MAXCHAN 24

int sethook_hdlr();

main()
{
    int i, chdev[MAXCHAN];
    char *chnamep;
    int srlmode;

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

    for (i=0; i<MAXCHAN; i++) {
        /* Set chnamep to the channel name - e.g, dxxxB1C1, dxxxB1C2, ... */

        /* open a channel with chdev[i] as descriptor */
        if ((chdev[i] = dx_open(chnamep,NULL)) == -1) {
            /* process error */
        }
    }
}
```
provides control of the hookswitch status  

```c
/* Using sr_enbhdlr(), set up handler function to handle sethook 
* events on this channel. 
*/
if (sr_enbhdlr(chdev[i], TDX_SETHOOK, sethook_hdlr) == -1) {
  /* process error */
}

/* put the channel on-hook */
if (dx_sethook(chdev[i],DX_ONHOOK,EV_ASYNC) == -1) {
  /* error setting hook state */
}

/* Use sr_waitevt() to wait for the completion of dx_sethook(). 
* On receiving the completion event, TDX_SETHOOK, control is transferred 
* to the handler function previously established using sr_enbhdlr(). 
*/
.
.
.
.
int sethook_hdlr()
{
  DX_CST *cstp;

  /* sr_getevtdatap() points to the call status transition 
  * event structure, which contains the hook state of the 
  * device. 
  */
  cstp = (DX_CST *)sr_getevtdatap();

  switch (cstp->cst_event) {
  case DX_ONHOOK:
    printf("Channel %s is ON hook\n", ATDX_NAMEP(sr_getevtdev()));
    break;
  case DX_OFFHOOK:
    printf("Channel %s is OFF hook\n", ATDX_NAMEP(sr_getevtdev()));
    break;
  default:
    /* process error */
    break;
  }

  /* Kick off next function in the state machine model. */
  .
  .
  return 0;
}
```

### Errors

If this function returns -1 to indicate failure, use `ATDV_LASTERR()` and `ATDV_ERRMSGP()` to retrieve one of the following error reasons:

- EDX_BADPARM  •  Invalid Parameter
dx_sethook() provides control of the hookswitch status

EDX_SYSTEM

- Windows NT system error - check errno

See Also

- DX_CST structure
- sr_getevtdatap() (Standard Runtime Library Programmer’s Guide for Windows NT, and Appendix A)
- ATDX_HOOKST()
- DV_TPT (Appendix A)
allows you to set the physical parameters  
dx_setparm()

Name: int dx_setparm(dev, parm, valuep)

Inputs: int dev
• valid Dialogic channel or board device handle
unsigned long parm
• parameter type to set
void *valuep
• pointer to parameter value

Returns: 0 if successful
-1 if failure

Includes: srllib.h
dxxxlib.h

Category: Configuration

Description

The dx_setparm() function allows you to set the physical parameters of a channel or board device, such as off-hook delay, length of a pause, and flash character. Parameters can be set only one at a time. The possible values of parm are defined in (Chapter 4. Voice Data Structures and Device Parameters).

The channel must be idle (i.e., no I/O function running) when calling dx_setparm(). Board and channel resources have different parameters that can be set. Setting board parameters affects all the channels on the board. Setting channel parameters only affects the specified channel.

To set board parameters the following requirements must be met:

• the board must be open
• all channels on the board must be closed

The function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>dev:</td>
<td>specifies the valid channel or board device handle obtained when the channel or board was opened using dx_open().</td>
</tr>
<tr>
<td>parm:</td>
<td>specifies the channel or board parameter to set. Board and channel parameter defines, defaults and descriptions are listed in Section 5.2. Clearing Voice Structures</td>
</tr>
</tbody>
</table>

NOTE: The parameters set in parm will remain valid after the device has
dx_setparm() allows you to set the physical parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>valuep:</td>
<td>points to the variable that specifies the channel or board parameter to set.</td>
</tr>
</tbody>
</table>

NOTE: You must use a void* cast on the address of the parameter being sent to the driver in valuep as shown in the example.

Cautions

1. A constant cannot be used in place of valuep. The value of the parameter to be set must be placed in a variable and the address of the variable cast as void * must be passed to the function.
2. When setting channel parameters, the channel must be open and in the idle state.
3. When setting board parameters, all channels on that board must be idle.

Example

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

main()
{
    int bddev, parmval;
    /* Open the board using dx_open( ). Get board device descriptor in * bddev. */
    if ((bddev = dx_open("dxxxB1",NULL)) == -1) {
        /* process error */
    /* Set the inter-ring delay to 6 seconds (default = 8) */
    parmval = 6;
    if (dx_setparm(bddev, DXBD_R_IRD, (void *)&parmval) == -1) {
        /* process error */
    /* now wait for an incoming ring */
    ...
}
```

Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR() and ATDV_ERRMSGP() to retrieve one of the following error reasons:
allows you to set the physical parameters \textit{dx_setparm( )}

\begin{itemize}
\item EDX_BADPARM: Invalid Parameter
\item EDX_SYSTEM: Windows NT system error - check \texttt{errno}
\end{itemize}

\textbf{See Also}

\begin{itemize}
\item \texttt{dx_getparm( )}
\end{itemize}
### `dx_setsvcond()` sets adjustments and adjustment conditions

<table>
<thead>
<tr>
<th>Name:</th>
<th><code>int dx_setsvcond( chdev, numblk, svcbp)</code></th>
</tr>
</thead>
</table>
| Inputs: | `int chdev` • valid channel device handle  
`unsigned short numblk` • number of `DX_SVCB` blocks  
`DX_SVCB * svcbp` • pointer to array of `DX_SVCB` structures |
| Returns: | 0 if success  
-1 if failure |
| Includes: | `srllib.h`  
`dxxxlib.h` |
| Category: | Speed and Volume |

**Description**

The `dx_setsvcond()` function sets adjustments and adjustment conditions for all subsequent plays on the specified channel (until changed or cancelled). An *adjustment* is a modification to play-speed or play-volume that takes place due to an *adjustment condition* such as start of play, or the occurrence of an incoming digit during play. This function uses the specified channel’s speed or volume Speed/Volume Modification Table. Detailed information about these tables are contained in the *Voice Features Guide for Windows NT*.

**NOTE:** Calls to `dx_setsvcond()` are cumulative. If adjustment blocks have been set previously, calling this function adds more adjustment blocks to the list. To replace existing adjustment blocks, clear the current set of blocks using `dx_clrsvcond()` before issuing a `dx_setsvcond()`.

The following adjustments and adjustment conditions are defined in the Speed/Volume Adjustment Condition Blocks (`DX_SVCB`):

- which Speed/Volume Modification Table to use (speed or volume)
- adjustment type (increase/decrease, absolute value, toggle)
- adjustment conditions (incoming digit, beginning of play)
- level/edge sensitivity for incoming digits

See 5.2. Clearing Voice Structures for a full description of the `DX_SVCB` structure. Up to 20 DX_SVCB blocks can be specified in the form of an array.

**NOTES:** 1. This function is similar to `dx_adjsv()`. Use `dx_adjsv()` to explicitly adjust the play immediately and use `dx_setsvcond()` to adjust the
*sets adjustments and adjustment conditions*  

`dx_setsvcond()`  

play in response to specified conditions. See the description of `dx_adjsv()` for more information.

2. Whenever the play is started its speed and volume is based on the most recent modification.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained by a call to <code>dx_open()</code> .</td>
</tr>
<tr>
<td>numblk:</td>
<td>specifies the number of <code>DX_SVCB</code> blocks in the array. Set to a value between 1 and 20.</td>
</tr>
<tr>
<td>svcbp:</td>
<td>points to an array of <code>DX_SVCB</code> structures.</td>
</tr>
</tbody>
</table>

**Cautions**

1. Condition blocks can only be added to the array (up to a maximum of 20). To reset or remove any condition, you should clear the whole array, and reset all conditions if required. (e.g., If DTMF digit "1" has already been set to increase play-speed by one step, a second call that attempts to redefine "1" to the origin, will have no affect. The digit will retain its original setting).

2. The digit that causes the play adjustment will not be passed to the digit buffer, so it cannot be retrieved using `dx_getdig()` or `ATDX_BUFDIGS()`.

3. Digits that are used for play adjustment will not be used as a terminating condition. If a digit is defined as both, then the play adjustment will take priority.

4. Speed and volume control is supported on the D/21D, D/21E, D/41D, D/41E, D/41ESC, D/81A, D/121B, D/160SC-T1, D/300SC-E1 and D/320SC boards only. Do not use the Speed and Volume control functions to control speed on the D/120, D/121, or D/121A boards.

**Example**

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

/*
 * Global Variables
*/
```
dx_setsvcond() sets adjustments and adjustment conditions

DX_SVCB svcb[ 10 ] = {
    { SV_SPEEDTBL | SV_RELCURPOS,   1, '1', 0 },   /* 1 */
    { SV_SPEEDTBL | SV_ABSPOS,     -4, '2', 0 },   /* 2 */
    { SV_VOLUMETBL | SV_ABSPOS,     1, '3', 0 },   /* 3 */
    { SV_SPEEDTBL | SV_ABSPOS,      1, '4', 0 },   /* 4 */
    { SV_SPEEDTBL | SV_ABSPOS,     1, '5', 0 },   /* 5 */
    { SV_VOLUMETBL | SV_ABSPOS,     1, '6', 0 },   /* 6 */
    { SV_SPEEDTBL | SV_RELCURPOS,  -1, '7', 0 },   /* 7 */
    { SV_SPEEDTBL | SV_ABSPOS,      6, '8', 0 },   /* 8 */
    { SV_VOLUMETBL | SV_RELCURPOS, -1, '9', 0 },   /* 9 */
    { SV_SPEEDTBL | SV_ABSPOS,      10, '0', 0 },   /*10*/
};

main()
{
    int dxxxdev;

    /* Open the Voice Channel Device and Enable a Handler */
    if ( ( dxxxdev = dx_open( "dxxxB1C1", NULL ) ) == -1 ) {
        perror( "dxxxB1C1" );
        exit( 1 );
    }

    /* Set Speed and Volume Adjustment Conditions */
    if ( dx_setsvcond( dxxxdev, 10, svcb ) == -1 ) {
        printf( "Unable to Set Speed and Volume:
" );
        printf( "Adjustment Conditions\n" );
        printf( "Lasterror = %d  Err Msg = %s\n",
                ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
        dx_close( dxxxdev );
        exit( 1 );
    }

    /* Continue Processing */
    /*:*/
    /* */
    /* */

    /* Close the opened Voice Channel Device */
    if ( dx_close( dxxxdev ) != 0 ) {
        perror( "Close\n" );
        /* Terminate the Program */
        exit( 0 );
    }
}

■ Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR() and ATDV_ERRMSGP() to retrieve one of the following error reasons:

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**sets adjustments and adjustment conditions**  
`dx_setsvcond()`

<table>
<thead>
<tr>
<th>Error Code</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_BADPARM</td>
<td>Invalid Parameter</td>
</tr>
<tr>
<td>EDX_BADPROD</td>
<td>Function not supported on this board</td>
</tr>
<tr>
<td>EDX_SVADJBLKS</td>
<td>Invalid Number of Speed/Volume Adjustment blocks</td>
</tr>
<tr>
<td>EDX_SYSTEM</td>
<td>Windows NT system error - check <code>errno</code></td>
</tr>
</tbody>
</table>

### See Also

- *Setting Speed and Volume conditions:*
  - `dx_clrsvcond()`
  - `DX_SVCB` (*Chapter 4. Voice Data Structures and Device Parameters*)

- *Related to Speed and Volume:*
  - `dx_setsvmt()`
  - `dx_getcursv()`
  - `dx_getsvmt()`
  - "Speed and Volume Modification Tables" (*Voice Features Guide for Windows NT*)
  - `dx_adjsv()`
**dx_setsvmt()**

updates the speed or volume

**Name:**
int dx_setsvmt(chdev, tabletype, svmt, flag)

**Inputs:**
- int chdev
- unsigned short tabletype
- DX_SVMT * svmt
- unsigned short flag

**Returns:**
- 0 if success
- -1 if failure

**Includes:**
srllib.h
dxxxlib.h

**Category:** Speed and Volume

---

**Description**

The **dx_setsvmt** function updates the speed or volume Speed/Volume Modification Table for a channel, with the values contained in a specified DX_SVMT structure.

**NOTE:** Refer to the *Voice Features Guide for Windows NT* for a detailed description of the Speed and Volume Modification Tables, including a description of default values and refer to Section 4.1.6. DX_SVMT - speed/volume modification table structure for a description of the DX_SVMT structure.

This function can also modify the Speed or Volume Modification Table to do one of the following:

- When speed or volume adjustments reach their highest or lowest value, wrap the next adjustment to the extreme opposite value. For example, if volume reaches a maximum level during a play, the next adjustment would modify the volume to its minimum level.

- Reset the Speed/Volume Modification Table to its default values. Defaults are listed in the *Voice Features Guide for Windows NT* which describes the Speed and Volume Modification Tables in full detail.

**Parameter** | **Description**
--- | ---
chdev: | specifies the valid channel device handle obtained by a call to **dx_open**().
updates the speed or volume

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>tabletype:</td>
<td>specifies whether to retrieve the Speed or the Volume Modification Table.</td>
</tr>
<tr>
<td>SV_SPEEDTBL</td>
<td>Update the Speed Modification Table values</td>
</tr>
<tr>
<td>SV_VOLUMETBL</td>
<td>Update the Volume Modification Table values</td>
</tr>
<tr>
<td>svmtp:</td>
<td>points to the DX_SVMT structure whose contents are used to update either the speed or the volume Speed/Volume Modification Table. The structure is not used when SV_SETDEFAULT has been set in the mode parameter.</td>
</tr>
<tr>
<td>flag:</td>
<td>specifies one of the following:</td>
</tr>
<tr>
<td>SV_WRAPMOD</td>
<td>Wrap around the speed or volume adjustments that occur at the top or bottom of the Speed/Volume Modification Table.</td>
</tr>
<tr>
<td>SV_SETDEFAULT</td>
<td>Reset the table to its default values. See the Voice Features Guide for Windows NT for the default values of the table.</td>
</tr>
</tbody>
</table>

In this case, the DX_SVMT pointed to by svmtp is ignored.

**NOTE:** Set flags to 0 if you do not want to use either SV_WRAPMOD or SV_SETDEFAULT.

### Cautions

Speed and volume control are supported on D/21D, D/41D, D/21E, D/41E, D/41ESC, D/81A, D/121B, D/160SC-L, D/240SC, D/240SC-T1, D/300SC-E1 and D/320SC boards. Speed control is not supported on the D/121A board.
dx_setsvmt() updates the speed or volume

Example

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

/* Global Variables */

main()
{
    DX_SVMT svmt;
    int dxxxdev, index;

    /* Open the Voice Channel Device and Enable a Handler */
    if ( (dxxxdev = dx_open( "dxxxB1C1", NULL) ) == -1 ) {
        perror( "dxxxB1C1" );
        exit( 1 );
    }

    /* Set up the Speed/Volume Modification */
    memset( &svmt, 0, sizeof( DX_SVMT ) );
    svmt.decrease[ 0 ] = -128;
    svmt.decrease[ 1 ] = -128;
    svmt.decrease[ 2 ] = -128;
    svmt.decrease[ 3 ] = -128;
    svmt.decrease[ 4 ] = -128;
    svmt.decrease[ 5 ] = -20;
    svmt.decrease[ 6 ] = -16;
    svmt.decrease[ 7 ] = -12;
    svmt.decrease[ 8 ] = -8;
    svmt.decrease[ 9 ] = -4;
    svmt.origin = 0;
    svmt.increase[ 0 ] = 4;
    svmt.increase[ 1 ] = 8;
    svmt.increase[ 2 ] = 10;
    svmt.increase[ 3 ] = -128;
    svmt.increase[ 4 ] = -128;
    svmt.increase[ 5 ] = -128;
    svmt.increase[ 6 ] = -128;
    svmt.increase[ 7 ] = -128;
    svmt.increase[ 8 ] = -128;
    svmt.increase[ 9 ] = -128;

    /* Update the Volume Modification Table without Wrap Mode */
    if ( dx_setsvmt( dxxxdev, SV_VOLUMETBL, &svmt, 0 ) == -1 ) {
        printf( "Unable to Set the Volume" );
        printf( "Modification Table\n" );
        printf( "Lasterror = %d  Err Msg = %s\n",
            ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
        dx_close( dxxxdev );
        exit( 1 );
    }
}
```

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updates the speed or volume

\begin{verbatim}
}  
/
   * Continue Processing
   * -
   * -
*/
/
   * Close the opened Voice Channel Device
*/
if ( dx_close( dxxxdev ) != 0 ) {
    perror( "close" );
} /* Terminate the Program */
exit( 0 );
\end{verbatim}

\section*{Errors}

If this function returns -1 to indicate failure, use \texttt{ATDV\_LASTERR()} and \texttt{ATDV\_ERRMSGP()} to retrieve one of the following error reasons:

- \texttt{EDX\_BADPARM} • Invalid Parameter
- \texttt{EDX\_BADPROD} • Function not supported on this board
- \texttt{EDX\_NONZEROSIZE} • Reset to Default was Requested but size was non-zero
- \texttt{EDX\_SPDVOL} • Must Specify either \texttt{SV\_SPEEDTBL} or \texttt{SV\_VOLUMETBL}
- \texttt{EDX\_SVMTRANGE} • An Entry in \texttt{DX\_SVMT} was out of Range
- \texttt{EDX\_SVMTSIZE} • Invalid Table Size Specified
- \texttt{EDX\_SYSTEM} • Windows NT system error - check \texttt{errno}

\section*{See Also}

- \texttt{dx\_adjsv( )}
- \texttt{dx\_getcursv( )}
- \texttt{dx\_getsvmt( )}
- "Speed and Volume Modification Tables" (\textit{Voice Features Guide for Windows NT})
- \texttt{DX\_SVMT} (Chapter 4. Voice Data Structures and Device Parameters)
**dx_setuio()**  
*allows an application to install a user I/O routine*

<table>
<thead>
<tr>
<th>Name:</th>
<th>int dx_setuio(uioblk)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inputs:</td>
<td>uioblk</td>
</tr>
<tr>
<td>Returns:</td>
<td>0 if success</td>
</tr>
<tr>
<td></td>
<td>-1 if failure</td>
</tr>
<tr>
<td>Includes:</td>
<td>srlib.h</td>
</tr>
<tr>
<td></td>
<td>dxxlib.h</td>
</tr>
<tr>
<td>Category:</td>
<td>miscellaneous function</td>
</tr>
</tbody>
</table>

**Description**

The `dx_setuio()` function allows an application to install a user I/O routine `read()`, `write()`, and `lseek()` functions. These functions are then used by the `dx_play()` and `dx_rec()` functions to read and/or write to nonstandard storage media.

The application provides the addresses of user-defined `read()`, `write()` and, optionally, `lseek()` functions by initializing the DX_UIO structure. The application then installs the functions by invoking the `dx_setuio()` function.

The application can override the standard I/O functions on a file-by-file basis by setting the IO_UIO flag in the `io_type` field of the DX_IOTT structure (see Chapter 4. Voice Data Structures and Device Parameters for details).

**NOTE:** The IO_UIO flag must be ORed with the IO_DEV flag for this feature to function properly.

When using the `dx_setuio()` function to record, a user-defined `write()` function must be provided. User-defined `read()` and `lseek()` functions are optional.

When using the `dx_setuio()` function to play, a user-defined `read()` function must be provided. User-defined `write()` and `lseek()` functions are optional.

**Cautions**

In order for the application to work properly, the user-provided functions must conform to standard I/O function semantics.

---

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allows an application to install a user I/O routine  

**Example**

```c
#include <stdio.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
#include "VOXLIB.H" /* Dialogic voice library header file */

DX_UIO myio; /* user definable I/O structure */

/* User defined I/O functions */
int my_read9(fd,ptr,cnt)
  int fd;
  char * ptr;
  unsigned cnt;
  printf("My read\n");
  return(read(fd,ptr,cnt));

/* my write function */
int my_write(fd,ptr,cnt)
  int fd;
  char * ptr;
  unsigned cnt;
  printf("My write \n");
  return(write(fd,ptr,cnt));

/* my seek function */
long my_seek(fd,offset,whence)
  int fd;
  long offset;
  int whence;
  printf("My seek\n");
  return(lseek(fd,offset,whence));

void main(argc,argv)
  int argc;
  char *argv[];
{
  /* Other initialization */

  DX_UIO uioblk;
  /* Initialize the I/O structure */
  uioblk.u_read=my_read;
  uioblk.u_write=my_write;
  uioblk.u_seek=my_seek;
  /* Install my I/O routines */
  dx_setuio(uioblk);
  vodat_fd = dx_fileopen("JUNK.VOX",O_RDWR|O_BINARY);
  /*This block uses standard I/O functions */
  iott->io_type = IO_DEV|IO_CONT
  iott->io_fhandle = vodat_fd;
  iott->io_offset = 0;
  iott->io_length = 20000;
  /*This block uses my I/O functions */
  iott++;
}
```
`dx_setuio()` allows an application to install a user I/O routine

```c
iott->io_type = IO_DEV|IO_UIO|IO_CONT
iott->io_fhandle = vodat_fd;
iott->io_offset = 20001;
iott->io_length = 20000;
/*This block uses standard I/O functions */
iott++
iott->io_type = IO_DEV|IO_CONT
iott->io_fhandle = vodat_fd;
iott->io_offset = 20002;
iott->io_length = 20000;
/*This block uses my I/O functions */
iott->io_type = IO_DEV|IO_UIO|IO_EOT
iott->io_fhandle = vodat_fd;
iott->io_offset = 10003;
iott->io_length = 20000;
devhandle = dx_open("dxxxB1C1", NULL);
dx_sethook(devhandle, DX_OFFHOOK, EV_SYNC)
dx_string(devhandle, i, DX_OFFHOOK, EV_SYNC);
dx_clrdigbuf;
    if(dx_rec(devhandle, iott, (DX_TPT*)NULL, RF_TONE|EV_SYNC) == -1) {
        perror("**");
        exit(1);
    }
dx_clrdigbuf(devhandle);
    if(dx_play(devhandle, iott, (DX_TPT*)EV_SYNC) == -1 {
        perror("**");
        exit(1);
    }
dx_close(devhandle);
```

### Errors

None.
The `dx_stopch()` function forces termination of currently active I/O functions on a channel. It forces a channel in the busy state to become idle. If the channel specified in `chdev` already is idle, `dx_stopch()` has no effect and will return a success.

Running this function asynchronously will initiate the `dx_stopch()` without affecting processes on other channels.

Running this function synchronously within a process does not block other processing. Other processes continue to be serviced.

When an I/O function terminates due to a `dx_stopch()` being issued on the channel, the termination reason returned by `ATDX_TERMMSK()` will be TM_USRSTOP. (If running the `dx_dial()` function with Call Analysis, you can call `ATDX_CPTERM()` to determine the reason for Call Analysis termination. This will return CR_STOPD if Call Analysis stopped due to a `dx_stopch()`.)

The function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
</table>
| chdev:    | specifies the valid channel device handle obtained when the channel was opened using `dx_open()`.
| mode:     | Set to EV_ASYNC. The stop will be issued, but the driver does not `sleep` and wait for the channel to become idle before `dx_stopch()` returns. |
**dx_stopch()** forces termination of currently active I/O functions

### Cautions

1. **dx_stopch()** will have no effect on a channel that has either of the following functions issued:
   - **dx_dial()** without Call Analysis enabled
   - **dx_wink()**

   The functions will continue to run normally, **dx_stopch()** will return a success. For **dx_dial()**, the digits specified in the `dialstrp` parameter will still be dialed.

2. If **dx_stopch()** is called on a channel dialing with Call Analysis enabled, the Call Analysis process will stop but dialing will be completed. Any Call Analysis information collected prior to the stop will be returned by Extended Attribute functions.

3. If an I/O function terminates (due to another reason) before **dx_stopch()** is issued, the reason for termination will not indicate **dx_stopch()** was called.

4. When calling **dx_stopch()** from a signal handler, **mode** must be set to EV_ASYNC.

### Example

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

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

    /* Open the channel using dx_open(). Get channel device descriptor in *chdev. */
    if ((chdev = dx_open("dxxxB1C1",NULL)) == -1) {
        /* process error */
    }

    /* continue processing */
    .
    .
    .
}
```

---

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forces termination of currently active I/O functions  

/* Force the channel idle. The I/O function that the channel is executing will be terminated, and control passed to the handler function previously enabled, using sr_enbhdlr(), for the termination event corresponding to that I/O function.
* In the asynchronous mode, dx_stopch() returns immediately, without waiting for the channel to go idle.
*/
if (dx_stopch(chdev, EV_ASYNC) == -1) {
    /* process error */
}

Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR() and ATDV_ERRMSGP() to retrieve one of the following error reasons:

- EDX_BADParm
- EDX_SYSTEM

See Also

Related I/O functions:

- dx_dial()
- dx_getdig()
- dx_play()
- dx_playf()
- dx_playtone()
- dx_rec()
- dx_recf()
- dx_wink()

Retrieving I/O termination reason due to dx_stopch():

- ATDX_TERMMSK()
- ATDX_CPTERM() - dx_dial() with Call Analysis
The `dx_wink()` function generates an outbound wink on the specified channel. A wink from a Voice board is a momentary rise of the A signaling bit, which corresponds to a wink on an E&M line. This is used for signaling T-1 spans. A wink’s typical duration of 150 to 250 milliseconds used for communication purposes between the called and calling stations.

**NOTE:** Do not call this function on a non-E&M line or for a SCbus T-1 digital interface device on a D/240SC or a D/240SC-T1 board. Transparent signaling for SCbus digital interface devices is not supported in System Release 4.1SC. See the Digital Network Interface Software Reference for Windows NT for information about E&M lines.

**Asynchronous Operation**

To run this function asynchronously set the `mode` field to EV_ASYNC. When running asynchronously, this function will return 0 to indicate it has initiated successfully, and will generate a TDX_WINK termination event to indicate completion. Use the SRL Event Management functions to handle the termination event. See Appendix A for more information about the Event Management functions.
generates an outbound wink \textit{dx\_wink( )}

\section*{Synchronous Operation}

By default, this function runs synchronously, and will return a 0 to indicate that it has completed successfully.

The function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>\texttt{chdev:}</td>
<td>specifies the valid channel device handle obtained when the channel was opened using \texttt{dx_open( ).}</td>
</tr>
<tr>
<td>\texttt{mode:}</td>
<td>specifies whether to run \texttt{dx_wink( )} asynchronously or synchronously. Specify one of the following:</td>
</tr>
</tbody>
</table>

\begin{itemize}
  \item \texttt{EV\_ASYNC:} Run \texttt{dx\_wink( )} asynchronously.
  \item \texttt{EV\_SYNC:} Run \texttt{dx\_wink( )} synchronously (default).
\end{itemize}

\textbf{NOTES:} 1. The \texttt{dx\_wink( )} function is supported on a T-1 E&M line connected to a DTI/101 board. In addition, the \texttt{dx\_wink( )} function is supported on the DTI/211 board in transparent mode.

2. The channel must be on-hook when \texttt{dx\_wink( )} is called.

3. All values referenced for this function are subject to a 10 ms clocking resolution. Actual values will be in a range: \((\text{parameter value} - 9 \text{ ms}) < \text{actual value} < \text{parameter value}\)

\section*{Setting Delay Prior to Wink}

The default delay prior to generating the outbound wink is 150 ms. To change the delay, use the \texttt{dx\_setparm( )} function to enter a value for the DXCH\_WINKDLY parameter where:

\texttt{delay} = \text{the value entered x 10 ms}

The syntax of the function is:

\begin{verbatim}
int delay;
delay = 15;
dx_setparm(dev,DXCH\_WINKDLY,(void*)&delay)
\end{verbatim}

If \texttt{delay} = 15, then DXCH\_WINKDLY = 15 x 10 or 150 ms.

\textit{313-CD}
Setting Wink Duration

The default outbound wink duration is 150 ms. To change the wink duration, use the `dx_setparm()` function to enter a value for the DXCH_WINKLEN parameter where:

\[ \text{duration} = \text{the value entered} \times 10 \text{ ms} \]

The syntax of the function is:

```c
int duration;
duration = 15;
dx_setparm(dev, DXCH_WINKLEN, (void *)&duration)
```

If `duration = 15`, then `DXCH_WINKLEN = 15 \times 10 = 150` ms.

Receiving an Inbound Wink

**NOTE:** The inbound wink duration must be between the values set for DXCH_MINRWINK and DXCH_MAXRWINK. The default value for DXCH_MINRWINK is 100 ms, and the default value for DXCH_MAXRWINK is 200 ms. Use the `dx_setparm()` function to change the minimum and maximum allowable inbound wink duration.

To receive an inbound wink on a channel:

1. Using the `dx_setparm()` function, set the off-hook delay interval (DXBD_OFFHDLY) parameter to 1 so that the channel is ready to detect an incoming wink immediately upon going off hook.
2. Using the `dx_setevtmsk()` function, enable the DM_WINK event.

**NOTE:** If DM_WINK is not specified in the mask parameter of the `dx_setevtmsk()` function, and DM_RINGS is specified, a wink will be interpreted as an incoming call.

A typical sequence of events for an inbound wink is:

1. The application calls the `dx_sethook()` function to initiate a call by going off hook.
2. When the incoming call is detected by the Central Office, the CO responds by sending a wink to the board.
generates an outbound wink \texttt{dx\_wink()} 

3. When the wink is received successfully, a DE\_WINK event is sent to the application.

\section*{Cautions}

Make sure the channel is on-hook when \texttt{dx\_wink()} is called.

\section*{Example 1: Using \texttt{dx\_wink()} in synchronous mode.}

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

main()
{
    int chdev;
    DV_TPT tpt;
    DV_DIGIT digitp;
    char buffer[8];

    /* open a channel with chdev as descriptor */
    if ((chdev = dx_open("dxxxB1C1", NULL)) == -1)
    {
        /* process error */
    }

    /* set hookstate to on-hook and wink */
    if (dx_sethook(chdev, DX_ONHOOK, EV_SYNC) == -1)
    {
        /* process error */
    }

    if (dx_wink(chdev, EV_SYNC) == -1)
    {
        /* error winking channel */
    }

dx_clrtpt(&tpt, 1);

    /* set up DV\_TPT */
    tpt.tp_type = IO_EOT;         /* only entry in the table */
    tpt.tp_termno = DX_MAXDTMF;   /* Maximum digits */
    tpt.tp_length = 1;           /* terminate on the first digit */
    tpt.tp_flags = TF_MAXDTMF;   /* Use the default flags */

    /* get digits while on-hook */
    if (dx_getdig(chdev, &tpt, &digitp, EV_SYNC) == -1)
    {
        /* error getting digits */
    }

    /* now we can go off-hook and continue */

    if (dx_sethook(chdev, DX_OFFHOOK, EV_SYNC) == -1)
    {
        /* process error */
    }
```

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dx_wink( ) generates an outbound wink

Example 2: Using dx_wink( ) in asynchronous mode.

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

#define MAXCHAN 24

int wink_handler();

main()
{
    int i, chdev[MAXCHAN];
    char *chnamep;
    int srlmode;

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

    for (i=0; i<MAXCHAN; i++) {
        /* Set chnamep to the channel name - e.g., dxxxB1C1 */
        /* open the channel with dx_open(). Obtain channel device */
        /* descriptor in chdev[i] */
        if ((chdev[i] = dx_open(chnamep, NULL)) == -1) {
            /* process error */
        }

        /* Using sr_enbhdlr(), set up handler function to handle wink */
        /* completion events on this channel. */
        if (sr_enbhdlr(chdev[i], TDX_WINK, wink_handler) == -1) {
            /* process error */
        }

        /* Before issuing dx_wink(), ensure that the channel is onhook, */
        /* else the wink will fail. */
        if (dx_sethook(chdev[i], DX_ONHOOK, EV_ASYNC) == -1) {
            /* error setting channel on-hook */
        }

        /* Use sr_waitev() to wait for the completion of dx_sethook(). */
        if (dx_wink(chdev[i], EV_ASYNC) == -1) {
            /* error winking channel */
        }
    }
}
```

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generates an outbound wink

dx_wink( )

/* Use sr_waitevt() to wait for the completion of wink. * On receiving the completion event, TDX_WINK, control is transferred * to the handler function previously established using sr_enbhdlr(). */

int wink_handler()
{
    printf("wink completed on channel %s\n", ATDX_NAMEP(sr_getevtdev()));
    return 0;
}

Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR() and ATDV_ERRMSGP() to retrieve one of the following error reasons:

<table>
<thead>
<tr>
<th>Error Code</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_BADPARM</td>
<td>Invalid Parameter</td>
</tr>
<tr>
<td>EDX_SYSTEM</td>
<td>Windows NT system error - check errno</td>
</tr>
</tbody>
</table>

See Also

Related Functions:

- dx_setparm( )
- dx_getparm( )

Handling and Retrieving dx_wink Termination Events:

- Event Management functions (Standard Runtime Library Programmer’s Guide for Windows NT and Appendix A)
- DV_TPT
- ATDX_TERMMSK( )

Handling outbound winks:

- dx_wstring( )

Handling inbound winks:

- dx_setevtmask()
- dx_sethook()
- DX_CST (Chapter 4. Voice Data Structures and Device Parameters)
dx_wink( ) generates an outbound wink

- dx_getevt( ) - synchronous
- EV_EBLK - synchronous applications (Chapter 4. Voice Data Structures and Device Parameters)
- sr_getevtdatap() - asynchronous (Standard Runtime Library Programmer’s Guide for Windows NT and Appendix A)
The `dx_wring()` function waits for a specified number of rings and sets the channel to on-hook or off-hook after the rings are detected. Using `dx_wring()` is equivalent to using `dx_setevtmsk()`, `dx_getevt()`, and `dx_sethook()` to wait for a ring. When `dx_wring()` is called, the specified channel’s event is set to DM_RINGS.

**NOTE:** Do not call this function for a digital T-1 SCbus configuration that includes a D/240SC, D/240SC-T1, or DTI/241SC board. Transparent signaling for SCbus digital interface devices is not supported in System Release 4.1SC.

The function parameters are defined as follows:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <code>dx_open()</code></td>
</tr>
<tr>
<td>nrings:</td>
<td>specifies the number of rings to wait for before setting the hook state.</td>
</tr>
<tr>
<td>hstate:</td>
<td>sets the hookstate of the channel after the number of rings specified in <code>nrings</code> are detected. <code>hstate</code> can have either of the following values:</td>
</tr>
</tbody>
</table>

DX_ONHOOK: channel remains on-hook when `nrings` number of rings are detected
**dx_wtring()** waits for a specified number of rings

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DX_OFFHOOK</td>
<td>channel goes off-hook when <em>nrings</em> number of rings are detected</td>
</tr>
<tr>
<td><strong>timeout:</strong></td>
<td>specifies the maximum length of time in tenths of seconds to wait for a ring.</td>
</tr>
<tr>
<td></td>
<td>timeout can have one of the following values:</td>
</tr>
<tr>
<td></td>
<td># of seconds: maximum length of time to wait for a ring.</td>
</tr>
<tr>
<td></td>
<td>-1: <em>dx_wtring()</em> waits forever and never times out.</td>
</tr>
<tr>
<td></td>
<td>0: <em>dx_wtring()</em> returns -1 immediately if a ring event does not already exist.</td>
</tr>
</tbody>
</table>

**Cautions**

1. *dx_wtring()* changes the event enabled on the channel to DM_RINGS. For example, "process A" issues *dx_setevtmsk()* to enable detection of another type of event, e.g., DM_SILON, on channel one. If "process B" issues *dx_wtring()* on channel one, then process A will now be waiting for a DM_RINGS event since process B has reset the channel event to DM_RINGS with *dx_wtring()*.

2. A channel can detect rings immediately after going on hook. Rings may be detected during the time interval between *dx_sethook()* and *dx_wtring()*.

   **NOTE:** If the number of rings detected before *dx_wtring()* returns is equal to or greater than *nrings*, *dx_wtring()* will not terminate. This may cause the application to miss calls that are already coming in when the application is first started.

3. Do not use the Windows NT *sigset()* system call with SIGALRM while waiting for rings.

**Example**

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

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```c
#include <windows.h>

main() {
    int chdev; /* channel descriptor */

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

    /* Wait for two rings on this channel - no timeout */
    if (dx_wtring(chdev, 2, DX_OFFHOOK, -1) == -1) {
        /* process error */
    }
}
```

## Errors

If this function returns -1 to indicate failure, use `ATDV_LASTERR()` and `ATDV_ERRMSGP()` to retrieve one of the following error reasons:

- **EDX_BADPARM**: Invalid Parameter
- **EDX_SYSTEM**: Windows NT system error - check `errno`
- **EDX_TIMEOUT**: Timeout limit is reached

## See Also

- `dx_setevtmsk()`
- `dx_getevt()`
- `dx_sethook()`
- `DX_EBLK` *(Chapter 4. Voice Data Structures and Device Parameters)*
**r2_creatfsig()** defines and enables leading edge detection

**Name:**
int r2_creatfsig(chdev,forwardsig)

**Inputs:**
- int chdev
  - channel device handle
- int forwardsig
  - group I/II forward signal

**Returns:**
- 0 if success
- -1 if failure

**Includes:**
srlib.h
dxxxlib.h

**Category:**
R2MF Convenience

---

**Description**

*r2_creatfsig()* is a convenience function that defines and enables leading edge detection of an R2MF forward signal on a channel.

**NOTE:** This function calls the *dx_blddt()* function to create the template.

User-defined tone IDs 101 through 115 are used by this function.

For detailed information about R2MF signaling see the *Voice Features Guide for Windows NT*.

**Parameter** | **Description**
--- | ---
**chdev:** specifies the valid channel device handle obtained when the channel was opened using *dx_open()*.

**forwardsig:** specifies the name of a Group I or Group II forward signal which provides the tone ID for detection of the associated R2MF tone (or tones). Set to R2_ALLSIG to enable detection of all 15 tones or set to one of the following defines:

<table>
<thead>
<tr>
<th>Group I</th>
<th>Group II</th>
<th>Associated Tone ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIGI_1</td>
<td>SIGII_1</td>
<td>101</td>
</tr>
<tr>
<td>SIGI_2</td>
<td>SIGII_2</td>
<td>102</td>
</tr>
<tr>
<td>SIGI_3</td>
<td>SIGII_3</td>
<td>103</td>
</tr>
<tr>
<td>SIGI_4</td>
<td>SIGII_4</td>
<td>104</td>
</tr>
<tr>
<td>SIGI_5</td>
<td>SIGII_5</td>
<td>105</td>
</tr>
<tr>
<td>SIGI_6</td>
<td>SIGII_6</td>
<td>106</td>
</tr>
</tbody>
</table>
defines and enables leading edge detection

Parameter Description

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIGI_7</td>
<td>SIGII_7</td>
</tr>
<tr>
<td>SIGI_8</td>
<td>SIGII_8</td>
</tr>
<tr>
<td>SIGI_9</td>
<td>SIGII_9</td>
</tr>
<tr>
<td>SIGI_10</td>
<td>SIGII_10</td>
</tr>
<tr>
<td>SIGI_11</td>
<td>SIGII_11</td>
</tr>
<tr>
<td>SIGI_12</td>
<td>SIGII_12</td>
</tr>
<tr>
<td>SIGI_13</td>
<td>SIGII_13</td>
</tr>
<tr>
<td>SIGI_14</td>
<td>SIGII_14</td>
</tr>
<tr>
<td>SIGI_15</td>
<td>SIGII_15</td>
</tr>
</tbody>
</table>

NOTE: Either the Group I or the Group II define can be used to specify the forward signal, because the Group I and Group II defines correspond to the same set of 15 forward signals, and the same user-defined tones are used for Group I and Group II.

■ Cautions

1. The channel must be idle when calling this function.

2. Prior to creating the R2MF tones on a channel, you should delete any previously created user-defined tones (including non-R2MF tones) to avoid getting an error for having too many tones enabled on a channel.

3. This function creates R2MF tones with user-defined tone IDs from 101 to 115, and you should reserve these tone IDs for R2MF. If you attempt to create a forward signal tone with this function and you previously created a tone with the same tone ID, an invalid tone ID error will occur.

4. Maximum number of user-defined tones is on a per board basis.

■ Example

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    int dxxxdev;
```
/** * Open the Voice Channel Device and Enable a Handler */ if ( ( dxxxdev = dx_open( "dxxxBiCl", NULL ) ) == -1 ) { perror( "dxxxBiCl" ); exit( 1 ); } /* * Create all forward signals */ if ( r2_creatfsig( dxxxdev, R2_ALLFSIG ) == -1 ) { printf( "Unable to Create the Forward Signals\n" ); printf( "Lasterror = %d  Err Msg = %s\n", ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) ); dx_close( dxxxdev ); exit( 1 ); } /* * Continue Processing */ /* */ /* */ /* */ /* Close the opened Voice Channel Device */ /* */ if ( dx_close( dxxxdev ) != 0 ) { perror( "close" ); } /* Terminate the Program */ exit( 0 ); */

## Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR() and ATDV_ERRMSGP() to retrieve one of the following error reasons:

- EDX_BADPARM • Invalid parameter
- EDX_BADPROD • Function not supported on this board
- EDX_SYSTEM • Windows NT System error - check errno
- EDX_TONEID • Invalid tone template ID
- EDX_MAXTMPLT • Maximum number of user-defined tones for the board
- EDX_INVSUBCMD • Invalid sub-command
- EDX_FREQDET • Invalid tone frequency
- EDX_CADENCE • Invalid cadence component value

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defines and enables leading edge detection

r2_creatfsig()

EDX.ASCII
• Invalid ASCII value in tone template description

EDX_DIGTYPE
• Invalid Dig_Type value in tone template description

See Also

• r2_playbsig()
• dx_addtone()
• dx_blddt()
• "R2MF Signaling" - (Voice Features Guide for Windows NT)
The `r2_playbsig()` function plays a specified backward R2MF signal on the specified channel until a tone-off event is detected for the specified forward signal.

The `r2_playbsig()` function is a convenience function that plays a tone and controls the timing sequence required by the R2MF compelled signaling procedure.

Compelled signaling sends each signal, until it is responded to by a return signal, which in turn is sent until responded to by the other party. See the Voice Features Guide for Windows NT for more information about R2MF Compelled signaling.

**NOTE:** This function calls the `dx_playtone()` function to play the tone.

### Asynchronous Operation

1. Enable forward signal detection using `r2_creatfsig()`.
2. Use SRL to asynchronously wait for TDX_CST event(s).
3. Use `sr_getevtdatap()` to retrieve the `DX_CST` structure, which will contain a DE_TONEON event in the `cst_event` field.
4. Determine which forward signal was detected by matching the tone ID returned `cst_data` field (from 101 to 115) with the forward signal Group I or Group II defines (see `forwardsig` argument description for a list of the forward signal defines).
5. Decide which backward signal should be played in response to the forward signal.

6. Use the `r2_playbsig()` function to play the desired backward signal.

7. `r2_playbsig()` will terminate automatically when a tone-off event is detected. There is a 60-second default duration for playing the backward signal. If the forward signal tone-off is not detected within 60 seconds, the backward signal will terminate with a TDX_PLAYTONE event, and `ATDX_TERMMSK()` will return TM_MAXTIME.

### Synchronous Operation

8. Enable forward signal detection using `r2_creatfsig()`.

9. Call `dx_getevt()` to wait for a DX_TONEON event. Events are returned in the `DX_EBLK` structure.

10. Determine which forward signal was detected by matching the tone ID contained in the `ev_data` field (from 101 to 115) with the forward signal Group I or Group II defines (see `forwardsig` argument description for a list of the forward signal defines).

11. Decide which backward signal should be played in response to the forward signal.

12. Use the `r2_playbsig()` function to play the desired backward signal.

13. `r2_playbsig()` will terminate automatically when a tone-off event is detected. There is a 60-second default duration for playing the backward signal. If the forward signal tone-off is not detected within 60 seconds, the backward signal will terminate, and `ATDX_TERMMSK()` will return TM_MAXTIME.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>chdev:</td>
<td>specifies the valid channel device handle obtained when the channel was opened using <code>dx_open()</code></td>
</tr>
<tr>
<td>backwardsig:</td>
<td>specifies the name of a Group A or Group B backward signal to play. Set to one of the defines in Group A or one of the defines in Group B:</td>
</tr>
</tbody>
</table>
*r2_playbsig( )*  
*plays a specified backward R2MF signal*

### Parameter Description

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Specify one of:</strong></td>
<td></td>
</tr>
<tr>
<td>Group A Group B</td>
<td>Associated Tone ID</td>
</tr>
<tr>
<td>SIGA_1</td>
<td>SIGB_1</td>
</tr>
<tr>
<td>SIGA_2</td>
<td>SIGB_2</td>
</tr>
<tr>
<td>SIGA_3</td>
<td>SIGB_3</td>
</tr>
<tr>
<td>SIGA_4</td>
<td>SIGB_4</td>
</tr>
<tr>
<td>SIGA_5</td>
<td>SIGB_5</td>
</tr>
<tr>
<td>SIGA_6</td>
<td>SIGB_6</td>
</tr>
<tr>
<td>SIGA_7</td>
<td>SIGB_7</td>
</tr>
<tr>
<td>SIGA_8</td>
<td>SIGB_8</td>
</tr>
<tr>
<td>SIGA_9</td>
<td>SIGB_9</td>
</tr>
<tr>
<td>SIGA_10</td>
<td>SIGB_10</td>
</tr>
<tr>
<td>SIGA_11</td>
<td>SIGB_11</td>
</tr>
<tr>
<td>SIGA_12</td>
<td>SIGB_12</td>
</tr>
<tr>
<td>SIGA_13</td>
<td>SIGB_13</td>
</tr>
<tr>
<td>SIGA_14</td>
<td>SIGB_14</td>
</tr>
<tr>
<td>SIGA_15</td>
<td>SIGB_15</td>
</tr>
</tbody>
</table>

### forwardsig:

specifies the name of the Group I or Group II forward signal for which a tone-on event was detected, and for which a tone-off event will terminate this function. Set to one of defines from Group I or one of the defines from Group II:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Specify one of:</strong></td>
<td></td>
</tr>
<tr>
<td>Group I Group II</td>
<td>Associated Tone ID</td>
</tr>
<tr>
<td>SIGI_1</td>
<td>SIGII_1</td>
</tr>
<tr>
<td>SIGI_2</td>
<td>SIGII_2</td>
</tr>
<tr>
<td>SIGI_3</td>
<td>SIGII_3</td>
</tr>
<tr>
<td>SIGI_4</td>
<td>SIGII_4</td>
</tr>
<tr>
<td>SIGI_5</td>
<td>SIGII_5</td>
</tr>
<tr>
<td>SIGI_6</td>
<td>SIGII_6</td>
</tr>
<tr>
<td>SIGI_7</td>
<td>SIGII_7</td>
</tr>
<tr>
<td>SIGI_8</td>
<td>SIGII_8</td>
</tr>
<tr>
<td>SIGI_9</td>
<td>SIGII_9</td>
</tr>
<tr>
<td>SIGI_10</td>
<td>SIGII_10</td>
</tr>
<tr>
<td>SIGI_11</td>
<td>SIGII_11</td>
</tr>
<tr>
<td>SIGI_12</td>
<td>SIGII_12</td>
</tr>
<tr>
<td>SIGI_13</td>
<td>SIGII_13</td>
</tr>
<tr>
<td>SIGI_14</td>
<td>SIGII_14</td>
</tr>
</tbody>
</table>
plays a specified backward R2MF signal \( r_2_{\text{playbsig}}() \)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIGI_15</td>
<td>SIGII_15</td>
</tr>
</tbody>
</table>

The following procedure describes how to use the \( r_2_{\text{playbsig}}() \) function:

### Example

```c
#include <stdio.h>
#include <errno.h>
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>

main()
{
    int dxxxdev;

    /*
     * Open the Voice Channel Device and Enable a Handler
     */
    if ( (dxxxdev = dx_open( "dxxxB1C1", NULL ) ) == -1 ) {
        perror( "dxxxB1C1" );
        exit( 1 );
    }

    /*
     * Create all forward signals
     */
    if ( r2_creatfsig( dxxxdev, R2_ALLFSIG ) == -1 ) {
        printf( "Unable to Create the Forward Signals\n" );
        printf( "Lasterror = %d  Err Msg = %s\n",
                ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
        dx_close( dxxxdev );
        exit( 1 );
    }

    /*
     * Detect an incoming call using dx_wtring()
     * Enable the detection of all forward signals using
     * dx_enbtone(). In this example, only the first
     * forward signal will be enabled.
     */
    if (dx_enbtone( dxxxdev, SIGI\_1, IM\_TONEMON | IM\_TONEOFF ) == -1 ) {
        printf( "Unable to Enable Detection of Tone %s\n", SIGI\_1 );
        printf( "Lasterror = %d  Err Msg = %s\n",
                ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
        dx_close( dxxxdev );
        exit( 1 );
    }
```

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r2_playbsig( )

plays a specified backward R2MF signal

if ( r2_playbsig( dxxxdev, SIGA_1, SIGI_1, EV_SYNC ) == -1 ) {
    printf( "Unable to generate the backward signals\n" );
    printf( "Lasterror = %d  Err Msg = %s\n",
            ATDV_LASTERR( dxxxdev ), ATDV_ERRMSGP( dxxxdev ) );
    dx_close( dxxxdev );
    exit( 1 );
}

if ( dx_close( dxxxdev ) != 0 ) {
    perror( "close" );
}

exit( 0 );

Errors

If this function returns -1 to indicate failure, use ATDV_LASTERR( ) and
ATDV_ERRMSGP( ) to retrieve one of the following error reasons:
plays a specified backward R2MF signal  

r2_playbsig( )

EDX_BADPARM  •  Invalid parameter
EDX_BADPROD  •  Function not supported on this board
EDX_BADTPT  •  Invalid DV_TPT entry
EDX_BUSY  •  Busy executing I/O function
EDX_AMPLGEN  •  Invalid amplitude value in TN_GEN structure
EDX_FREQGEN  •  Invalid frequency component in TN_GEN structure
EDX_FLAGGEN  •  Invalid tn_dflag field in TN_GEN structure
EDX_SYSTEM  •  Windows NT system error - check errno

Cautions

The channel must be idle when calling this function.

See Also

- r2_creatfsig( )
- dx_blddt( )
- dx_playtone( )
- "R2MF Signaling" - (Voice Features Guide for Windows NT)
$r_2\text{playbsig}(\ )$ plays a specified backward R2MF signal
4. Voice Data Structures and Device Parameters

This chapter provides a description of the voice library data structures and voice board parameters. The following topics are included:

- Data Structures and Tables from the Voice Library (Section 4.1)
- Parameters used by `dx_setparm()` and `dx_getparm()` (Section 4.2)

4.1. Voice Library Data Structures

The following sections describe each of the data structures used by the Voice Library functions.

The voice software includes structures that indicate I/O termination, contain a device’s status information, or provide other function-specific information. The following structures are used:

- `DV_DIGIT` - User Digit Buffer Structure
- `DX_CST` - Call Status Transition Structure
- `DX_CAP` - Call Analysis Parameter Structure
- `DX_EBLK` - Event Block Structure
- `DX_IOTT` - I/O Transfer Table Structure
- `DX_SVCB` - Speed/Volume Adjustment Condition Block
- `DX_SVMT` - Speed/Volume Modification Block
- `DX_TPB` - Test Parameter Block Structure
- `DX_UIO` - User-definable I/O Structure
- `DX_XPB` - I/O Transfer Parameter Block
- `TN_GEN` - Tone Generation Template structure

**NOTE:** `DV_TPT` termination parameter structure, which is defined in the Standard Runtime Library is described in Appendix A.
4.1.1. DV_DIGIT - user digit buffer

When a `dx_getdig()` is performed, the digits are collected from the firmware and transferred to the user’s digit buffer. The digits are stored as an array inside the DV_DIGIT structure.

The typedef for the structure is shown below:

```c
typedef struct DV_DIGIT {
    char dg_value[DG_MAXDIGS +1]; /* ASCII values of digits */
    char dg_type[DG_MAXDIGS +1]; /* Type of digits */
} DV_DIGIT;
```

where:

**dg_value** is a NULL-terminated string of the ASCII values of the digits collected.

**dg_type** is an array (terminated by DG_END) of the digit types corresponding to each of the digits contained in dg_value. The defines for the digit types are:

- DG_DTMF: DTMF digit
- DG_LPD: Loop Pulse digit
- DG_MF: MF digit
- DG_USER1: User defined tone
- DG_USER2: User defined tone
- DG_USER3: User defined tone
- DG_USER4: User defined tone
- DG_USER5: User defined tone

DG_MAXDIGS is the define for the maximum number of digits (31) that can be returned by a single call to `dx_getdig()`. Refer to `dxxxlib.h` for the value of DG_MAXDIGS.

4.1.2. DX_CAP - change default call analysis parameters

The DX_CAP structure modifies parameters that control Frequency Detection, Cadence Detection, Loop Current, and Positive Voice Detection. DX_CAP structure is used for input to the `setcparm()` function to modify call analysis channel parameters when using `dx_dial()`.

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4. Voice Data Structures and Device Parameters

This structure provides the ability to change the default Call Analysis parameters when using `dx_dial()` . This structure may be used only under the following circumstances:

- When dialing on D/41ESC, D/xxxSC (D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1, D/320SC, DTI/241SC, DTI/301SC, LSI/81SC, or LSI/161SC) channels
- When clearing DX_CAP fields using `dx_clrcap()`

**NOTE:** For more detail about Call Analysis and how and when to use the DX_CAP structure see the Voice Features Guide for Windows NT.

To clear the fields in a DX_CAP structure, use the `dx_clrcap()` function.

If you set any DX_CAP field to 0, the field will be reset to the default value for the field. The setting used by a previously called `dx_dial()` function is ignored.

The typedef for the structure is shown on the following pages.

```c
typedef struct DX_CAP {
    unsigned short ca_nbrdna;    /* # of rings before no answer. */
    unsigned short ca_stdely;    /* Delay after dialing before analysis. */
    unsigned short ca_cnossig;   /* Duration of no signal timeout delay. */
    unsigned short ca_lcdly;     /* Delay after dial before lc drop connect */
    unsigned short ca_lcdly1;    /* Delay after lc drop conn. before msg. */
    unsigned short ca_hedge;     /* Edge of answer to send connect message. */
    unsigned short ca_cnosil;    /* Initial continuous noise timeout delay. */
    unsigned short ca_lo1tola;   /* % acceptable pos. dev of short low sig. */
    unsigned short ca_lo1tolb;   /* % acceptable neg. dev of short low sig. */
    unsigned short ca_lo2tola;   /* % acceptable pos. dev of long low sig. */
    unsigned short ca_lo2tolb;   /* % acceptable neg. dev of long low sig. */
    unsigned short ca_hi1tola;   /* % acceptable pos. dev of high signal. */
    unsigned short ca_hi1tolb;   /* % acceptable neg. dev of high signal. */
} DX_CAP;
```
unsigned short ca_lo1bmax;        /* Maximum interval for short low for busy */
unsigned short ca_lo2bmax;        /* Maximum interval for long low for busy */
unsigned short ca_hi1bmax;        /* Maximum interval for 1st high for busy */
unsigned short ca_nbBusy;         /* Num. of highs after nbdna busy check */
unsigned short ca_logltch;        /* Silence deglitch duration */
unsigned short ca_hiogltch;       /* Non-silence deglitch duration */
unsigned short ca_lo1rmax;        /* Max. short low dur. of double ring */
unsigned short ca_lo2rmin;        /* Min. long low dur. of double ring */
unsigned short ca_intflg;         /* Operator intercept mode */
unsigned short ca_intfltr;        /* Minimum signal to qualify freq detect */
unsigned short rfu1;              /* reserved for future use */
unsigned short rfu2;              /* reserved for future use */
unsigned short rfu3;              /* reserved for future use */
unsigned short rfu4;              /* reserved for future use */
unsigned short ca_hisiz;          /* Used to determine which lowmax to use */
unsigned short ca_alowmax;        /* Max. low before con. if high >hisme */
unsigned short ca_blowmax;        /* Max. low before con. if high <hisme */
unsigned short ca_nbrbeg;         /* Number of rings before analysis begins */
unsigned short ca_hi1ceil;        /* Maximum 2nd high dur. for a retrain */
unsigned short ca_lo1ceil;        /* Maximum 1st low dur. for a retrain */
unsigned short ca_lowerfrq;       /* Lower allowable frequency in hz */
unsigned short ca_upperfrq;       /* Upper allowable frequency in hz */
unsigned short ca_timefrq;        /* Total duration of good signal required */
unsigned short ca_rejctfrq;       /* Allowable % of bad signal */
unsigned short ca_maxansr;        /* Maximum duration of answer */
unsigned short ca_ansrdgl;        /* Silence deglitching value for answer */
unsigned short ca_maxtimefrq;     /* Max time for 1st freq to remain in bounds */
unsigned short ca_lower2frq;      /* Lower bound for second frequency */
unsigned short ca_upper2frq;      /* Upper bound for second frequency */
unsigned short ca_time2frq;       /* Min time for 2nd freq to remains in bounds */
unsigned short ca_maxtime2frq;    /* Max time for 2nd freq to remain in bounds */
unsigned short ca_lower3frq;      /* Lower bound for third frequency */
unsigned short ca_upper3frq;      /* Upper bound for third frequency */
unsigned short ca_time3frq;       /* Min time for 3rd freq to remains in bounds */
unsigned short ca_maxtime3frq;    /* Max time for 3rd freq to remain in bounds */
unsigned short ca_dtn_pres;       /* Length of a valid dial tone */
                             /* (def=1sec) */
unsigned short ca_dtn_npres;      /* Max time to wait for dial tone */
                             /* (def=3sec) */
unsigned short ca_dtn_deboff;     /* The dialtone off debouncer */
                             /* (def=100ms) */
4. Voice Data Structures and Device Parameters

```c
unsigned short ca_pamd_failtime; /* Wait for AMD/PVD after cadence break (default=4sec) */
unsigned short ca_pamd_minring; /* Min allowable ring duration (def=1.9sec) */
byte ca_pamd_spdval; /* Set to 2 selects quick decision (def=1) */
byte ca_pamd_qtemp; /* The Qualification template to use for PAMD */
unsigned short ca_noanswer; /* Time before no answer after first ring (default=30sec) */
unsigned short ca_maxintering; /* Max inter ring delay before connect (8 sec) */
}
```

**DX_CAP Parameter Descriptions**

**ca_nbrdna**  
Number of Rings Before Detecting No Answer: The number of single or double rings to wait before returning a no answer. (CA: Basic only)


**ca_stdely**  
Start Delay: The delay after dialing has been completed and before starting analysis for Cadence Detection, Frequency Detection, and Positive Voice Detection. (CA)


**ca_cnosig**  
Continuous No Signal: The maximum time of silence (no signal) allowed immediately after Cadence Detection begins. If exceeded, a no ringback is returned. (CA)

Length: 2. Default: 4000. Units: 10 ms.

**ca_lcdly**  
Loop Current Delay: The delay after dialing has been completed and before beginning Loop Current Detection. (CA)

-1: Disable Loop Current Detection.


**ca_lcdly1**  
Loop Current Delay 1: The delay after Loop Current Detection detects a transient drop in loop current and before Call Analysis returns a connect to the application. (CA)
**Voice Programmer's Guide for Windows NT**


**ca_hedge**

Hello Edge: The point at which a *connect* will be returned to the application. (CA)

1: Rising Edge (immediately when a connect is detected).
2: Falling Edge (after the end of the salutation).


**ca_cnosil**

Continuous Nonsilence: The maximum length of the first or second period of nonsilence allowed. If exceeded, a *no ringback* is returned. (CA)


**ca_lo1tola**

Low 1 Tolerance Above: Percent acceptable positive deviation of short low signal. (CA: Basic only)


**ca_lo1tolb**

Low 1 Tolerance Below: Percent acceptable negative deviation of short low signal. (CA: Basic only)


**ca_lo2tola**

Low 2 Tolerance Above: Percent acceptable positive deviation of long low signal. (CA: Basic only)


**ca_lo2tolb**

Low 2 Tolerance Below: Percent acceptable negative deviation of long low signal. (CA: Basic only)


**ca_hi1tola**

High 1 Tolerance Above: Percent acceptable positive deviation of high signal. (CA: Basic only)


**ca_hi1tolb**

High 1 Tolerance Below: Percent acceptable negative deviation of high signal. (CA: Basic only)

4. Voice Data Structures and Device Parameters

ca_lo1bmax  Low 1 Busy Maximum: Maximum interval for short low for busy. (CA: Basic only)
Length: 2. Default: 90. Units: 10 ms.

ca_lo2bmax  Low 2 Busy Maximum: Maximum interval for long low for busy. (CA: Basic only)
Length: 2. Default: 90. Units: 10 ms.

ca_hi1bmax  High 1 Busy Maximum: Maximum interval for first high for busy. (CA: Basic only)
Length: 2. Default: 90. Units: 10 ms.

ca_nsbusy  Nonsilence Busy: The number of nonsilence periods in addition to nbdn to wait before returning a busy. (CA: Basic only)
Length: 1. Default: 0. Negative values are valid.

ca_logltch  Low Glitch: The maximum silence period to ignore. Used to help eliminate spurious silence intervals. (CA)
Length: 2. Default: 15. Units: 10 ms.

ca_higltch  High Glitch: The maximum nonsilence period to ignore. Used to help eliminate spurious nonsilence intervals. (CA)

ca_lo1rmax  Low 1 Ring Maximum: Maximum short low duration of double ring. (CA: Basic only)
Length: 2. Default: 90. Units: 10 ms.

ca_lo2rmin  Low 2 Ring Minimum: Minimum long low duration of double ring. (CA: Basic only)

ca_intflg  Intercept Mode Flag: This parameter enables or disables SIT Frequency Detection, Positive Voice Detection (PVD), and/or Positive Answering Machine Detection (PAMD), and selects the mode of operation for Frequency
Detection. (CA)

**DX_OPTEN:**
Enable Frequency Detection and wait for detection of a connect using Cadence Detection or Loop Current Detection before returning an *intercept.*

**DX_OPTDIS:**
Disable Frequency Detection and PVD.

**DX_OPTNOCON:**
Enable Frequency Detection return an *intercept* immediately after detecting a valid frequency.

**DX_PVDENABLE:**
Enable PVD.

**DX_PVDOPTEN:**
Enable PVD and DX_OPTEN.

**DX_PVDOPTNOCON:**
Enable PVD and DX_OPTNOCON.

**DX_PAMDENABLE:**
Enable PAMD.

**DX_PAMDOPTEN:**
Enable PAMD and DX_OPTEN.

Length: 1. Default: DX_OPTEN.

- **ca_intfltr**
  Not used.

- **ca_hisz**
  High Size: Used to determine whether to use alowmax or blowmax. (CA: Basic only)
  Length: 2. Default: 90. Units: 10 ms.

- **ca_alowmax**
  A Low Maximum: Maximum low before connect if high > hisiz. (CA: Basic only)

- **ca_blowmax**
  B Low Maximum: Maximum low before connect if high < hisiz. (CA: Basic only)
  Length: 2. Default: 530. Units: 10 ms.

- **ca_nbrbeg**
  Number Before Beginning: Number of nonsilence periods before analysis begins. (CA: Basic only)
4. Voice Data Structures and Device Parameters


ca_hi1ceil High 1 Ceiling: Maximum 2nd high duration for a retrain.
(CA: Basic only)
Length: 2. Default: 78. Units: 10 ms.

ca_lo1ceil Low 1 Ceiling: Maximum 1st low duration for a retrain.
(CA: Basic only)

ca_lowerfrq Lower Frequency: Lower bound for 1st tone in an SIT.
(CA)

ca_upperfrq Upper Frequency: Upper bound for 1st tone in an SIT.
(CA)

ca_timefrq Time Frequency: Minimum time for 1st tone in an SIT to remain in bounds. The minimum amount of time required for the audio signal to remain within the frequency detection range specified by upperfrq and lowerfrq for it to be considered valid. (CA)
Length: 1. Default: 5. Units: 10 ms.

ca_rejctfrq Not used.

ca_maxansr Maximum Answer: The maximum allowable length of ansrssize. When ansrssize exceeds maxansr, a connect is returned to the application. (CA)
Length: 2. Default: 1000. Units: 10 ms.

ca_ansrdgl Answer Deglitcher: The maximum silence period allowed between words in a salutation. This parameter should be enabled only when you are interested in measuring the length of the salutation. (CA)
-1: Disable this condition.
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Length: 2. Default: -1. Units: 10 ms.

ca_pvdmxper Not used.
ca_pvdszwnd Not used.
ca_pvddly Not used.
ca_mxtimefrq Maximum Time Frequency: Maximum allowable time for 1st tone in an SIT to be present.

Default: 0. Units: 10 ms..

ca_lower2frq Lower Bound for 2nd Frequency: Lower bound for 2nd tone in an SIT.

Default: 0. Units: Hz.

ca_upper2frq Upper Bound for 2nd Frequency: Upper bound for 2nd tone in an SIT.

Default: 0. Units: Hz.

ca_time2frq Time for 2nd Frequency: Minimum time for 2nd tone in an SIT to remain in bounds.

Default: 0. Units: 10 ms.

ca_mxtime2frq Maximum Time for 2nd Frequency: Maximum allowable time for 2nd tone in an SIT to be present.

Default: 0. Units: 10 ms.

ca_lower3frq Lower Bound for 3rd Frequency: Lower bound for 3rd tone in an SIT.

Default: 0. Units: Hz.

ca_upper3frq Upper Bound for 3rd Frequency: Upper bound for 3rd tone in an SIT.

Default: 0. Units: Hz.

ca_time3frq Time for 3rd Frequency: Minimum time for 3rd tone in an SIT to remain in bounds.
4. Voice Data Structures and Device Parameters

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
<th>Default</th>
<th>Units</th>
</tr>
</thead>
<tbody>
<tr>
<td>ca_mxtime3frq</td>
<td>Maximum Time for 3rd Frequency: Maximum allowable time for 3rd tone in an SIT to be present.</td>
<td>0</td>
<td>10 ms</td>
</tr>
<tr>
<td>ca_dtn_pres</td>
<td>Dial Tone Present: Length of time that a dial tone must be continuously present. (CA: Enhanced only)</td>
<td>100</td>
<td>10 ms</td>
</tr>
<tr>
<td>ca_dtn_npres</td>
<td>Dial Tone Not Present: Maximum length of time to wait before declaring dial tone failure. (CA: Enhanced only)</td>
<td>300</td>
<td>10 ms</td>
</tr>
<tr>
<td>ca_dtn_deboff</td>
<td>Dial Tone Debounce: Maximum gap allowed in an otherwise continuous dial tone before it is considered invalid. (CA: Enhanced only)</td>
<td>10</td>
<td>10 ms</td>
</tr>
<tr>
<td>ca_pamd_faiTime</td>
<td>PAMD Fail Time: Maximum time to wait for Positive Answering Machine Detection or Positive Voice Detection after a cadence break. (CA: Enhanced only)</td>
<td>400</td>
<td>10 ms</td>
</tr>
<tr>
<td>ca_pamd_minring</td>
<td>Minimum PAMD Ring: Minimum allowable ring duration for Positive Answering Machine Detection. (CA: Enhanced only)</td>
<td>190</td>
<td>10 ms</td>
</tr>
<tr>
<td>ca_pamd_spdval</td>
<td>PAMD Speed Value: Quick or full evaluation for PAMD detection. PAMD_FULL = Full evaluation of response PAMD_QUICK = Quick look at connect circumstances (CA: Enhanced only)</td>
<td>PAMD_FULL</td>
<td></td>
</tr>
</tbody>
</table>
ca_pamd_qtemp PAMD Qualification Template: Which PAMD template to use. Options are PAMD_QUAL1TMP or PAMD_QUAL2TMP; at present, only PAMD_QUAL1TMP is available. (CA: Enhanced only)

Default: PAMD_QUAL1TMP.

c_a_noanswer No Answer: Length of time to wait after first ringback before deciding that the call is not answered. (CA: Enhanced only)

Default: 3000. Units: 10 ms.

c_a_maxintering Maximum Inter-ring Delay: Maximum time to wait between consecutive ringback signals before deciding that the call has been connected. (CA: Enhanced only)

Default: 800. Units: 10 ms.

4.1.3. DX_CST - call status transition structure

DX_CST contains call status transition information after an asynchronous TDX_CST termination or TDX_SETHOOK event occurs. Use the Event Management function, sr_getevtdatap( ) (see Appendix A) to retrieve the structure.

The typedef for DX_CST is shown below:

typedef struct DX_CST {
  unsigned short cst_event;
  unsigned short cst_data;
} DX_CST;

cst_event contains one of the following events:

- DE_DIGITS received a digit
- DE_LCOFF loop current off event
- DE_LCON loop current on event
- DE_LCREV loop current reversal event
- DE_RINGS rings received event
- DE_RNGOFF caller hang up (incoming call is dropped before being
4. Voice Data Structures and Device Parameters

DE_SILOFF • silence off event
DE_SILON • silence on event
DE_TONEOFF • tone off event
DE_TONEON • tone on event
DE_WINK • received a wink
DX_OFFHOOK • offhook event
DX_ONHOOK • onhook event

NOTE: DX_ONHOOK and DX_OFFHOOK are returned if a TDX_SETHOOK termination event is received.

cst_data contains data associated with cst_event. Valid values are:

<table>
<thead>
<tr>
<th>CST event type</th>
<th>CST event data</th>
</tr>
</thead>
<tbody>
<tr>
<td>DE_DIGITS</td>
<td>ASCII digit (low byte) and the digit type (high byte)</td>
</tr>
<tr>
<td>DE_LCOFF</td>
<td>time previous last loop current &quot;on&quot; transition in 10 ms units</td>
</tr>
<tr>
<td>DE_LCON</td>
<td>time since previous loop current &quot;off&quot; transition in 10 ms units</td>
</tr>
<tr>
<td>DE_LCREV</td>
<td>time since previous loop current reversal transition in 10 ms units</td>
</tr>
<tr>
<td>DE_RINGS</td>
<td>0</td>
</tr>
<tr>
<td>DE_SILOFF</td>
<td>time since previous silence started in 10 ms units</td>
</tr>
<tr>
<td>DE_SILON</td>
<td>time since previous silence stopped in 10 ms units</td>
</tr>
<tr>
<td>DE_TONEOFF</td>
<td>user-specified tone ID</td>
</tr>
<tr>
<td>DE_TONEON</td>
<td>user-specified tone ID</td>
</tr>
<tr>
<td>DE_WINK</td>
<td>N/A</td>
</tr>
<tr>
<td>DX_OFFHOOK</td>
<td>N/A</td>
</tr>
<tr>
<td>DX_ONHOOK</td>
<td>N/A</td>
</tr>
</tbody>
</table>
4.1.4. DX_EBLK- call status event block structure

This structure is returned by `dx_getevt()` and indicates which Call Status Transition event occurred.

**NOTE:** `dx_getevt()` is a synchronous function which blocks until an event occurs. For information about asynchronously waiting for CST events, see `dx_setevmsk()`.

The typedef for the structure is shown below:

```c
typedef struct DX_EBLK {
    unsigned short ev_event;      /* Event that occurred */
    unsigned short ev_data;       /* Event specific data */
    unsigned char ev_rfu[12];     /* Reserved for future use*/
}DX_EBLK;
```

where:

- `ev_event` specifies the event that occurred on the device. The following defines can be entered in `ev_event`:
  - `DE_DIGITS` • digit received
  - `DE_LCOFF` • loop current off
  - `DE_LCON` • loop current on
  - `DE_LCREV` • loop current reversal
  - `DE_RINGS` • rings received
  - `DE_SILOFF` • non-silence detected
  - `DE_SILON` • silence detected
  - `DE_TONEOFF` • tone off event occurred
  - `DE_TONEON` • tone on event occurred
  - `DE_WINK` • wink has occurred

`ev_data` contains data specific to the event contained in `ev_event`. Table 6 shows what data is returned for each event that appears in `ev_event`. All the lengths of time are in 10 ms units.
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Table 6. Values Returned in ev_data

<table>
<thead>
<tr>
<th>Event</th>
<th>Data Returned in ev_data</th>
</tr>
</thead>
<tbody>
<tr>
<td>DE_DIGITS</td>
<td>Digit and digit type</td>
</tr>
<tr>
<td>DE_LCOFF</td>
<td>The length of time that loop current was on before the loop-current-off event was detected</td>
</tr>
<tr>
<td>DE_LCON</td>
<td>The length of time that loop current was off before the loop-current-on event was detected</td>
</tr>
<tr>
<td>DE_LCREV</td>
<td>The length of time that loop current was reversed before the loop-current-reversal event was detected</td>
</tr>
<tr>
<td>DE_RINGS</td>
<td>Returns no data</td>
</tr>
<tr>
<td>DE_SILOFF</td>
<td>The length of time that silence occurred before non-silence (noise or meaningful sound) was detected</td>
</tr>
<tr>
<td>DE_SILOON</td>
<td>The length of time that non-silence occurred before silence was detected</td>
</tr>
<tr>
<td>DE_TONEOFF</td>
<td>The tone ID for the tone-off event</td>
</tr>
<tr>
<td>DE_TONEON</td>
<td>The tone ID for the tone-on event</td>
</tr>
<tr>
<td>DE_WINK</td>
<td>Returns no data</td>
</tr>
</tbody>
</table>

ev_rfu is reserved for future use.

4.1.5. DX_IOTT - I/O transfer table

The DX_IOTT structure identifies a source or destination for voice data. It is used with the `dx_play()` and `dx_rec()` functions.

The typedef for the structure is shown below:
typedef struct dx_iott {
    unsigned short io_type;          /* Transfer type */
    unsigned short rfu;              /* Reserved */
    int io_fhandle;                  /* File descriptor */
    char * io_bufp;                   /* Pointer to base memory */
    unsigned long io_offset;         /* File/Buffer offset */
    long int io_length;              /* Length of data */
    DX_IOTT *io_nextp;               /* Ptr to next DX_IOTT if IO_LINK set */
    DX_IOTT *io_prevp;               /* (Optional) Ptr to previous DX_IOTT */
} DX_IOTT;

where:

io_type specifies whether the data is stored in a file or in memory. It also determines if the next DX_IOTT structure is contiguous in memory, linked, or if this is the last DX_IOTT in the chain. Set the io_type field to an OR combination of the required defines listed below.

Specify data transfer type as follows:

IO_DEV • data is being played from a file
IO_MEM • data is being played from memory

Specify structure linkage as follows:

IO_CONT • the next DX_IOTT structure is contiguous (default)
IO_LINK • the next DX_IOTT structure is part of a linked list
IO_EOT • this is the last DX_IOTT structure in the chain

If none of IO_CONT, IO_LINK, or IO_EOT are specified, IO_CONT is assumed.

io_fhandle contains a unique file descriptor if IO_DEV is set in io_type. If IO_DEV is not set in io_type, io_fhandle should be set to 0.

io_bufp field indicates a base memory address if IO_MEM is set in io_type.

io_offset indicates

• an offset from the beginning of a file if IO_DEV is specified in io_type
• an offset from the base buffer address specified in io_bufp if IO_MEM is specified in io_type.
4. Voice Data Structures and Device Parameters

io_length indicates the number of bytes allocated for recording or the byte length of the playback file. Specify -1 to play until end of data. During dx_play(), a value of -1 causes playback to continue until an EOF is received or one of the terminating conditions is satisfied. During dx_rec(), a value of -1 in io_length causes recording to continue until one of the terminating conditions is satisfied.

io_nextp points to the next DX_IOTT structure in the linked list if IO_LINK is set in io_type.

io_prevp points to the previous DX_IOTT structure. This field is automatically filled in when dx_rec() or dx_play() are called. The io_prevp field of the first DX_IOTT structure is set to NULL.

A DX_IOTT structure describes a single data transfer to or from one file, memory block, or custom device. If the voice data is stored on a custom device, the device must have a standard Windows NT device interface. The device must support open(), close(), read(), and write() and lseek().

To use multiple combinations, each source or destination of I/O is specified as one element in an array of DX_IOTT structures. The last DX_IOTT entry must have IO_EOT specified in the io_type field.

For example, to use different sources for playback, an array or linked list of DX_IOTT structures can be specified as follows:

### Playback Array Example

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

DX_IOTT iott[3];
/* first iott: voice data in a file with descriptor fd1 */
iott[0].io_fhandle = fd1;
iott[0].io_offset = 0;
iott[0].io_length = -1;
iott[0].io_type = IO_DEV;

/* second iott: voice data in a file with descriptor fd2 */
iott[1].io_fhandle = fd2;
iott[1].io_offset = 0;
iott[1].io_length = -1;
iott[1].io_type = IO_DEV;

/* third iott: voice data in a file with descriptor fd3 */
iott[2].io_fhandle = fd3;
iott[2].io_offset = 0;
iott[2].io_length = -1;
iott[2].io_type = IO_DEV | IO_EOT;
```
/* play all three voice files: pass &iott[0] as argument to dx_play() */
/* form a linked list of iott[0] and iott[2] */
iott[0].io_nextp=iott[2];
iott[0].io_type|=IO_LINK
/* pass &iott[0] as argument to dx_play(). This time only files 1 and 3
* will be played. */

4.1.6. DX_SVMT - speed/volume modification table structure

The DX_SVMT structure has 21 entries that represent different levels of speed or volume. This structure is used to set or retrieve the Speed/Volume Modification Table values (using dx_setsvmt() and dx_getsvmt() respectively).

The DX_SVMT typedef is shown below:

```c
typedef struct DX_SVMT {
    char decrease[10];        /* Ten Downward Steps */
    char origin;              /* Regular Speed or Volume */
    char increase[10];        /* Ten Upward Steps */
} DX_SVMT;
```

Table 7 describes the valid speed and volume settings for each of the DX_SVMT entries.

**NOTE:** Although there are 21 entries available in the DX_SVMT structure, they do not all have to be utilized for changing speed or volume - the range can be as small as you require. Ensure that you insert -128 (80h) in any entries that do not contain a speed or volume adjustment.

**Table 7. DX_SVMT Entries**

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>decrease[10]</td>
<td>Array that provides a maximum of 10 downward steps in speed or volume. The size of the steps is specified in this table. Specify the value -128 (80h) in any entries you are not using.</td>
</tr>
</tbody>
</table>

**Valid Values:**

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4. Voice Data Structures and Device Parameters

**Speed** - Percentage decrease from the origin (which is set to 0). Valid Values must be between -1 and -50.

**Volume** - Step in dB from the origin (which is set to 0). Valid Values must be between -1 and -30.

**origin**
Represents the standard play speed or volume. This is the original setting or starting point for Speed and Volume Control.

Valid Values:
Set to 0 for speed or volume.

**increase[10]**
Array that provides a maximum of 10 upward steps in speed or volume. The size of the steps is specified in this table. Specify the value -128 (80h) in any entries you are not using.

Valid Values:

**Speed** - Percentage increase from the origin (which is set to 0). Valid Values must be between 1 and 50. Specify the value -128 (80h) in any entries you are not using.

**Volume** - Step in dB from the origin (which is set to 0). Valid Values must be between 1 and 10. Specify the value -128 (80h) in any entries you are not using.

4.1.7. **DX_SVCB - speed/volume adjustment condition block**

This structure is used by `dx_setsvcond()` to set the following:

- which Speed/Volume Modification Table to use (speed or volume)
- adjustment type (increase/decrease, absolute value, toggle)
- adjustment conditions (incoming digit, beginning of play)
- level/edge sensitivity for incoming digits
The typedef for the DX_SVCB structure is described below:

```c
typedef struct DX_SVCB {
    unsigned short type;          /* Bit Mask */
    short adjsize;                /* Adjustment Size */
    unsigned char digit;          /* ASCII digit value that causes the action */
    unsigned char digtype;        /* Digit Type (e.g., 0 = DTMF) */
} DX_SVCB;
```

**NOTES:**

1. To clear the DX_SVCB adjustment condition blocks, call the `dx_clrsvcond()` function.

2. DX_SVCB adjustment condition blocks can only be added to the existing conditions. To reset or remove any DX_SVCB adjustment condition blocks, all conditions must be cleared (using `dx_clrsvcond()`).

Table 8 describes each of the valid values for the fields.

**Table 8. DX_SVCB Entries**

<table>
<thead>
<tr>
<th>Defines (type field)</th>
<th>Description (for type field)</th>
<th>Description (for adjsize field)</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Speed or Volume</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Choose one:</td>
<td></td>
<td></td>
</tr>
<tr>
<td>SV_SPEEDTBL</td>
<td>Modify speed table.</td>
<td>N/A.</td>
</tr>
<tr>
<td>SV_VOLUMETBL</td>
<td>Modify volume table.</td>
<td>N/A.</td>
</tr>
<tr>
<td><strong>Adjustment Type</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Choose one:</td>
<td></td>
<td></td>
</tr>
<tr>
<td>SV_ABSPOS</td>
<td>Sets adjsize field to indicate an absolute volume adjustment position position in the Speed or Volume Modification Tables.</td>
<td>Specify the required speed or between -10 and +10 in the Volume Modification Speed or Tables.</td>
</tr>
</tbody>
</table>
4. Voice Data Structures and Device Parameters

<table>
<thead>
<tr>
<th>Defines (type field)</th>
<th>Description (for type field)</th>
<th>Description (for adjsize field)</th>
</tr>
</thead>
<tbody>
<tr>
<td>SV_RELCURPOS</td>
<td>Sets adjsize field to indicate a number of steps by which to adjust speed or volume.</td>
<td>Specify how many positive or negative &quot;steps&quot; in the Speed or Volume Modification Tables by which to adjust the speed or volume. For example, specify -2 to lower the speed or volume by 2 steps in the Speed or Volume Modification Tables.</td>
</tr>
</tbody>
</table>
| SV_TOGGLE            | Sets adjsize field to use one of the toggle defines | Set the "toggle values by specifying one of the following:  
  
  SV_TOGORIGIN - sets the current speed or volume to toggle between the origin and the last modified level of speed or volume. 
  
  SV_CURORIGIN - resets the current speed or volume level to the origin (i.e., regular speed or volume). 
  
  SV_CURLASTMOD - sets the current speed or volume to the last modified speed volume level. |
<table>
<thead>
<tr>
<th>Defines (type field)</th>
<th>Description (for type field)</th>
<th>Description (for adjsize field)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>SV_RESETORIG - resets the current speed or volume to the origin and the last modified speed or volume in the origin.</td>
<td></td>
</tr>
</tbody>
</table>

**Optional - Choose one:**

**SV_LEVEL**
- Sets the digit adjustment condition to be level sensitive. At the start of play, adjustments will be made according to adjustment condition digits contained in the digit buffer.
- If SV_LEVEL is not specified, the digit adjustment condition is edge sensitive, and will wait for a new occurrence of the digit before play adjusting.

**SV_BEGINPLAY**
- Adjusts speed or volume at the beginning of each play. In this case, digit and digtype fields are ignored.

**Choose one:**

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4. Voice Data Structures and Device Parameters

<table>
<thead>
<tr>
<th>Defines (type field)</th>
<th>Description (for type field)</th>
<th>Description (for adjsize field)</th>
</tr>
</thead>
<tbody>
<tr>
<td>0-9, a-d, #, * ASCII digit that adjusts play.</td>
<td>N/A.</td>
<td>Digit Type</td>
</tr>
</tbody>
</table>

Specify the following:

- **DG_DTMF**: Specifies DTMF digits. N/A.

Volume Example

The following `DX_SVCB` structure is set to decrease the volume by one step whenever the DTMF digit "1" is detected:

```c
svcb[0].type    = SV_VOLUMETBL | SV_RELCURPOS;
svcb[0].adjsize = -1;
svcb[0].digit    = '1';
svcb[0].digtype    = DG_DTMF;
```

The following `DX_SVCB` structure will set speed to the value in Speed Modification Table position 5 whenever the DTMF digit "2" is detected:

```c
svcb[0].type    = SV_SPEEDTBL | SV_ABSPOS;
svcb[0].adjsize = 5;
svcb[0].digit    = '2';
svcb[0].digtype    = DG_DTMF;
```

4.1.8. DX_UIO - user-definable I/O structure

This structure, returned by `dx_setuio()`, contains pointers to user-defined I/O functions for accessing nonstandard storage devices.

```c
/*
 * Structure for user-defined I/O functions
 */
typedef struct DX_UIO {
  int (*u_read) ( );
  int (*u_write) ( );
  int (*u_seek) ( );
} DX_UIO;
```
The **u_read** field is a pointer to the user-defined **read( )** function, which returns an integer equal to the number of bytes read or -1 for error.

The **u_write** field is a pointer to the user-defined **write( )** function, which returns an integer equal to the number of bytes written or -1 for error.

The **u_seek** field is a pointer to the user-defined **lseek( )** function, which returns a long equal to the offset into the I/O device where the read or write is to start or -1 for error.

### 4.1.9. TN_GEN - tone generation template structure

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 shown below:

```c
typedef struct {
    unsigned short tg_dflag;  /* Dual Tone - 1, Single Tone - 0 */
    unsigned short tg_freq1;  /* Frequency for Tone 1 (HZ) */
    unsigned short tg_freq2;  /* Frequency for Tone 2 (HZ) */
    short          tg_ampl1;  /* Amplitude for Tone 1 (dB) */
    short          tg_ampl2;  /* Amplitude for Tone 2 (dB) */
    short          tg_dur;    /* Duration of the Generated Tone */
                        /* Units = 10ms */
} TN_GEN;
```

Table 9 lists the valid values for each field.

**Table 9. TN_GEN Values**

<table>
<thead>
<tr>
<th>TN_GEN Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>tg_dflag</td>
<td>Specifies single or dual tone. If single, the values in tg_freq2 and tg_ampl2 will be ignored.</td>
</tr>
</tbody>
</table>

Choose one:
- TN_SINGLE     single tone
- TN_DUAL       dual tone
4. Voice Data Structures and Device Parameters

<table>
<thead>
<tr>
<th>TN_GEN Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>tg_freq1</td>
<td>Frequency in Hz for tone 1 (range 200 to 2000 Hz)</td>
</tr>
<tr>
<td>tg_freq2</td>
<td>Frequency in Hz for tone 2; (range 200 to 2000 Hz)</td>
</tr>
<tr>
<td>tg_ampl1</td>
<td>Amplitude in dB for tone 1; (range 0 to -40 dB)</td>
</tr>
<tr>
<td>tg_ampl2</td>
<td>Amplitude in dB for tone 2; (range 0 to -40 dB)</td>
</tr>
<tr>
<td>tg_dur</td>
<td>Duration of the tone in 10 ms units (-1 = infinite duration)</td>
</tr>
</tbody>
</table>

4.1.10. DX_XPB - I/O transfer parameter block

The I/O Transfer Parameter Block (DX_XPB) data structure is used by the extended play and record functions to specify the file format (either VOX file or WAVE file), the data format (ADPCM, Mu-law PCM, A-law PCM, Linear PCM), the sampling rate (6, 8, or 11 KHz), and the resolution (4 or 8 bits per sample).

typedef struct {
    USHORT      wFileFormat; // file format
    USHORT      wDataFormat; // audio data format
    ULONG       nSamplesPerSec; // sampling rate
    ULONG       nBitsPerSample; // bits per sample
} DX_XPB;

The dx_playwav() convenience function does not specify a DX_XPB structure because the WAVE file contains the necessary format information.

<table>
<thead>
<tr>
<th>DX_XPB Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>wFileFormat</td>
<td>Specifies one of the following audio file formats. Note that this field is ignored by the convenience functions dx_recwav() , dx_playwav() , dx_recvox() , and dx_playvox() . Choose one: FILE_FORMAT_VOX Dialogic VOX file format FILE_FORMAT_WAVE Microsoft WAVE file format</td>
</tr>
</tbody>
</table>
### DX_XPB Field Description

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>wDataFormat</strong></td>
<td>Specifies one of the following data formats</td>
</tr>
<tr>
<td></td>
<td>Choose one:</td>
</tr>
<tr>
<td></td>
<td>DATA_FORMAT_DIALOGIC C_ADPCM</td>
</tr>
<tr>
<td></td>
<td>4-bit OKI ADPCM (Dialogic registered)</td>
</tr>
<tr>
<td></td>
<td>DATA_FORMAT_MULAW</td>
</tr>
<tr>
<td></td>
<td>8-bit mu-law PCM</td>
</tr>
<tr>
<td></td>
<td>DATA_FORMAT_ALAW</td>
</tr>
<tr>
<td></td>
<td>8-bit a-law PCM</td>
</tr>
<tr>
<td></td>
<td>DATA_FORMAT_PCM</td>
</tr>
<tr>
<td></td>
<td>8-bit Linear PCM</td>
</tr>
<tr>
<td><strong>nSamplesPerSec</strong></td>
<td>Specifies one of the following digitization rates</td>
</tr>
<tr>
<td></td>
<td>Choose one:</td>
</tr>
<tr>
<td></td>
<td>DRT_6KHZ</td>
</tr>
<tr>
<td></td>
<td>6 KHz sampling rate.</td>
</tr>
<tr>
<td></td>
<td>DRT_8KHZ</td>
</tr>
<tr>
<td></td>
<td>8 KHz sampling rate.</td>
</tr>
<tr>
<td></td>
<td>DRT_11KHZ</td>
</tr>
<tr>
<td></td>
<td>11 KHz sampling rate.</td>
</tr>
<tr>
<td></td>
<td>Note: 11KHz OKI ADPCM is not supported.</td>
</tr>
<tr>
<td><strong>nBitsPerSample</strong></td>
<td>Specifies number of bits per sample. This field must be set to 4 for ADPCM and 8 for all others.</td>
</tr>
</tbody>
</table>

### 4.2. Voice Board Parameter Defines for dx_getparm( )

The and dx_setparm( ) file `<install drive:><install directory>\dialogic \include\dxxxlib.h` contains defined masks for the D/4x parameters that can be examined and set using `dx_getparm()` and `dx_setparm()`. These parameters apply to any real or emulated D/4x board.
Channels and boards have different parameters. The board parameters, their default settings, read/write privileges, and descriptions are listed in Table 10.

The channel parameters are listed in Table 11. All time units are in 10 ms unless otherwise noted.

Table 10. Voice Board Parameters

<table>
<thead>
<tr>
<th>Define</th>
<th>Bytes</th>
<th>Read/Write</th>
<th>Default</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DXBD_CHNUM</td>
<td>1</td>
<td>R</td>
<td>-</td>
<td>Channel Number - number of channels on the board</td>
</tr>
<tr>
<td>DXBD_FLASHCHR</td>
<td>1</td>
<td>R/W &amp;</td>
<td>&amp;</td>
<td>Flash character - character that causes a hook flash when detected.</td>
</tr>
<tr>
<td>DXBD_FLASHTM</td>
<td>2</td>
<td>R/W 50</td>
<td>50</td>
<td>Flash Time - length of time onhook during flash</td>
</tr>
<tr>
<td>DXBD_HWTYP</td>
<td>1</td>
<td>R</td>
<td>-</td>
<td>Hardware Type - value can be:</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>TYP_D40</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>TYP_D41</td>
</tr>
<tr>
<td>DXBD_MAXPDOFF</td>
<td>2</td>
<td>R/W 50</td>
<td>50</td>
<td>Maximum Pulse Digit Off - max. time loop current may be off before the</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>existing loop pulse digit is considered invalid and reception is</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>reinitialized</td>
</tr>
</tbody>
</table>

359-CD
<table>
<thead>
<tr>
<th>Define</th>
<th>Bytes</th>
<th>Read/Write</th>
<th>Default</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DXBD_MAXSLOFF</td>
<td>2</td>
<td>R/W</td>
<td>25</td>
<td>Maximum Silence Off - maximum time for silence being off, during audio pulse detection</td>
</tr>
<tr>
<td>DXBD_MINIPD</td>
<td>2</td>
<td>R/W</td>
<td>25</td>
<td>Minimum Loop Interpulse Detection - minimum time between loop pulse digits during loop pulse detection</td>
</tr>
<tr>
<td>DXBD_MINISL</td>
<td>2</td>
<td>R/W</td>
<td>25</td>
<td>Minimum Interdigit Silence - minimum time for silence on between pulse digits for audio pulse detection</td>
</tr>
<tr>
<td>DXBD_MINLCOFF</td>
<td>2</td>
<td>R/W</td>
<td>0</td>
<td>Minimum Loop Current Off - minimum time before loop current drop message is sent</td>
</tr>
<tr>
<td>DXBD_MINPDOFF</td>
<td>1</td>
<td>R/W</td>
<td>2</td>
<td>Minimum Pulse Detection Off - minimum break interval for valid loop pulse detection</td>
</tr>
<tr>
<td>DXBD_MINPDON</td>
<td>1</td>
<td>R/W</td>
<td>2</td>
<td>Minimum Pulse Detection On - minimum make interval for valid loop pulse detection</td>
</tr>
</tbody>
</table>
4. **Voice Data Structures and Device Parameters**

<table>
<thead>
<tr>
<th>Define</th>
<th>Bytes</th>
<th>Read/Write</th>
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<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DXBD_MINSLOFF</td>
<td>1</td>
<td>R/W</td>
<td>2</td>
<td>Minimum Silence Off - min. time for silence to be off for valid audio pulse detection</td>
</tr>
<tr>
<td>DXBD_MINSLON</td>
<td>1</td>
<td>R/W</td>
<td>1</td>
<td>Minimum Silence On - min. time for silence to be on for valid audio pulse detection</td>
</tr>
<tr>
<td>DXBD_MINTIOF</td>
<td>1</td>
<td>R/W</td>
<td>5</td>
<td>Minimum DTI Off - minimum time required between rings-received events</td>
</tr>
<tr>
<td>DXBD_MINTION</td>
<td>1</td>
<td>R/W</td>
<td>5</td>
<td>Minimum DTI On - minimum time required for rings received event</td>
</tr>
<tr>
<td>DXBD_OFFHDLY</td>
<td>2</td>
<td>R/W</td>
<td>50</td>
<td>Offhook Delay - period after offhook, during which no events are generated e.g., no DTMF digits will be detected during this period.</td>
</tr>
<tr>
<td>DXBD_PAUSETM</td>
<td>2</td>
<td>R/W</td>
<td>200</td>
<td>Pause Time - delay caused by a comma in the dialing string</td>
</tr>
<tr>
<td>Define</td>
<td>Bytes</td>
<td>Read/Write</td>
<td>Default</td>
<td>Description</td>
</tr>
<tr>
<td>---------------</td>
<td>-------</td>
<td>------------</td>
<td>---------</td>
<td>---------------------------------------------------------</td>
</tr>
<tr>
<td>DXBD_P_BK</td>
<td>2</td>
<td>R/W</td>
<td>6</td>
<td>Pulse Dial Break - duration of pulse dial off-hook interval</td>
</tr>
<tr>
<td>DXBD_P_IDD</td>
<td>2</td>
<td>R/W</td>
<td>100</td>
<td>Pulse Interdigit Delay - time between digits in pulse dialing</td>
</tr>
<tr>
<td>DXBD_P_MK</td>
<td>2</td>
<td>R/W</td>
<td>4</td>
<td>Pulse Dial Make - duration of pulse dial offhook interval</td>
</tr>
<tr>
<td>DXBD_R_EDGE</td>
<td>1</td>
<td>R/W</td>
<td>ET_ROFF</td>
<td>Ring Edge - detection of ring edge, values can be: ET_RON (beginning of ring) ET_ROFF (end of ring)</td>
</tr>
<tr>
<td>DXBD_R_IRD</td>
<td>2</td>
<td>R/W</td>
<td>80</td>
<td>Inter-ring Delay - maximum time to wait for the next ring (100 ms units). Used to distinguish between calls. Set to 1 for T-1 applications.</td>
</tr>
<tr>
<td>DXBD_R_OFF</td>
<td>2</td>
<td>R/W</td>
<td>5</td>
<td>Ring-off Interval - minimum time for ring not to be present before qualifying as “not ringing” (100 ms units)</td>
</tr>
<tr>
<td>DXBD_R_ON</td>
<td>2</td>
<td>R/W</td>
<td>3</td>
<td>Ring-on Interval - minimum time ring must be present to</td>
</tr>
</tbody>
</table>
### 4. Voice Data Structures and Device Parameters

<table>
<thead>
<tr>
<th>Define</th>
<th>Bytes</th>
<th>Read/Write</th>
<th>Default</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DXBD_SYSCFG</td>
<td>1</td>
<td>R</td>
<td>-</td>
<td>qualify as a ring (100 ms units) System Configuration - JP8 status for D/4x boards: 0 = loop start interface (JP8 in); 1 = DTI/xxx interface (JP8 out): Silence and Non-silence Debounce - length of a changed state before Call Status Transition message is generated</td>
</tr>
<tr>
<td>DXBD_S_BNC</td>
<td>2</td>
<td>R/W</td>
<td>4</td>
<td></td>
</tr>
<tr>
<td>DXBD_TTDATA</td>
<td>1</td>
<td>R/W</td>
<td>10</td>
<td>Duration of DTMF digits for dialing.</td>
</tr>
<tr>
<td>DXBD_MFMINON</td>
<td>2</td>
<td>R/W</td>
<td>0</td>
<td>Minimum MF On - The duration to be added to the standard MF tone duration before the tone is detected. The minimum detection duration is 65 ms for KP tones and 40 ms for all other tones. This parameter affects all the channels on the board. (10 ms units)</td>
</tr>
<tr>
<td>DXBD_MFTONE</td>
<td>2</td>
<td>R/W</td>
<td>6</td>
<td>MF Minimum Tone Duration - The duration of a dialed</td>
</tr>
<tr>
<td>Define</td>
<td>Bytes</td>
<td>Read/Write</td>
<td>Default</td>
<td>Description</td>
</tr>
<tr>
<td>---------------------</td>
<td>-------</td>
<td>------------</td>
<td>---------</td>
<td>-------------------------------------------------------------------------------------------------------</td>
</tr>
<tr>
<td>Define Bytes</td>
<td>Read/Write</td>
<td>Default</td>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>MF tone. This parameter affects all the channels on the board.</td>
<td>Maximum value: 10 (10 ms units)</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>DxBD_MFDELAY</td>
<td>2</td>
<td>R/W</td>
<td>6</td>
<td>MF Interdigit Delay - The length of the silence period between tones during MF dialing. This parameter affects all the channels on the board. (10 ms units)</td>
</tr>
<tr>
<td>DxBD_MFLKPTONE</td>
<td>2</td>
<td>R/W</td>
<td>10</td>
<td>MF Length of LKP Tone - The length of the LKP tone during MF dialing. This parameter affects all the channels on the specified board. (10 ms units)</td>
</tr>
<tr>
<td>DxBD_T_IDD</td>
<td>2</td>
<td>R/W</td>
<td>5</td>
<td>DTMF Interdigit Delay - time between digits in DTMF dialing</td>
</tr>
<tr>
<td>DxBD_MINOFFHKTM</td>
<td>2</td>
<td>R/W</td>
<td>250</td>
<td>Minimum offhook time (10 ms)</td>
</tr>
</tbody>
</table>
4. Voice Data Structures and Device Parameters

<table>
<thead>
<tr>
<th>Define</th>
<th>Bytes</th>
<th>Read/Write</th>
<th>Default</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DXCH_DFLAGS</td>
<td>2</td>
<td>R/W</td>
<td>0</td>
<td>DTMF detection edge select</td>
</tr>
<tr>
<td>DXCH_DTINITSET</td>
<td>2</td>
<td>R/W</td>
<td>0</td>
<td>Specifies which DTMF digits to initiate play on. Values of different DTMF digits may be ORed together to form the bit mask. Possible values are listed below:</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Value</th>
<th>DTMF Digit</th>
</tr>
</thead>
<tbody>
<tr>
<td>-DM_1</td>
<td>1</td>
</tr>
<tr>
<td>-DM_2</td>
<td>2</td>
</tr>
<tr>
<td>-DM_3</td>
<td>3</td>
</tr>
<tr>
<td>-DM_4</td>
<td>4</td>
</tr>
<tr>
<td>-DM_5</td>
<td>5</td>
</tr>
<tr>
<td>-DM_6</td>
<td>6</td>
</tr>
<tr>
<td>-DM_7</td>
<td>7</td>
</tr>
<tr>
<td>-DM_8</td>
<td>8</td>
</tr>
<tr>
<td>-DM_9</td>
<td>9</td>
</tr>
<tr>
<td>-DM_0</td>
<td>0</td>
</tr>
<tr>
<td>-DM_S</td>
<td>*</td>
</tr>
<tr>
<td>-DM_P</td>
<td>#</td>
</tr>
<tr>
<td>-DM_A</td>
<td>a</td>
</tr>
<tr>
<td>-DM_B</td>
<td>b</td>
</tr>
<tr>
<td>-DM_C</td>
<td>c</td>
</tr>
<tr>
<td>-DM_D</td>
<td>d</td>
</tr>
<tr>
<td>DXCH_DTMFTLK</td>
<td>2</td>
</tr>
</tbody>
</table>
**Voice Programmer’s Guide for Windows NT**

<table>
<thead>
<tr>
<th>Define</th>
<th>Bytes</th>
<th>Read/Write</th>
<th>Default</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Increasing the value provides more immunity to talk-off/playoff.</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Set to -1 to disable.</td>
</tr>
</tbody>
</table>

- **DXCH_DTMFDEB** 2 R/W 0  
  DTMF debounce time - maximum length of time in which DTMF can be absent and then come back on again and still be considered the same DTMF tone.

- **DXCH_MFMODE** 2 R/W 2  
  This is a word-length bit mask that selects the minimum length of KP tones to be detected. The possible values of this field are:

  - 0 - detect KP tone > 40 ms
  - 2 - detect KP tone > 65 ms

  If the value is set to 2, any KP tone greater than 65ms will be returned to the application during MF detection. This ensures that only standard-length KP tones (100ms) are detected. If set to 0 (zero), any KP tone longer than 40ms will be detected.
### 4. Voice Data Structures and Device Parameters

<table>
<thead>
<tr>
<th>Define</th>
<th>Bytes</th>
<th>Read/Write</th>
<th>Default</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DXCH_MAXRWINK</td>
<td>1</td>
<td>R/W</td>
<td>20</td>
<td>Maximum Loop Current for Wink - The maximum time that loop current needs to be on before recognizing a wink (10 ms units)</td>
</tr>
<tr>
<td>DXCH_MINRWINK</td>
<td>1</td>
<td>R/W</td>
<td>10</td>
<td>Minimum Loop Current for Wink - The minimum time that loop current needs to be on before recognizing a wink (10 ms units)</td>
</tr>
<tr>
<td>DXCH_WINKDLY</td>
<td>1</td>
<td>R/W</td>
<td>15</td>
<td>Wink Delay - The delay after a ring is received before issuing a wink (10 ms units)</td>
</tr>
<tr>
<td>DXCH_RINGCNT</td>
<td>2</td>
<td>R/W</td>
<td>4</td>
<td>Number of rings to wait before returning a ring event.</td>
</tr>
<tr>
<td>DXCH_WINKLEN</td>
<td>1</td>
<td>R/W</td>
<td>15</td>
<td>Wink Length - The duration of a wink in the off-hook state (10 ms units)</td>
</tr>
<tr>
<td>DXCH_PLAYDRATE</td>
<td>2</td>
<td>R/W</td>
<td>6000</td>
<td>Play Digitization Rate - This parameter sets the digitization rate of the voice data that is</td>
</tr>
</tbody>
</table>
The Voice Programmer’s Guide for Windows NT

<table>
<thead>
<tr>
<th>Define</th>
<th>Bytes</th>
<th>Read/Write</th>
<th>Default</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DXCH_RECRDRATE</td>
<td>2</td>
<td>R/W</td>
<td>6000</td>
<td>Record Digitization Rate - This parameter sets the rate at which the recorded voice data is digitized. Voice Data can be digitized at 6k or 8k sampling rates. Valid values are: 6000 - 6K sampling rate 8000 - 8k sampling rate</td>
</tr>
</tbody>
</table>

Voice data must be played at the same rate it was recorded at.

### Table 11. Voice Channel Parameters

<table>
<thead>
<tr>
<th>Define</th>
<th>Bytes</th>
<th>Read/Write</th>
<th>Default</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DXCH_DFLAGS</td>
<td>2</td>
<td>R/W</td>
<td>0</td>
<td>DTMF detection edge</td>
</tr>
</tbody>
</table>

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### 4. Voice Data Structures and Device Parameters

<table>
<thead>
<tr>
<th>Define</th>
<th>Bytes</th>
<th>Read/Write</th>
<th>Default</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DXCH_DTMFTLK</td>
<td>2</td>
<td>R/W</td>
<td>5</td>
<td>Sets the minimum time for DTMF to be present during playback to be considered valid.</td>
</tr>
</tbody>
</table>

Define Bytes

<table>
<thead>
<tr>
<th>Value</th>
<th>DTMF Digit</th>
</tr>
</thead>
<tbody>
<tr>
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</tr>
<tr>
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<td>2</td>
</tr>
<tr>
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<td>3</td>
</tr>
<tr>
<td>-DM_4</td>
<td>4</td>
</tr>
<tr>
<td>-DM_5</td>
<td>5</td>
</tr>
<tr>
<td>-DM_6</td>
<td>6</td>
</tr>
<tr>
<td>-DM_7</td>
<td>7</td>
</tr>
<tr>
<td>-DM_8</td>
<td>8</td>
</tr>
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<td>-DM_P</td>
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<tr>
<td>-DM_B</td>
<td>b</td>
</tr>
<tr>
<td>-DM_C</td>
<td>c</td>
</tr>
<tr>
<td>-DM_D</td>
<td>d</td>
</tr>
</tbody>
</table>
### Define Bytes

<table>
<thead>
<tr>
<th>Define</th>
<th>Bytes</th>
<th>Read/Write</th>
<th>Default</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>DXCH_DTMFDEB</td>
<td>2</td>
<td>R/W</td>
<td>0</td>
<td>DTMF debounce time - maximum length of time in which DTMF can be absent and then come back on again and still be considered the same DTMF tone.</td>
</tr>
<tr>
<td>DXCH_MFMODE</td>
<td>2</td>
<td>R/W</td>
<td>2</td>
<td>This is a word-length bit mask that selects the minimum length of KP tones to be detected. The possible values of this field are:</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>0 - detect KP tone &gt; 40 ms</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>2 - detect KP tone &gt; 65 ms</td>
</tr>
</tbody>
</table>

Set to -1 to disable.

If the value is set to 2, any KP tone greater than 65ms will be returned to the application during MF detection. This ensures that only standard-length KP tones (100ms) are detected.
### 4. Voice Data Structures and Device Parameters

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<td>10</td>
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<td>R/W</td>
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<td>R/W</td>
<td>15</td>
<td>Wink Length - The duration of a wink in the off-hook state (10 ms units)</td>
</tr>
</tbody>
</table>

If set to 0 (zero), any KP tone longer than 40ms will be detected.
**Define Bytes**

<table>
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<tr>
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<th>Bytes</th>
<th>Read/Write</th>
<th>Default</th>
<th>Description</th>
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<tbody>
<tr>
<td>DXCH_PLAYDRATE</td>
<td>2</td>
<td>R/W</td>
<td>6000</td>
<td>Play Digitization Rate - This parameter sets the digitization rate of the voice data that is played on this channel. Voice Data can be played at 6k or 8k sampling rates. Valid parameter values are: 6000 - 6K sampling rate 8000 - 8k sampling rate Voice data must be played at the same rate it was recorded at.</td>
</tr>
<tr>
<td>DXCH_RECRDRATE</td>
<td>2</td>
<td>R/W</td>
<td>6000</td>
<td>Record Digitization Rate - This parameter sets the rate at which the recorded voice data is digitized. Voice Data can be digitized at 6k or 8k sampling rates. Valid values are: 6000 - 6K</td>
</tr>
</tbody>
</table>

*Voice Programmer’s Guide for Windows NT*
4. Voice Data Structures and Device Parameters

<table>
<thead>
<tr>
<th>Define</th>
<th>Bytes</th>
<th>Read/Write</th>
<th>Default</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>sampling rate</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>8000 - 8k</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>sampling rate</td>
</tr>
</tbody>
</table>
5. Voice Programming Conventions

This chapter provides several techniques that you can use to simplify programming with the Dialogic Voice Library.

5.1. Always Check Return Code in Voice Programming

All the Dialogic Voice Library functions return a value to indicate success or failure of the function. All Voice Library functions indicate success by a return value of zero or a non-negative number.

NOTE: Asynchronous I/O functions return immediately to indicate success or failure of the function initiating.

Extended Attribute functions that return pointers return a pointer to the ASCIIZ string "Unknown device" if they fail.

Extended Attribute functions that do not return pointers return a value of AT_FAILURE if they fail.

Non-attribute functions return a value of -1 to indicate a failure.

If a function has failed, the reason for failure can be found by calling the Standard Attribute functions ATDV_LASTERR() and ATDV_ERRMSGP(). These functions are described in the Standard Runtime Library Programmer’s Guide for Windows NT.

If the error is EDX_SYSTEM check errno.

When using the asynchronous programming model you should always install a handler to get TDX_ERROR events.

5.2. Clearing Voice Structures

Two library functions are provided to clear structures. dx_clrcap() clears DX_CAP structures and dx_clrtpt() clears DV_TPT structures. See the function descriptions 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.

5.3. Using the Voice dx_playf() and dx_recf() Convenience Functions

dx_playf() and dx_recf() are synchronous Voice Library functions provided as a convenience to the programmer. These functions are specific cases of the dx_play() and dx_rec() functions.

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. The dx_recf() provides the same single file convenience for the dx_rec() function.

5.4. Using the Voice Asynchronous Programming Model

Asynchronous programming allows you to have multiple threads of control within the one process. Each of the I/O functions can operate synchronously or asynchronously. See the Standard Runtime Library Programmer’s Guide for Windows NT for information about asynchronous programming models.

5.5. Using Multiple Processes in Voice Synchronous Applications

When writing multiple processes for synchronous applications, you should use the following model: Create a master control process and spawn of a child process for each channel. Each child process is responsible for:

- opening and closing the channel
- adjusting the channel parameters
- performing channel-specific operations
- monitoring events that occur on the channel
NOTE: In an application that spawns a child process from a parent process, a device handle is not inheritable by the child process. Devices must be opened in the child process.
Appendix A

Standard Runtime Library

Voice Device Entries and Returns

The Standard Runtime Library is a device-independent library containing Event Management functions, Standard Attribute functions and the DV_TPT Termination Parameter Table. Dialogic SRL functions and data structures are described fully in the Standard Runtime Library Programmer’s Guide for Windows NT.

This appendix lists the Voice board entries and returns for each of the Standard Runtime Library (SRL) components.

Event Management Functions

The Event Management functions retrieve and handle Voice device termination events for the following functions:

- dx_dial( )
- dx_getdig( )
- dx_play( )
- dx_rec( )
- dx_playtone( )
- dx_sethook( )
- dx_wink( )
- r2_playbsig( )

Each of the Event Management functions applicable to the Voice boards are listed in the following tables. Table 12 and Table 13 list values that are required by or returned for event management functions that are used with Voice devices.

Table 12. Voice Device Inputs for Event Management Functions
### Voice Programmer’s Guide for Windows NT

<table>
<thead>
<tr>
<th>Event Management Function</th>
<th>Voice Device Input</th>
<th>Valid Value</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>sr_enbhdlr()</code></td>
<td><code>evt_type</code></td>
<td><code>TDX_PLAY</code></td>
</tr>
<tr>
<td><em>Enable event handler</em></td>
<td></td>
<td><em>TDX_PLAYTONE</em></td>
</tr>
<tr>
<td></td>
<td></td>
<td><em>TDX_RECORD</em></td>
</tr>
<tr>
<td></td>
<td></td>
<td><em>TDX_GETDIG</em></td>
</tr>
<tr>
<td></td>
<td></td>
<td><em>TDX_DIAL</em></td>
</tr>
<tr>
<td></td>
<td></td>
<td><em>TDX_CALLP</em></td>
</tr>
<tr>
<td></td>
<td></td>
<td><em>TDX_CST</em></td>
</tr>
<tr>
<td></td>
<td></td>
<td><em>TDX_SETHOOK</em></td>
</tr>
<tr>
<td></td>
<td></td>
<td><em>TDX_WINK</em></td>
</tr>
<tr>
<td></td>
<td></td>
<td><em>TDX_ERROR</em></td>
</tr>
</tbody>
</table>

| `sr_dishdlr()`             | `evt_type`         | *As Above*                      |
| *Disable event handler*    |                    |                                  |

**Table 13. Voice Device Returns from Event Management Functions**

<table>
<thead>
<tr>
<th>Event Management Function</th>
<th>Return Description</th>
<th>Returned Value</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>sr_getevtdev()</code></td>
<td><code>device</code></td>
<td>Voice device handle</td>
</tr>
<tr>
<td><em>Get Dialogic Device handle</em></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

| `sr_getevtttype()`         | `event type`       | `TDX_PLAY`                      |
| *Get event type*           |                    | *TDX_PLAYTONE*                  |
|                           |                    | *TDX_RECORD*                    |
|                           |                    | *TDX_GETDIG*                    |
|                           |                    | *TDX_DIAL*                      |
|                           |                    | *TDX_CALLP*                     |
|                           |                    | *TDX_CST*                       |
|                           |                    | *TDX_SETHOOK*                   |
|                           |                    | *TDX_WINK*                      |
|                           |                    | *TDX_ERROR*                     |

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Appendix A - Standard Runtime Library

sr_getevtlens()  
Get event data length

sr_getevtdatap()  
Get pointer to event data

Standard Attribute Functions

Standard Attribute functions return general Dialogic device information, such as the device name or the last error that occurred on the device. The Standard Attribute functions and the Voice device information they return are listed in Table 14.

Table 14. Standard Attribute Functions

<table>
<thead>
<tr>
<th>Standard Attribute Function</th>
<th>Information Returned for Voice Devices</th>
</tr>
</thead>
<tbody>
<tr>
<td>ATDV_ERRMSGP()</td>
<td>Pointer to string describing the error that occurred during the last function call on the specified device. (See Appendix B for a list of all the possible errors, and see each function description for possible errors for that function).</td>
</tr>
<tr>
<td>ATDV_IOPORT()</td>
<td>0 for D/21D, D/41D, D/21E, D/41E, D/41ESC, D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1, and D/320SC boards. Valid port address of the SpringBoard device.</td>
</tr>
<tr>
<td>ATDV_IRQNUM()</td>
<td>Interrupt number for the specified device.</td>
</tr>
<tr>
<td>ATDV_LASTERR()</td>
<td>The error that occurred during the last function call on a specified device. See the function description for possible errors for the function.</td>
</tr>
<tr>
<td>ATDV_NAMEP()</td>
<td>Pointer to device name (e.g., dxxxBbCc). Refer</td>
</tr>
</tbody>
</table>
Voice Programmer’s Guide for Windows NT

to the System Release Software Installation Reference for Windows NT for information about device names.

ATDV_SUBDEVS( )

Number of sub-devices (channels) (Refer to the Standard Runtime Library Programmer’s Guide for Windows NT for information on sub-devices):

- 4 for a D/4x board (emulated or real)
- 2 for a D/2x board

DV_TPT Structure

The DV_TPT termination parameter table sets termination conditions for a range of Dialogic products. The valid values for the DV_TPT when using a Voice board are contained in this section.

The DV_TPT structure is used to set I/O function termination conditions. This structure is used by the following I/O functions:

- dx_clrtpt()
- dx_getdig()
- dx_play()
- dx_rec()
- dx_playtone()

The I/O functions will terminate when one of the conditions set in the DV_TPT structure occurs. 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 terminating condition.

The structure has the following format:

```c
typedef struct DV_TPT {
    unsigned short tp_type;   /* Flags describing this entry */
    unsigned short tp_termno; /* Termination Parameter number */
    unsigned short tp_length; /* Length of terminator */
    unsigned short tp_flags;  /* Parameter attribute flag */
    unsigned short tp_data;   /* Optional additional data */
    unsigned short rfu;       /* Reserved */
    DV_TPT *tp_nextp;         /* Pointer to next termination parameter if IO_LINK set */
} DV_TPT;
```
Appendix A - Standard Runtime Library

Each field is defined in the sections that follow. Table 15 located after the field descriptions, contains a summary of the valid field settings for each termination condition.

**tp_type**

*tp_type* specifies whether the structure is part of a linked list, part of an array, or the last DV_TPT entry in the DV_TPT table. Enter one of the following defines in *tp_type*:

- **IO_LINK**
  - *tp_nextp* points to next DV_TPT structure
- **IO_EOT**
  - last DV_TPT in the chain
- **IO_CONT**
  - next DV_TPT entry is contiguous

**tp_termno**

*tp_termno* specifies the termination condition. The Voice device termination defines are:

- **DX_MAXDTMF**
  - Maximum number of digits received
- **DX_MAXSIL**
  - Maximum length of silence
- **DX_MAXNOSIL**
  - Maximum length of non-silence
- **DX_LCOFF**
  - Loop current drop
- **DX_IDDTIME**
  - Maximum delay between digits
- **DX_MAXTIME**
  - Maximum function time
- **DX_DIGMASK**
  - Specific digit received
- **DX_PMOFF**
  - Pattern of non-silence
- **DX_PMON**
  - Pattern of silence
  - Digit termination for user-defined tone. (D/21D, D/41D, D/21E, D/41E, D/41ESC, D/81A, D/12x, D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1 and D/320SC only).
- **DX_TONE**
  - Tone On/Off termination (D/21D, D/41D, D/41E, D/41ESC, D/81A, D/12x, D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1 and D/320SC only GTD term conditions)
A more detailed description of these I/O terminations is contained in the Voice Features Guide for Windows NT.

**NOTE:** When using the DX_PMON and DX_PMOFF termination conditions, some of the DV_TPT fields are set differently from the other termination conditions. See the section, *Using DX_PMOFF and DX_PMON*, located at the end of this appendix for information.

You can call the Extended Attribute function **ATDX TERMMSK( )** to determine all the terminating conditions that occurred. This function returns a bitmap of terminating conditions. The "TM_" defines corresponding to this bitmap of terminating conditions are provided in the function description for **ATDX TERMMSK( )**.

**tp_length**

**tp_length** refers to the length or size for each specific terminating condition. When **tp_length** represents length of time for a terminating condition, the maximum value allowed is 6000. The field can represent the following:

**Table 15. tp_length Settings**

<table>
<thead>
<tr>
<th>tp_length value</th>
<th>tp_length description</th>
</tr>
</thead>
<tbody>
<tr>
<td>time in 10 or 100ms units</td>
<td>Applies to any terminating condition that specifies termination after a specific period of time, up to 6000.</td>
</tr>
<tr>
<td># of digits</td>
<td>Applies when using DX_MAXDTMF which specifies termination after a certain number of digits is received.</td>
</tr>
<tr>
<td>digit type description</td>
<td>Applies when using DX_DIGTYPE which specifies termination on a user-specified digit. Specify the digit type in the high byte and the ASCII digit value in the low byte. See the Global Tone Detection functions in the Voice Features Guide for Windows NT for information.</td>
</tr>
</tbody>
</table>
| digit bit mask | Applies to DX_DIGMASK, which specifies a bit mask of digits to terminate on. Set the digit bitmask using one or
more of the appropriate "Digit Defines" from the table below:

<table>
<thead>
<tr>
<th>Digit</th>
<th>Digit Define</th>
<th>Digit</th>
<th>Digit Define</th>
<th>Digit</th>
<th>Digit Define</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>DM_0</td>
<td>6</td>
<td>DM_6</td>
<td>#</td>
<td>DM_P</td>
</tr>
<tr>
<td>1</td>
<td>DM_1</td>
<td>7</td>
<td>DM_7</td>
<td>a</td>
<td>DM_A</td>
</tr>
<tr>
<td>2</td>
<td>DM_2</td>
<td>8</td>
<td>DM_8</td>
<td>b</td>
<td>DM_B</td>
</tr>
<tr>
<td>3</td>
<td>DM_3</td>
<td>9</td>
<td>DM_9</td>
<td>c</td>
<td>DM_C</td>
</tr>
<tr>
<td>4</td>
<td>DM_4</td>
<td>*</td>
<td>DM_S</td>
<td>d</td>
<td>DM_D</td>
</tr>
</tbody>
</table>

number of pattern repetitions Applies to DX_PMOFF, which specifies the number of times a pattern should repeat before termination.

**NOTE:** Then DX_PMON is the termination condition, *tp_length* contains the *tp_flags* information. See the *tp_flags* description and the Using DX_PMON and DX_PMOFF section (at the end of this Appendix) for information.

### tp_flags

*tp_flags* is a bit mask representing various characteristics of the termination condition to use.

The defines for the termination flags are:

- **TF_EDGE** • termination condition is edge-sensitive
- **TF_LEVEL** • termination condition is level-sensitive
- **TF_CLREND** • history cleared when function terminates
- **TF_CLRBEG** • history cleared when function begins
- **TF_USE** • terminator used for termination
- **TF_SETINIT** • DX_MAXSIL only - initial length of silence to terminate on
- **TF_10MS** • set units of time to 10 ms (default is 100 ms)
- **TF_FIRST** • DX_IDDTIME only - start looking for termination condition (interdigit delay) to be satisfied after first digit is received
A set of default \texttt{tp\_flags} values for the termination conditions is available. These default values are:

\begin{verbatim}
TF_MAXDTMF          (TF_LEVEL|TF_USE)
TF_MAXSIL           (TF_EDGE|TF_USE)
TF_MAXNOSIL         (TF_EDGE|TF_USE)
TF_LCOFF            (TF_LEVEL|TF_USE|TF_CLREND)
TF_IDDTIME          (TF_EDGE)
TF_MAXTIME          (TF_EDGE)
TF_DIGMASK          (TF_LEVEL)          (TF_EDGE)
TF_TONE             (TF_LEVEL|TF_USE|TF_CLREND)
TF_DIGTYPE          (TF_LEVEL)
\end{verbatim}

\textbf{NOTES:}  
1. DX_PMOFF and DX_PMON  
2. DX_PMOFF does not have a default \texttt{tp\_flags} value.  
The \texttt{tp\_flags} value for \texttt{is} is set in \texttt{tp\_length} (i.e., TF_PMON is set in \texttt{tp\_length}). See the \texttt{tp\_length} description and the \textit{Using DX_PMON and DX_PMOFF} section (at the end of this Appendix) for information.  
3. TF_IDDTIME or TF_MAXTIME must be specified in \texttt{tp\_flags} if DX_IDDTIME or DX_MAXTIME are specified in \texttt{tp\_termno}. Other flags may be set at the same time using an OR combination.

The bitmap for the \texttt{tp\_flags} field is as follows:

\begin{verbatim}
| rfu | rfu | units | ini | use | beg | end | level |
\hline
| 7   |    | bits  | 0   |     |     |     |       |
\end{verbatim}

The descriptions of each bit are listed below:

bit 0 (level): If set, the termination condition is level-sensitive.  
Level-sensitive means that if the condition is satisfied when the function starts, termination will occur immediately.
Terminating conditions that can be level have a 'history' associated with them which records the state of the terminator before the function started. If this bit is not set, the termination condition is edge-sensitive and the function will not terminate unless the condition occurs after the function starts. The table below shows which terminating conditions can be edge-sensitive and which can be level-sensitive.

**NOTE:** A level-sensitive termination condition only has to have occurred sometime in the history associated with that terminator to cause the function to terminate; the condition does not have to be present when the function starts in order to terminate the function.

<table>
<thead>
<tr>
<th>Term. Condition</th>
<th>Level-sensitive</th>
<th>Edge-sensitive</th>
</tr>
</thead>
<tbody>
<tr>
<td>DX_DIGTYPE</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>DX_MAXDTMF</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>DX_MAXSIL</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>DX_MAXNOSIL</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>DX_LCOFF</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>DX_DIGMASK</td>
<td>X</td>
<td>X</td>
</tr>
<tr>
<td>DX_IDDTIME</td>
<td>-</td>
<td>X</td>
</tr>
<tr>
<td>DX_MAXTIME</td>
<td>-</td>
<td>X</td>
</tr>
<tr>
<td>DX_PMON/OFF</td>
<td>-</td>
<td>X</td>
</tr>
<tr>
<td>DX_TONE</td>
<td>X</td>
<td>X</td>
</tr>
</tbody>
</table>
bit 1 (end): If set, the history of this terminator will be cleared when the function terminates. This bit has special meaning for DX_IDDTIME. If set, the terminator will be started after the first digit is received. Otherwise it will be started as soon as the function is started.

bit 2 (beg): If set, the history of this terminator will be cleared when the function starts. This bit will override the level bit (bit 0). If both are set, the history will be cleared and no past history of this terminator will be taken into account.

bit 3 (use): If this bit is set, the terminator will be used for termination. If the bit is not set, the history for the terminator will be cleared (depending on bits 1 & 2), but the terminator will still not be used for termination. This bit is not valid for the following terminating conditions:

- DX_MAXTIME
- DX_IDDTIME
- DX_DIGMASK
- DX_PMOFF
- DX_PMON

bit 4 (ini): This bit is only used for DX_MAXSIL termination. If the termination is edge-sensitive and this bit is set, the tp_data field should contain an initial length of silence to terminate upon if silence is detected before non-silence. In general, the initial length of silence to terminate on in tp_data should be greater than the value in tp_length.

If the termination is level sensitive then this bit must be set to 0 and tp_length will be used for the termination.

bit 5 (units): If set the units of time will be in 10 ms. The default is 100 ms units.

NOTES: 1. The 10 ms timer resolution is only available with version 0.62 or later of the D/4x firmware.
2. `tp_flags` is used differently for DX_PMON.
   See Using DX_PMOFF and DX_PMON at the end of this Appendix for information.

**tp_data**

`tp_data` specifies optional additional data. This bit can be set as follows:

**Table 16. tp_data Valid Values**

<table>
<thead>
<tr>
<th>Value in tp_termno</th>
<th>tp_data Entry</th>
</tr>
</thead>
<tbody>
<tr>
<td>DX_MAXSIL</td>
<td>initial length of silence to terminate on</td>
</tr>
<tr>
<td>DX_PMOFF</td>
<td>maximum time of silence off</td>
</tr>
<tr>
<td>DX_PMON</td>
<td>maximum time of silence on</td>
</tr>
<tr>
<td></td>
<td>DX_TONEON - terminate after a &quot;tone-on&quot; event</td>
</tr>
<tr>
<td></td>
<td>DX_TONEOFF - terminate after a &quot;tone-off&quot; event</td>
</tr>
</tbody>
</table>

**tp_nextp**

`tp_nextp` contains a pointer to the next DV_TPT structure in a linked list. The `tp_type` field must be set to IO_LINK in this case.

The table that follows indicates how DV_TPT fields should be filled.

**NOTE:** An asterisk indicates the default `tp_flags` setting, defined when `tp_flags` is set to TF_{(term name)}, where (term name) is the suffix of the `tp_termno` setting, as in DX_{(term name)}. To override defaults, set the bits in `tp_flags` individually, as required.

**Table 17. DV_TPT Fields Settings Summary**

**NOTE:** The `tp_flags` column describes the effect of the field when set to one and not set to one. "*" indicates the default value for each bit. The defaults defines for the `tp_flags` field are listed in the `tp_flags` description in this Appendix.
### Voice Programmer's Guide for Windows NT

<table>
<thead>
<tr>
<th>tp_errno</th>
<th>tp_type</th>
<th>tp_length</th>
<th>tp_flags: not set</th>
<th>tp_flags: set</th>
<th>tp_data</th>
<th>tp_nextp</th>
</tr>
</thead>
<tbody>
<tr>
<td>DX_MAXDTMF</td>
<td>IO_LINK</td>
<td>IO_EOT</td>
<td>IO_CONT</td>
<td>max number of digits</td>
<td>bit 0: TF_EDGE</td>
<td>TF_LEVEL*</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 1: no clr*</td>
<td>bit 2: no clr*</td>
<td>TF_CLREN</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 3: clr hist</td>
<td></td>
<td>TF_CLRBEG</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 4: no-clr</td>
<td></td>
<td>TF_USE*</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 5: 100ms*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>DX_MAXSIL</td>
<td>IO_LINK</td>
<td>IO_EOT</td>
<td>IO_CONT</td>
<td>max length silence</td>
<td>bit 0: TF_EDGE*</td>
<td>TF_LEVEL*</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 1: no clr</td>
<td>bit 2: no clr*</td>
<td>TF_CLREN</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 3: clr hist</td>
<td>bit 4: no-setinit</td>
<td>TF_CLRBEG</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 5: 100ms*</td>
<td></td>
<td>TF_USE*</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>TF_SETINIT</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>TF_10MS</td>
</tr>
<tr>
<td>DX_MAXNOSIL</td>
<td>IO_LINK</td>
<td>IO_EOT</td>
<td>IO_CONT</td>
<td>max length non-silence</td>
<td>bit 0: TF_EDGE*</td>
<td>TF_LEVEL*</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 1: no clr</td>
<td>bit 2: no clr*</td>
<td>TF_CLREN</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 3: clr hist</td>
<td>bit 4: N/A</td>
<td>TF_CLRBEG</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 5: 100ms*</td>
<td></td>
<td>TF_USE*</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>TF_SETINIT</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>TF_10MS</td>
</tr>
<tr>
<td>DX_LCOFF</td>
<td>IO_LINK</td>
<td>IO_EOT</td>
<td>IO_CONT</td>
<td>max length loop current drop</td>
<td>bit 0: TF_EDGE*</td>
<td>TF_LEVEL*</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 1: no clr</td>
<td>bit 2: no clr*</td>
<td>TF_CLREN</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 3: clr hist</td>
<td>bit 4: N/A</td>
<td>TF_CLRBEG</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 5: 100ms*</td>
<td></td>
<td>TF_USE*</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>TF_SETINIT</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>TF_10MS</td>
</tr>
<tr>
<td>DX_IDDTIME</td>
<td>IO_LINK</td>
<td>IO_EOT</td>
<td>IO_CONT</td>
<td>max length interdigit delay</td>
<td>bit 0: TF_EDGE*</td>
<td>N/A</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 1: strt@call*</td>
<td>bit 2: N/A</td>
<td>strt@1st</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 3: N/A</td>
<td>bit 4: N/A</td>
<td>N/A</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 5: 100ms*</td>
<td></td>
<td>N/A</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>TF_10MS</td>
</tr>
<tr>
<td>DX_MAXTIME</td>
<td>IO_LINK</td>
<td>IO_EOT</td>
<td>IO_CONT</td>
<td>max length function time</td>
<td>bit 0: TF_EDGE*</td>
<td>N/A</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 1: N/A</td>
<td>bit 2: N/A</td>
<td>N/A</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 3: N/A</td>
<td>bit 4: N/A</td>
<td>N/A</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td>bit 5: 100ms*</td>
<td></td>
<td>N/A</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>TF_10MS</td>
</tr>
<tr>
<td>DX_DIGMASK</td>
<td>IO_LINK</td>
<td>bit 0: d (set)</td>
<td>bit 0: TF_LEVEL*</td>
<td>N/A</td>
<td>pointer to</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

390-CD
### Appendix A - Standard Runtime Library

<table>
<thead>
<tr>
<th>Field</th>
<th>Description</th>
<th>Value</th>
<th>Pointer</th>
</tr>
</thead>
<tbody>
<tr>
<td>IO_CONT</td>
<td>number of pattern repetitions</td>
<td>minimum time silence off max time silence off</td>
<td></td>
</tr>
<tr>
<td>DX_PM+OFF</td>
<td>IO_LINK IO_EOT IO_CONT</td>
<td>bit 0: TF_EDGE* / TF_LEVEL*</td>
<td>pointer to next DV_TPT if linked list</td>
</tr>
<tr>
<td></td>
<td>bit 1: N/A bit 2: N/A bit 3: N/A bit 4: N/A bit 5: 100ms/TF_10MS</td>
<td>maximum time silence on max time silence on</td>
<td></td>
</tr>
<tr>
<td>DX_PMON</td>
<td>IO_LINK IO_EOT IO_CONT</td>
<td>bit 0: TF_EDGE / TF_LEVEL</td>
<td>pointer to next DV_TPT if linked list</td>
</tr>
<tr>
<td></td>
<td>bit 1: no clr bit 2: no clr* bit 3: clr hist</td>
<td>max time silence on</td>
<td></td>
</tr>
<tr>
<td>DX_TONE</td>
<td>IO_LINK IO_EOT IO_CONT</td>
<td>Tone ID</td>
<td>pointer to next DV_TPT if linked list</td>
</tr>
<tr>
<td></td>
<td>low byte:ASCII val. *hi byte: digit type</td>
<td>bit 0: TF_EDGE / TF_LEVEL</td>
<td></td>
</tr>
<tr>
<td>DX_DIGTYPE</td>
<td>IO_LINK IO_EOT IO_CONT</td>
<td>bit 0: TF_EDGE</td>
<td>N/A</td>
</tr>
</tbody>
</table>

*The `tp_flags` column describes the effect of the field when set to one and not set to one. "*" indicates the default value for each bit. The defaults defines for the `tp_flags` field are listed in the `tp_flags` description in this Appendix.

Since DX_PMON requires that the `tp_length` field is set with `tp_flags` values, the previous statements apply to `tp_length` for DX_PMON.
Using DX_PMOFF and DX_PMON

The DX_PMOFF and DX_PMON termination conditions must be used in tandem. In other words, the DX_PMON terminating condition must directly follow the DX_PMOFF terminating condition. A combination of both DV_TPT structures using these conditions is used to form a single termination condition. When used, both must be specified together or else an error will result in the execution of the function.

In the first block, tp_termno is set to DX_PMOFF. The tp_length holds the number of patterns before termination. tp_flags holds the minimum time for silence off while tp_data holds the maximum time for silence off. In the next DV_TPT structure, tp_termno is DX_PMON, and the tp_length field holds the flag bit mask as shown above. Only the "units" bit is valid; all other bits must be 0. The tp_flags field holds the minimum time for silence on, while tp_data holds the maximum time for silence on. An example of this would be:

**DV_TPT Example**

```c
#include <srllib.h>
#include <dxxxlib.h>
#include <windows.h>
DV_TPT   tpt[2];
/*
 * detect a pattern which repeats 4 times of approximately 2 seconds
 * off 2 seconds on.
 */
tpt[0].tp_type = IO_CONT;    /* next entry is contiguous */
tpt[0].tp_termno = DX_PMOFF; /* specify pattern match off */
tpt[0].tp_length = 4;        /* terminate if pattern repeats 4 times */
tpt[0].tp_flags = 175;       /* minimum silence off is 1.75 seconds */
                           /* (10 ms units) */
tpt[0].tp_data   = 225;      /* maximum silence off is 2.25 seconds */
                           /* (10 ms units) */
tp[1].tp_type = IO_EOT;    /* This is the last in the chain */
tp[1].tp_termno = DX_PMON;  /* specify pattern match on */
tp[1].tp_length = TF_10MS;  /* use 10 ms timer units */
tp[1].tp_flags = 175;       /* minimum silence on is 1.75 seconds */
                           /* (10 ms units) */
tp[1].tp_data = 225;        /* maximum silence on is 2.25 seconds */
                           /* (10 ms units) */
/* issue the function */
```
Appendix B

Error Defines

Errors - Voice Library

This appendix lists the error defines that may be returned for the Voice Library functions.

For error codes returned for SCbus functions and a description of the error, refer to the SCbus Routing Function Reference for Windows NT.

The following table contains the list of errors that can be returned using ATDV_LASTERR() and ATDV_ERRMSGP() functions.

<table>
<thead>
<tr>
<th>Error Define</th>
<th>Error String</th>
</tr>
</thead>
<tbody>
<tr>
<td>EDX_AMPLGEN</td>
<td>Invalid Amplitude Value in Tone Generation Template</td>
</tr>
<tr>
<td>EDX_ASCII</td>
<td>Invalid ASCII Value in Tone Template Description</td>
</tr>
<tr>
<td>EDX_BADDEV</td>
<td>Device Descriptor error</td>
</tr>
<tr>
<td>EDX_BADIOTT</td>
<td>DX_IOTT structure error</td>
</tr>
<tr>
<td>EDX_BADPARM</td>
<td>Parameter error</td>
</tr>
<tr>
<td>EDX_BADPROD</td>
<td>Function Not Supported on this Board</td>
</tr>
<tr>
<td>EDX_BADTPT</td>
<td>DX_TPT structure error</td>
</tr>
<tr>
<td>EDX_BUSY</td>
<td>Device or channel is Busy</td>
</tr>
<tr>
<td>EDX_CADENCE</td>
<td>Invalid Cadence Component Values in Tone Template Description</td>
</tr>
<tr>
<td>EDX_CHANNUM</td>
<td>Invalid Channel Number Specified</td>
</tr>
<tr>
<td>EDX_DIGTYPE</td>
<td>Invalid Dig_type Value in Tone Template Description</td>
</tr>
<tr>
<td>EDX_FLAGGEN</td>
<td>Invalid tn_dflag field in Tone Generation Template</td>
</tr>
<tr>
<td>EDX_FREQDET</td>
<td>Invalid Frequency Component Values in Tone Template Description</td>
</tr>
<tr>
<td>EDX_FREQGEN</td>
<td>Invalid Frequency Component in Tone</td>
</tr>
<tr>
<td>Code</td>
<td>Description</td>
</tr>
<tr>
<td>-------------</td>
<td>-----------------------------------------------------------------------------</td>
</tr>
<tr>
<td>EDX_FWERROR</td>
<td>Firmware Error</td>
</tr>
<tr>
<td>EDX_IDLE</td>
<td>Device is Idle</td>
</tr>
<tr>
<td>EDX_INVSUBCMD</td>
<td>Invalid sub-command number</td>
</tr>
<tr>
<td>EDX_MAXTPLMT</td>
<td>Max number of Tone Templates Exist or user-defined tones for the board[from r2_creatfsig()]</td>
</tr>
<tr>
<td>EDX_MSGSTATUS</td>
<td>Invalid Message Status Setting</td>
</tr>
<tr>
<td>EDX_NOERROR</td>
<td>No Error</td>
</tr>
<tr>
<td>EDX_NONZEROSIZE</td>
<td>Reset to Default was Requested but size was non-zero</td>
</tr>
<tr>
<td>EDX_SPDVL</td>
<td>Must Specify either SV_SPEEDTBL or SV_VOLUMETBL</td>
</tr>
<tr>
<td>EDX_SVADJBLKS</td>
<td>Invalid Number of Speed/Volume Adjustment Blocks</td>
</tr>
<tr>
<td>EDX_SVMTRANGE</td>
<td>An Entry in SV_SVMT was out of Range</td>
</tr>
<tr>
<td>EDX_SVMTSIZE</td>
<td>Invalid Table Size Specified</td>
</tr>
<tr>
<td>EDX_SYSTEM</td>
<td>Windows NT System Error; check the global variable errno for more information</td>
</tr>
<tr>
<td>EDX_TIMEOUT</td>
<td>I/O Function Timed Out</td>
</tr>
<tr>
<td>EDX_TONEID</td>
<td>Invalid Tone Template ID</td>
</tr>
</tbody>
</table>
# Appendix C

## DTMF and MF Tone Specifications

The following two charts show the tone specifications for MF and DTMF tones.

### MF Tone Specifications (CCITT R1 Tone Plan)

<table>
<thead>
<tr>
<th>Code</th>
<th>Tone Pair Frequencies (Hz)</th>
<th>Default Length (ms)</th>
<th>Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>700, 900</td>
<td>60</td>
<td>1</td>
</tr>
<tr>
<td>2</td>
<td>700, 1100</td>
<td>60</td>
<td>2</td>
</tr>
<tr>
<td>3</td>
<td>900, 1100</td>
<td>60</td>
<td>3</td>
</tr>
<tr>
<td>4</td>
<td>700, 1300</td>
<td>60</td>
<td>4</td>
</tr>
<tr>
<td>5</td>
<td>900, 1300</td>
<td>60</td>
<td>5</td>
</tr>
<tr>
<td>6</td>
<td>1100, 1300</td>
<td>60</td>
<td>6</td>
</tr>
<tr>
<td>7</td>
<td>700, 1500</td>
<td>60</td>
<td>7</td>
</tr>
<tr>
<td>8</td>
<td>900, 1500</td>
<td>60</td>
<td>8</td>
</tr>
<tr>
<td>9</td>
<td>1100, 1500</td>
<td>60</td>
<td>9</td>
</tr>
<tr>
<td>0</td>
<td>1300, 1500</td>
<td>60</td>
<td>0</td>
</tr>
<tr>
<td>*</td>
<td>1100, 1700</td>
<td>100</td>
<td>KP</td>
</tr>
<tr>
<td>#</td>
<td>1500, 1700</td>
<td>60</td>
<td>ST</td>
</tr>
<tr>
<td>a</td>
<td>900, 1700</td>
<td>60</td>
<td>ST1</td>
</tr>
<tr>
<td>b</td>
<td>1300, 1700</td>
<td>60</td>
<td>ST2</td>
</tr>
<tr>
<td>c</td>
<td>700, 1700</td>
<td>60</td>
<td>ST3</td>
</tr>
</tbody>
</table>
Voice Programmer’s Guide for Windows NT

* The standard length of a KP tone is 100 ms.
## DTMF Tone Specifications

<table>
<thead>
<tr>
<th>Code</th>
<th>Tone Pair Frequencies (Hz)</th>
<th>Default Length (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>697, 1209</td>
<td>100</td>
</tr>
<tr>
<td>2</td>
<td>697, 1336</td>
<td>100</td>
</tr>
<tr>
<td>3</td>
<td>697, 1477</td>
<td>100</td>
</tr>
<tr>
<td>4</td>
<td>770, 1209</td>
<td>100</td>
</tr>
<tr>
<td>5</td>
<td>770, 1336</td>
<td>100</td>
</tr>
<tr>
<td>6</td>
<td>770, 1477</td>
<td>100</td>
</tr>
<tr>
<td>7</td>
<td>852, 1209</td>
<td>100</td>
</tr>
<tr>
<td>8</td>
<td>852, 1336</td>
<td>100</td>
</tr>
<tr>
<td>9</td>
<td>852, 1477</td>
<td>100</td>
</tr>
<tr>
<td>0</td>
<td>941, 1336</td>
<td>100</td>
</tr>
<tr>
<td>*</td>
<td>941, 1209</td>
<td>100</td>
</tr>
<tr>
<td>#</td>
<td>941, 1477</td>
<td>100</td>
</tr>
<tr>
<td>a</td>
<td>697, 1633</td>
<td>100</td>
</tr>
<tr>
<td>b</td>
<td>770, 1633</td>
<td>100</td>
</tr>
<tr>
<td>c</td>
<td>852, 1633</td>
<td>100</td>
</tr>
<tr>
<td>d</td>
<td>941, 1633</td>
<td>100</td>
</tr>
</tbody>
</table>
Using MF Detection

Some MF digits use approximately the same frequencies as DTMF digits (see above charts). Because there is a frequency overlap, if you have the incorrect kind of detection enabled, MF digits may be mistaken for DTMF digits, and vice versa. To ensure that digits are correctly detected, only one kind of detection should be enabled at any time.

Digit detection accuracy depends on two things:

- which digit is sent
- the kind of detection enabled when the digit is detected

The two tables that follow show the digits that are detected when each type of detection is enabled. Table 19 shows which digits are detected when MF digits are sent. Table 20 shows which digits are detected when DTMF digits are sent.
**Appendix C - DTMF and MF Tone Specifications**

**Table 19. Detecting MF Digits**

<table>
<thead>
<tr>
<th>MF Digit Sent</th>
<th>Only MF Detection Enabled</th>
<th>Only DTMF Detection Enabled</th>
<th>MF and DTMF Detection Enabled</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>2</td>
<td>2</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>3</td>
<td>3</td>
<td>3</td>
<td>3</td>
</tr>
<tr>
<td>4</td>
<td>4</td>
<td>2*</td>
<td>4,2*</td>
</tr>
<tr>
<td>5</td>
<td>5</td>
<td>5</td>
<td>5</td>
</tr>
<tr>
<td>6</td>
<td>6</td>
<td>6</td>
<td>6</td>
</tr>
<tr>
<td>7</td>
<td>7</td>
<td>3*</td>
<td>7,3*</td>
</tr>
<tr>
<td>8</td>
<td>8</td>
<td></td>
<td>8</td>
</tr>
<tr>
<td>9</td>
<td>9</td>
<td></td>
<td>9</td>
</tr>
<tr>
<td>0</td>
<td>0</td>
<td></td>
<td>0</td>
</tr>
<tr>
<td>*</td>
<td>*</td>
<td>*</td>
<td>*</td>
</tr>
<tr>
<td>#</td>
<td>#</td>
<td></td>
<td>#</td>
</tr>
<tr>
<td>a</td>
<td>a</td>
<td></td>
<td>a</td>
</tr>
<tr>
<td>b</td>
<td>b</td>
<td></td>
<td>b</td>
</tr>
<tr>
<td>c</td>
<td>c</td>
<td></td>
<td>c</td>
</tr>
</tbody>
</table>

* = detection error
### Table 20. Detecting DTMF Digits

<table>
<thead>
<tr>
<th>String Received</th>
<th>DTMF Digit Sent</th>
<th>Only DTMF Detection Enabled</th>
<th>Only MF Detection Enabled</th>
<th>DTMF &amp; MF Detection Enabled</th>
</tr>
</thead>
<tbody>
<tr>
<td>Only DTMF Detection Enabled</td>
<td>Only MF Detection Enabled</td>
<td>DTMF &amp; MF Detection Enabled</td>
<td></td>
<td></td>
</tr>
<tr>
<td>1 1</td>
<td>1</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>2 2</td>
<td>4*</td>
<td>4,2*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>3 3</td>
<td>7*</td>
<td>7,3*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>4 4</td>
<td></td>
<td>4</td>
<td></td>
<td></td>
</tr>
<tr>
<td>5 5</td>
<td>4*</td>
<td>4,5*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>6 6</td>
<td>7*</td>
<td>7,6*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>7 7</td>
<td></td>
<td>7</td>
<td></td>
<td></td>
</tr>
<tr>
<td>8 8</td>
<td>5*</td>
<td>5,8*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>9 9</td>
<td>8*</td>
<td>8,9*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>0 0</td>
<td>5*</td>
<td>5,0*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>* *</td>
<td>*</td>
<td>*</td>
<td></td>
<td></td>
</tr>
<tr>
<td># #</td>
<td>8*</td>
<td>8,#*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>a a</td>
<td>c*</td>
<td>c,a*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>b b</td>
<td>c*</td>
<td>c,b*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>c c</td>
<td>a*</td>
<td>a,c*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>d d</td>
<td>a*</td>
<td>a,d*</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

* = detection error
Appendix D
Related Voice Publications

For more information on Voice hardware and software products see the following Dialogic publications:

- For information about installing Voice software, see the *System Release Software Installation Reference for Windows NT*.
- For information about the Standard Runtime Library, see the *Standard Runtime Library Programmer’s Guide for Windows NT*.
- For information about the SCbus, see the *SCbus Routing Guide* and the *SCbus Routing Function Reference for Windows NT*.
- For information about the D/2x, D/4x, D/81A, D/12x, and D/xxxSC (D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1 and D/320SC) voice boards, see the *Quick Install Cards* shipped with the boards.
- For information about the digital interface device functions for the D/240SC-T1 and D/300SC-E1 boards, see the *Digital Network Interface Software Reference for Windows NT*.
- For information about the primary rate functions, see the *Primary Rate Software Reference for Windows NT*. 
Glossary

Mu-law: (1) A pulse-code modulation (PCM) algorithm used in digitizing telephone audio signals in T-1 areas. (2) The PCM coding and compounding standard used in Japan and North America.

A-LAW: Pulse Code Modulation (PCM) algorithm used in digitizing telephone audio signals in E-1 areas.

Adaptive Differential Pulse Code Modulation: See ADPCM.

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 also reduces storage requirements from 64K bits/second to as low as 24K bits/second.

ADSI: Analog Display Services Interface. A Bellcore standard defining a protocol on the flow of information between a switch, a server, a voice mail system, a service bureau, or a similar device and a subscriber’s telephone, PC, data terminal, or other communicating device with a screen. The idea of ADSI is to add words to a system that usually only uses touch tones. In a typical voice mail system, you call up and hear choices: “to listen to new messages, press 1, to hear saved messages, press 2,” etc. ADSI is designed to display the choices you’re hearing on a screen attached to your phone. Your response is the same: a touch tone button. ADSI’s signaling is DTMF and standard Bell 202 modem signals from the service to your 202-modem equipped phone. From the phone to the service it’s only touch tone. ADSI works on every phone line in the world.

AGC: Automatic Gain Control. An electronic circuit used to maintain the audio signal volume at a constant level.

AMIS: Audio Messaging Interchange Specification. A series of standards aimed at addressing the problem of how voice messaging systems produced by different vendors can network or inter-network. It deals specifically with the interaction of the systems and does not affect the systems themselves. There are two specifications: 1. AMIS-digital: All the control information and the voice messages are ported between systems digitally. 2. AMIS-analog: Control information and messages are transferred in analog form. For AMIS specifications, call Hartfield Associates (Boulder, CO) at (303) 442-5395.
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.

Analog Expansion Bus (AEB): Analog electrical connection (bus) between Dialogic network interface modules and analog resource modules. The AEB interfaces network boards and voice boards, which fit in the AT-expansion slot of a PC. See Also PEB, SCSA

ANI: Automatic Number Identification.

Antares: A Dialogic open platform for easily incorporating speech recognition, Text-To-Speech, fax and many other DSP technologies. Dialogic PC-based expansion board with fourTI floating point DSPs, SPOX DSP operating system, and the Antares board downloadable firmware and device driver.

API: See Application Programming Interface

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. See synchronous function.

AT: Used to describe an IBM or IBM-compatible Personal Computer (PC) containing an 80286 or higher microprocessor, a 16-bit bus architecture, and a compatible BIOS.

AT bus: The common communication channel in a PC AT. The channel uses a 16-bit data path architecture, which allows up to 16 bits of data transfer. This bus architecture includes the standard PC bus plus a set of 36 lines for additional data transmission, addressing, and interrupt request handling.

Automatic Gain Control: See AGC.

base memory address: A starting memory location (address) from which other addresses are referenced.
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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: A board-level object that can be manipulated by a physical library. Board devices can be real physical devices, such as a D/4x board, or emulated devices, such as one of the D/4x boards that is emulated by a D/81A, D/12x or D/xxxSC board.

Board Locator Technology: Operates in conjunction with a rotary switch to determine and set non-conflicting slot and IRQ interrupt-level parameters, thus eliminating the need to set confusing jumpers or DIP switches.

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 which allows communication between multiple points or devices in a system.

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

cadence: A rhythmic sequence or pattern. Once established, it can be classified as a single ring, a double ring, or a busy signal by comparing the periods of sound and silence to establish parameters.

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: The process used to automatically determine what happened after an outgoing call is dialed. Also referred to as call analysis or call progress.

Call Status Transition Event Functions: Functions that set and monitor events on devices.

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 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 subdevice.

**channel:** 1. When used in reference to a Dialogic expansion board that is analog, 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 a Dialogic expansion board that is digital, 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.

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

**computer telephony:** The extension of computer-based intelligence and processing over the telephone network to a telephone. Lets you interact with computer databases or applications from a telephone and also enables computer-based applications to access the telephone network. Computer telephony makes computer-based information readily available over the worldwide telephone network from your telephone. Computer telephony technology incorporated into PCs supports applications such as: automatic call processing; automatic speech recognition; text-to-speech conversion for information-on-demand; call switching and conferencing; unified messaging that 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; call centers handling a large number of agents or telephone operators for processing requests for products, services or information; etc.

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

**Configuration Functions:** Functions that alter the configuration of devices.
Convenience Functions: Functions that simplify application writing.

D/81A: 8 port DSP-based voice board that runs SpringWare firmware. Connects via PEB to a standalone telephone network interface board.

D/120: A 12-channel voice board from Dialogic that consists of a SpringBoard-based expansion device and downloaded software. On the PEB bus, the D/120 serves as a resource module to the installed network module.

D/121: A 12-channel voice-store-and-forward product from Dialogic with all the features of the D/120 plus patented call analysis algorithms for outbound applications and multifrequency (MF) tone capability.

D/12x System: A Voice System that uses D/12x boards. See Voice System.

D/121A: A 12-channel voice board from Dialogic with all the features of the D/121 plus additional RAM, increased performance and reliability, and improved downstream compatibility.

D/121B: 12 port DSP-based voice board that runs SpringWare firmware. Connects via PEB to a standalone telephone network interface board.

D/12x: Any model of the Dialogic series of 12-channel voice-store-and-forward expansion boards for the AT-bus architecture. Includes: D/120 and D/121 boards.

D/160SC-LS: 16 port DSP-based voice board that runs SpringWare firmware and has onboard analog loop start telephone interfaces and an SCbus interface.

D/21D, D/41D: 2 and 4 port DSP-based voice boards with onboard analog telephone interface; runs SpringWare downloadable firmware.

D/21E, D/41E: 2 and 4 port DSP-based voice boards with onboard analog telephone interface; runs SpringWare downloadable firmware.

D/2x: A term used to refer to any 2-channel voice-store-and-forward expansion board made by Dialogic.

D/40: A model of 4-channel voice-store-and-forward expansion board by Dialogic with an on-board processor and shared RAM. The D/40 features real-time digitization, compression and playback of audio, DTMF reception, automatic answering, DTMF or rotary pulse dialing, and direct connection to telephone lines.
D/41: A model of the four-channel voice-store-and-forward expansion boards by Dialogic that has all of the features of a D/40 plus patented call analysis algorithms for outbound applications.

D/4x: Any model of the Dialogic series of 4-channel voice-store-and-forward expansion boards for the AT-bus architecture. Includes D/4xD and D/4xE boards.

D/240SC: 24 port DSP-based voice board that runs SpringWare firmware and has an onboard SCbus interface. Connects to a standalone telephone network interface board.

D/240SC-T1: 24 port DSP-based voice board that runs SpringWare firmware and has an onboard digital T-1 telephone interface and an SCbus interface.

D/300SC-E1: 30 port DSP-based voice board that runs SpringWare firmware and has an onboard digital E-1 telephone interface and an SCbus interface.

D/320SC: 30 port DSP-based voice board that runs SpringWare firmware and has an onboard SCbus interface. Connects to a standalone telephone network interface board.

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.

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: Eliminating false signal detection by filtering out rapid signal changes. Any signal change shorter than that specified by the deglitching parameters is ignored.

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). There are 4 device channels on a D/4x, 12 on a D/12x, 16 on a
D/160SC-LS, 24 on a D/240SC or D/240SC-T1, 30 on a D/300SC-E1, and 32 on a D/320SC board.

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 Management Functions: Functions that open and close devices.

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.

DIALOG/HD Series: Dialogic High Density products, including the D/160SC-LS, D/240SC, D/240SC-T1, D/300SC-E1, and D/320SC, provide a powerful set of advanced computer telephony features that developers can use to create cost-efficient, high-density systems.

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

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

downloadable SpringWare firmware: Software features loaded to Dialogic voice hardware. Features include voice recording and playback, enhanced voice coding, tone detection, tone generation, dialing, call progress analysis, voice detection, answering machine detection, speed control, volume control, ADSI support, automatic gain control, and silence detection.

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


DTI: (Digital Telephony Interface) The naming convention used with Dialogic boards such as the DTI/211. This interface is designed to work with the T-1 telephony standard used in North American and Japanese markets. A general term used to refer to any Dialogic digital telephony interface device.
DTI/124: A model of Dialogic’s digital telephony interface device designed for use with the T-1 telephony standard used in North American and Japanese markets. This model connects to D/4x devices.

DTI/211: 24 port standalone telephone network interface for use with voice-only boards; digital T-1 interface.

DTI/212: 24 port standalone telephone network interface for use with voice-only boards; digital E-1 interface.

DTI/2xx: Refer’s to Dialogic’s DTI/211 or DTI/212 digital telephony interface boards.

DTI/xxx: Refers to any of Dialogic’s second-generation digital telephony interface boards.

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

1: 697,1209  2: 697,1336  3:697,1477
4: 770,1209  5: 770,1336  6: 770,1477
7: 852,1209  8: 852,1336  9: 852,1477
0: 941,1336  *: 941,1209  #: 941,1477

E-1: A CEPT digital telephony format devised by the CCITT. A digital transmission channel that carries data at the rate of 2.048 Mbps (DS-1 level).

emulated device: A virtual device whose software interface mimics the interface of a particular physical device, such as a D/4x boards that is emulated by a D/12x or a D/xxxSC board. On a functional level, a D/12x board is perceived by an application as three D/4x boards. See physical device.

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 a Dialogic application program designed to trap and control processing of device-specific events. The rules for creating a DTI/1xx event handler are the same as those for creating a Windows NT signal handler.

Event Management functions: Class of device-independent functions (contained in the Standard Runtime Library) that connect events to

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application-specified event handlers, allowing users to retrieve and handle events that occur on the device. See Standard Runtime Library.

**Extended Attribute functions:** 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 Standard Runtime Library.

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

**flash:** A signal which consists of a momentary on-hook condition used by the Voice hardware to alert a telephone switch. This signal usually initiates a call transfer.

**frequency detection:** A voice driver feature that detects the tri-tone Special Information Tone (SIT) sequences and other single-frequency tones for call progress analysis.

**Global Tone Detection:** A feature that allows the creation and detection of user-defined tone descriptions on a channel by channel basis.

**hook state:** A general term for the current line status of the channel: either on-hook or off-hook. A telephone station is said to be on-hook when the conductor loop between the station and the switch is open and no current is flowing. When the loop is closed and current is flowing the station is off-hook. These terms are derived from the position of the old fashioned telephone set receiver in relation to the mounting hook provided for it.

**hook switch:** The name given to the circuitry which controls on-hook and off-hook state of the Voice device telephone interface.

**I/O Functions:** Functions that transfer data to and from devices.

**I/O:** Input-Output

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

**in-band:** Refers to the use of robbed-bit signaling (T-1 systems only) on the network or PEB. "In-band" refers to the fact that 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.

**in-band signaling:** (1) In an analog telephony circuit, in-band refers to signaling that occupies the same transmission path and frequency band used to transmit
voice tones. (2) In digital telephony, "in-band" means signaling transmitted within an 8-bit voice sample or time slot, as in T-1 "robbed-bit" signaling. (3) On the Dialogic PCM Expansion Bus (PEB), signaling is considered "in-band" only if it occupies the same transmission path and frequency band used to transmit voice data.

**interrupt request level:** A signal sent to the central processing unit (CPU) to temporarily suspend normal processing and transfer control to an interrupt handling routine. Interrupts may be generated by conditions such as completion of an I/O process, detection of hardware failure, power failures, etc.

**IRQ:** Interrupt ReQuest. A signal sent to the CPU to temporarily suspend normal processing and transfer control to an interrupt handling routine. A means of toggling between applications so that your system does not crash.

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

**loop:** The physical circuit between the telephone switch and the D/xxx board.

**loop current:** The current that flows through the circuit from the telephone switch when the Voice device is off-hook.

**loop current detection:** A voice driver feature that returns a connect after detecting a loop current drop.

**loop start:** In an analog environment, an electrical circuit consisting of two wires (or leads) called tip and ring, which are the two conductors of a telephone cable pair. The CO provides voltage (called "talk battery" or just "battery") to power the line. When the circuit is complete, this voltage produces a current called loop current. The circuit provides a method of starting (seizing) a telephone line or trunk by sending a supervisory signal (going off-hook) to the CO.

**LSI/120:** A Dialogic 12-line loop start interface expansion board.

**off-hook:** The state of a telephone station when the conductor loop between the station and the switch is closed and current is flowing. When a telephone handset is lifted from its cradle (or equivalent condition), the telephone line state is said to be off-hook.

**on-hook:** When a telephone handset is returned to its cradle (or equivalent condition), the telephone line state is said to be on-hook.
Glossary

PC: Personal Computer. In this manual, the term refers to an IBM Personal Computer or compatible machine.

PCM Expansion Bus: See PEB.

PEB: PCM Expansion Bus. A Dialogic open platform, digital voice bus for electrically and digitally connecting different voice processing components. Information on the PEB is encoded using the Pulse Code Modulation (PCM) method. Non-Dialogic products using PCM encoding may interface with Dialogic products by using this bus.

PerfectDigit: Dialogic SpringWare DTMF or MF signaling.

PerfectLevel: Dialogic SpringWare Volume control

PerfectPitch: Dialogic SpringWare Speed control

PerfectVoice: Dialogic SpringWare Enhanced voice coding

physical device: A device that is an actual piece of hardware, such as a D/4x board; not an emulated device. See emulated device.

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

polling functions: Voice library functions check the current status of a voice device. Polling functions are also used to examine the number and configuration of devices in the system and to detect when events occur on a device.

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

resource: Functionality (e.g. voice-store-and-forward) that can be assigned to call. Resources are shared when functionality is selectively assigned to a call (usually via a PEB time slot) and may be shared among multiple calls. Resources are dedicated when functionality is fixed to the one call.

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.

robbed-bit signaling: The type of signaling protocol implemented in areas using the T-1 telephony standard. In robbed-bit signaling, signaling information is carried in-band, within the 8-bit voice samples. These bits are later stripped away, or "robbed," to produce the signaling information for each of the 24 time slots.

routing functions: For SCbus, functions that assign analog and digital channels to specific SCbus time slots; these SCbus time slots can then be connected to transmit or listen to other SCbus time slots. For PEB, functions that change the routing of channels to the time slots on the PCM Expansion Bus (PEB).

sampling rate: Frequency with which a digitizer takes measurements of the analog voice signal.

SCbus: Signal Computing Bus. Third generation TDM (Time Division Multiplexed) resource sharing bus that allows information to be transmitted and received among resources over multiple data lines.


Signal Computer System Architecture: SCSA. A Dialogic standard open development platform. An open hardware and software standard that incorporates virtually every other standard in PC-based switching. All signaling is out of band. In addition, SCSA offers time slot bundling and allows for scalability.

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. In signaling insertion, 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.

solicited event: An expected event. It is specified using one of the device library's asynchronous functions. For example, for dx_play(), the solicited event is "play complete."

Special Information Tones: 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 Tone. Detection of a SIT sequence indicates an operator intercept or other problem in completing the call.

**speed and volume control:** Voice software that contains functions and data structures to control the speed and volume of play on a channel. The end user controls the speed or volume of a message by entering a DTMF tone.

**speed and volume modification table:** Each channel on a voice board has a table with twenty entries that allow for a maximum of ten increases and decreases in speed or volume, and one "origin" entry that represents regular speed or volume.

**SpringBoard:** A Dialogic expansion board using digital signal processing to emulate the functions of other products. SpringBoard is a development platform for Dialogic products such as the D/120 and D/121. The SpringBoard-MC is a development platform for Dialogic Micro Channel products such as the D/81-MC.

**SpringBoard functions:** Functions used on SpringBoard devices only.

**SpringWare:** Software algorithms build into the downloadable firmware that provides the voice processing features available on all Dialogic voice boards.

**SRL:** See Standard Runtime Library.

**Standard Attribute functions:** Class of functions that take one input parameter (a valid Dialogic 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 all Dialogic devices are contained in the Dialogic SRL. See Standard Runtime Library.

**Standard Runtime Library:** A Dialogic software resource containing Event-Management and Standard Attribute functions and data structures used by all Dialogic devices, but which return data unique to the device. See the Standard Runtime Library Programmer’s Guide for Windows NT.

**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. See asynchronous function.

**System Release Development Package:** The software and user documentation provided by Dialogic that is required to develop applications.

**T-1:** The digital telephony format used in North America. In T-1, 24 voice conversations are time-division multiplexed into a single digital data stream containing 24 time slots, and signaling data are carried "in-band." Since all available time slots are used for conversations, signaling bits are substituted for voice bits in certain frames. Hardware at the receiving end must use the robbed-bit" technique for extracting signaling information. T-1 carries data at the rate of 1.544 Mbps (DS-1 level).

**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 asynchronous function.

**time slot:** 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 device channel.

**transparent signaling:** The mode in which a network interface device accepts signaling data from a resource device transparently, or without modification. In transparent signaling, outgoing T-1 signaling bits are generated by a PEB or SCbus resource device. In effect the resource device performs signaling to the network.

**Universal Dialogic Diagnostic program:** Software diagnostic routines for testing board-level functions of Dialogic hardware.
**voice processing:** Science of converting human voice into data that can be reconstructed and played back at a later time. Dialogic equipment can place 2-30 ports in one PC slot. They also use common API’s for scalability and the SCbus to connect to a broad range of technologies.

**Voice System:** A combination of expansion boards and software that let you develop and run high-density voice processing applications.

**wink:** In T-1 or E-1 systems, a signaling bit transition from on to off, or off to on, and back again to the original state. In T-1 systems, the wink signal can be transmitted on either the A or B signaling bit. In E-1 systems, the wink signal can be transmitted on either the A, B, C, or D signaling bit. Using either system, the choice of signaling bit and wink polarity (on-off-on or off-on-off hook) is configurable through DTI/2xx board download parameters.
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