



Dialogic® Brooktrout® Fax Products SDK

Developer Guide

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Preface

Introduction

The *Dialogic® Brooktrout® Fax Products SDK Developer Guide* describes the Bfv API used to create applications to control the features of the Dialogic® Brooktrout® TR1034 Fax Boards, Dialogic® Brooktrout® TruFax® Fax Boards, and the Dialogic® Brooktrout® SR140 Fax Software. The manual gives information about Call Transfer, IP functionality, and BSMI functionality. Finally, it explains how one can include and package software supporting Brooktrout® Fax Boards or Dialogic® Brooktrout® SR140 Fax Software in your product.

The manual contains the following chapters:

- Chapters 1 through 4 provide information about:
 - ◆ The structure of the BFv API
 - ◆ How to develop applications using the Bfv API
 - ◆ Debugging
 - ◆ Sample applications
- Chapters 5 through 9 describe some advanced topics such as:
 - ◆ Call Transfer
 - ◆ Internet Fax Sessions
 - ◆ Robbed bit signaling
 - ◆ ISDN protocols
 - ◆ R2 signaling
- Chapter 10 describes how to package software supporting Brooktrout software or SR140 Fax in your product.

-
- Appendix A provides instructions for a set of legacy G3 utility programs.
 - Appendix B provides instructions for recompiling the driver to support new kernel patches.
 - A glossary gives definitions for some of the terms used in the manual.

A copy of this manual in Adobe Acrobat PDF format is installed in the *Documents* directory on the Dialogic® Brooktrout® TR1034/SR140/TruFax® SDK CD-ROM.

Related Documents

The *Dialogic® Brooktrout® Bfv API Reference Manual* is made up of six volumes that contain the Bfv API function libraries, including the Bfv API, BSMI API and messages.

The installation and configuration guide that came with your software explains how to install the software (firmware, Bfv API, and driver for the TR1034/SR140/TruFax® products) on your host system. It also describes how to configure the driver, configure call control, and download the firmware to a board.

For product information, white papers, FAQs, and more, access the Dialogic web site at www.dialogic.com.

Operating System Support

See the latest *Dialogic® Brooktrout® Release Notes* that came with your SDK for the supported operating systems and their versions. The *Release Notes* are located in the *Documents* directory on the Dialogic® Brooktrout® TR1034/SR140/TruFax® SDK CD-ROM.

Manual Conventions

This manual uses the following conventions:

- *Italics* denote the names of variables in the prototype of a function and file names, directory names, and program names within the general text.

-
- The **Courier** font in bold indicates a command sequence entered by the user at the system prompt, for example:

```
cd /Brooktrout/Boston/bfv.api
```

- The Courier font not bolded indicates system output, for example:

```
C:>Files installed.
```

- The Courier font also denotes programming code, such as C, C++, Microsoft Visual Basic, and TSL. Programming code appears in program examples.
- *Bold* indicates the data type of the prototype of a function, Bfv API functions, dialog boxes, dialog box controls, windows, and menu items.
- Square brackets [] indicate that the information to be typed is optional.
- Angle brackets < > indicate that you must supply a value with the parameter.



The Caution icon is used to indicate an action that could cause harm to the software or hardware.



The Warning icon is used to indicate an action that could cause harm to the user.

Terminology

Updated Terminology

The current version of this document includes terminology that differs from previous versions. Please note the changes below:

Former Terminology	Replaced with...
Host-based fax	Dialogic® Brooktrout® SR140 Fax Software or Brooktrout SR140 Fax Software or SR140 Software or SR140
Virtual modules	
Virtual boards	
Software modules	
VoIP modules	
SR140 virtual modules	
TR1000 Series SDK	Dialogic® Brooktrout® SDK
TR1000 Series Product	Dialogic® Brooktrout® Fax Board or Brooktrout fax board or board
TR1000 Series Module	
TR1000 Series Board	
Brooktrout System Software	Dialogic® Brooktrout® Runtime Software

Dialogic® Brooktrout® TR1034 Fax Board Terminology

The Dialogic® Brooktrout® TR1034 Fax Board is also referred to herein by one or more of the following terms, or like terms including “TR1034”:

- Brooktrout TR1034 Fax Board
- Brooktrout TR1034 Board
- TR1034 Fax Board
- TR1034 Board

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To obtain technical support, please use the web site below:

[*www.dialogic.com/support*](http://www.dialogic.com/support)

1 - Introduction to the Dialogic® Brooktrout® Bfv API

This chapter describes the Dialogic® Brooktrout® Bfv API and its capabilities.

The chapter has the following sections:

- [*Bfv API and Associated Libraries on page 21*](#)
- [*The Bfv API Functions on page 24*](#)

Bfv API and Associated Libraries

The Bfv Application Programming Interface (API) provides a set of functions that enables applications programmers to write applications that run on the Dialogic® Brooktrout® SR140 Software and Dialogic® Brooktrout® TR1034/TruFax telecommunications products. Using the Bfv API, you can generate sophisticated, multichannel voice and fax applications under Linux and Windows operating systems.

The Bfv API comprises of several libraries that work together to give flexibility in a variety of applications such as:

- Voice processing (Dialogic® Brooktrout® TR1034 Fax Boards only) and signal generation and detection
- Fax
- Connection to a variety of telephony interfaces:
 - ◆ T1 robbed bit
 - ◆ T1/E1 PRI
 - ◆ R2 CAS signaling
 - ◆ Analog
 - ◆ BRI
 - ◆ QSIG
 - ◆ IP (SIP and H.323)

The libraries include:

- Bfv API
 - Provides telephone line administration and initialization; board, firmware, call control, and packet network configuration and control; debugging and error handling, high-level call control for analog, T1 robbed bit signaling, BRI, QSIG, and T1/E1 PRI; voice play and record; signal generation and detection; fax manipulation from high to low level; and file formatting for voice and fax messages.
- Boston Simple Message Interface (BSMI)
 - Provides very low-level call control for T1/E1 ISDN and R2 signaling. The Bfv API uses the BSMI library to handle the T1/E1 call control, but the higher-level call control functions manage BSMI for you.

By using the Bfv API libraries, the application running on the host processor can communicate through the driver and firmware to one or more Dialogic® Brooktrout® boards.

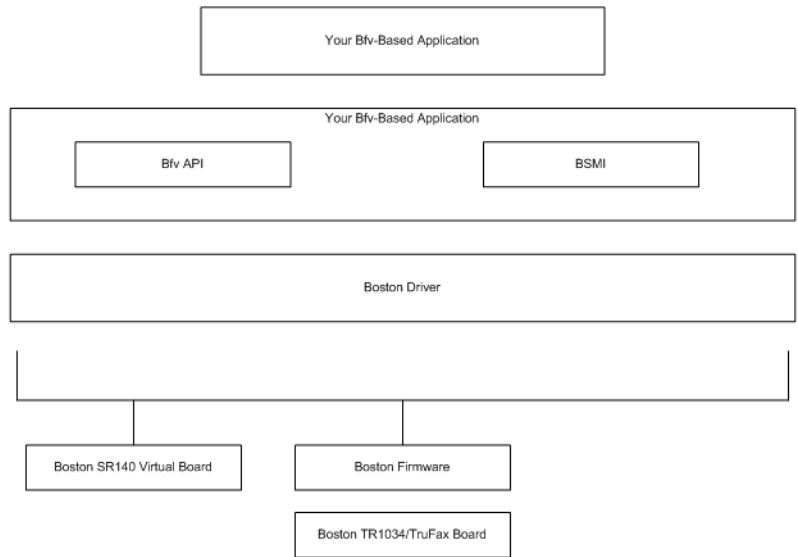


Figure 1. Bfv Application Configuration

The TR1034 and TruFax® have an assigned module number as indicated on the rotary switch on top of the board, so you can have control over channels on individual boards in a multi-board system. The Dialogic® Brooktrout® SR140 uses a module numbers starting at 0x41 with up to a maximum of 120 channels per SR140 module.

The SR140 does the following:

- Works on supported Linux and Windows platforms running on Linux and Windows platforms.
- Works with SIP and H.323 IP call control.
- Operates with only one IP stack at a time.
- Provides the same level of debugging and tracing that is available on the TR1034 platform.
- After configuration and licensing, the same application supports the SR140 and your hardware based T.38 solution.

Media on a Boston modules is driven by the Bfv API. Call control on the TR1034/TruFax Boston module's can be driven by BSMI or the Bfv API. SR140 Boston module can only be used with the Bfv API.

The Bfv API libraries are based on the BTLINE structure, which is a logical abstraction of a physical channel. Each active channel stores its information within its own BTLINE structure. You can access and modify the BTLINE information through the Bfv API functions. You can access other information kept in the Bfv API library, using macros found in *btlib.h*.

The Bfv API Functions

The Bfv API functions in all the Bfv API libraries are separated into categories according to the tasks they perform. They are:

- Configuration, Administration, and Management
 - ◆ Administration and Initialization
 - ◆ Firmware
 - ◆ Configuration
 - ◆ Debugging, Error Handling, and Return Values
 - ◆ Miscellaneous (for example, *_dll_* and *getopt*)
- Call Control
 - ◆ High Level Call Control
 - ◆ Low Level Call Control
- Media Processing
 - ◆ Signal Generation and Detection
 - ◆ Voice Play and Record
 - ◆ Fax
 - ◆ File Format Manipulation

In addition to the functions, Dialogic supplies macros to provide information or perform a specific task.

Administration, Management, and Configuration

Administration and Initialization Functions and Macros

The administration and initialization functions allow you to:

- Attach and detach from a line or a session.
- Configure the module instead of using a user-defined configuration file such as *btcall.cfg*.
- Interrupt a thread or process on an active line.
- Reset the specified channel.
- Get information about the module and channel address for the specified channel.
- Get the number of available channels.

You can also use specialized functions to:

- Check for an address or facility.
- Download and query the feature set.
- Get information about and deactivate a board or SR140.
- Receive a packet containing Boston addresses and commands and perform internal Bfv API processing on all commands with the packet (requires the Boston command set).

The line administration and initialization macros allow you to:

- Get information about the current version of the Bfv API or driver and some information about the operating system environment.
- Clear an item.
- Get information about a line.

The BTLINE Structure

When an application calls the `BfvLineAttach` (or `BfvSessionAttach`) function to open and attach to a specified channel, the function creates a separate BTLINE structure for the channel and returns a pointer to the line structure. All information about the channel is stored in its BTLINE structure, but only the line state, the line type, and channel number are actually relevant. The *BfvLineDetach* (or *BfvSessionDetach*) function deallocates a BTLINE structure.

Applications do not directly access the internal fields of the BTLINE structure, but instead use functions and macros described in detail in the *Dialogic® Brooktrout® Bfv API Reference Manual*:

`LINE_HAS_CAP(lp, cap)`

Confirms whether or not the line has the specified capability *cap*.

`LINE_STATE(lp)`

Returns or sets the line state of the specified line.

`LINE_TYPE(lp)`

Returns the line type of the specified line.

`LINE_UNIT_NUM(lp)`

Returns the channel number of the specified line.

A line is always in one of the following states:

`LINE_STATE_AWAIT_TRAINING`

`LINE_STATE_CONNECTED`

`LINE_STATE_FAX_MODE`

`LINE_STATE_HOLDUP`

`LINE_STATE_IDLE`

`LINE_STATE_NOLOOP`

`LINE_STATE_OFF_HOOK`

`LINE_STATE_RCV_INFO`

`LINE_STATE_RESETTING`

`LINE_STATE_RETAIN`

`LINE_STATE_RINGING`

`LINE_STATE_TURNAROUND`

Hereafter each of the line states is referred to by the descriptive part of its name only (for example, `LINE_STATE_IDLE` is referred to as `IDLE`).

The current state of the line is stored in the `BTLINE` structure. A pointer to this structure is passed as an argument to nearly all Bfv API entry points and is provided to the application by the *BfvLineAttach* function.

A number of functions and interrupts serve as inputs to the `BTLINE` structure and affect the transition to different line states. Other functions check the current line state.

Some functions conditionally branch to other points in the code, and some prevent inappropriate action from occurring, for example, frequent checking for `CONNECTED` before attempting to transmit data.

The following is a partial list of the functions and interrupts and the line state they set:

FUNCTION	LINE STATE
<i>BfvLineAnswer</i>	Sets the state to <code>CONNECTED</code> .
<i>BfvLineAttach</i>	Initializes the state to <code>IDLE</code> .
<i>BfvLineOriginateCall</i>	Sets the state to <code>CONNECTED</code> or <code>OFF_HOOK</code> depending on the results from call progress monitoring.
<i>BfvLineReset</i>	Resets the state to <code>IDLE</code> .
<i>BfvLineTerminateCall</i>	Sets the state to <code>IDLE</code> .

INTERRUPTS	LINE STATE
Answer tone detect	Sets the state to <code>CONNECTED</code> .
Direction change	Sets the state to <code>TURNAROUND</code> .
Disconnect	Sets the state to <code>IDLE</code> .
Received FSK data	Sets the state to <code>AWAIT_TRAINING</code> .
Ring detect	Sets the state to <code>RINGING</code> .
Training	Sets the state to <code>FAX_MODE</code> .

Channel Numbering

The Bfv API uses two numbering schemes when referencing channels within a system. One is the *unit number* or *ordinal channel number*; the other is the *logical channel number*.

The *unit number* is a number range 0...n-1, where n is the number of channels in the system. The *BfvLineAttach* function uses the unit number in its argument and returns a pointer to the BTLINE structure, providing a means to reference the channel in future function calls. For example, a system comprising two 60-channel modules would have a unit number range of 0-119. The module that had the firmware downloaded first would contain the channels starting from 0.

The *logical channel number* is used together with the module number to reference a *work channel* (also called a *hardware channel*) in a system. The *BfvSessionAttach* function uses the module and logical channel numbers in its arguments. Logical channels not only include work channels traditionally considered to be channels, but also *administrative channels*. The work channel number range for logical channels is 2...n+1, where n is the number of work channels on this hardware module.

Each module has a unique module number. For example, the same system comprising two 60-channel modules could have the following configuration:

- First module: Module 2, work channels 2-61
- Second module: Module 3, work channels 2-61

Each virtual module can have up to 120 channels.

The *BfvSessionAttach* function also returns a BTLINE structure; other functions that accept a BTLINE structure as an argument can use either that returned from *BfvLineAttach* or *BfvSessionAttach*. When detaching, use the corresponding detach functions *BfvLineDetach* or *BfvSessionDetach*.

Firmware Functions and Macros

With the specialized firmware functions, you can:

- Download firmware to the module from a file or a buffer
- Get information about a module's firmware configuration options

With the firmware macros, you can determine:

- Version number, build number, and date of the control processor firmware
- Version number, build number, and date of the boot ROM firmware
- Version number, build number, date of each DSP firmware, and the number of DSPs on the module

Configuration Functions

The Bfv API provides functions that allow you to get the current information about the telephony configuration, reset the telephony state, and save telephony parameters to Non-Volatile RAM (NVRAM).

You can also establish a connection between source and destination telephony resources; get information about the connections, their ports and classes; and clear all switching connections for a module.

Configuration Files

The Bfv API uses several configuration files that let you configure the Bfv API and driver, call control, and country-specific parameters. These files are described below. Sample versions of the files are stored in the directory */Brooktrout/Boston/config*.

- The user-defined configuration file
A file that contains configuration parameters for the Bfv API and driver. A sample of this file, called *btcall.cfg*, is provided with the software, but you can write your own or modify/rename the existing one. Many of the sample applications (see [Sample Applications and Utilities](#) on [page 120](#)) use *btcall.cfg*.
- The call control configuration file
A user-supplied file that contains call control configuration parameters. Several samples of this file are provided with the software. One sample is called *callctrl.cfg*, while others have names that specify the type of telephony interface. See the directory */Brooktrout/Boston/config/samples.cfg* for the names of the files.
- The telephony configuration file
This file is obsolete and has been superseded by the call control configuration file.
- The *BT_CPARM.CFG* file.
A read-only file that contains country-specific parameters.

Module Status and Monitoring Functions

With the module status and monitoring functions, you can:

- Set and get the state of the module by reading the status LED.
- Set the module temperature threshold.
- Get the temperature of the module.
- Have the module perform a series of self tests and, optionally report the results.
- Have the module notify the application of events or conditions on the module such as a network alarm, network error, H.100/H.110 clock event, temperature alarm, RTP/RTCP transport layer events, and the general status of the module.

Debugging, Error Handling, and Return Values

Functions

Several Bfv API functions help you debug your application program and discover/recover from errors.

You can enable debug mode so that the Bfv API prints commands, data, interrupts, and status messages, or you can set up a function to be used with Bfv API debug mode that directs output to a file or filter (see [Debugging](#) on [page 90](#)).

When you install the Bfv API, you enable recording of the history of the activity of the driver along with the hardware type, the firmware version, and the boot ROM version. You can then use functions to dump the buffer containing the driver's history for a module and channel to a file. You can also clear the history buffer for a module and channel so that it contains information relevant to the current application.

If you have a RES structure that contains returned error information from a previous Bfv API call, you can use the *BfvErrorMessage* function to create a short and a long error message in a BTERR structure. Then, you can print either the long or short message from the structure.

When you start call control using *BfvCallCtrlInit*, you can enable a call control log file.

Structures and Return Values

The Bfv API uses argument structures to pass values to and from functions. The application declares the argument structure and passes a pointer to it to the function. The argument structure type is named *args_...*; for example, *struct args_speech*. The same argument structure type is used for functions that are related or in the same category.

Contained within the argument structure are structure fields that are used for input and/or output. Each function that uses an argument structure has marked the fields that are used for each purpose. Not all fields are used by all functions taking any particular argument structure type.

Result structures are the most commonly used structures to return information to the function. They are:

RES

Returns status information in *res.status* and some additional information in *res.line_status*.

CALL_RES

Returns information about a call such as its type and destination. If applicable, information such as called party and redirect information are returned as well.

Miscellaneous Functions and Macros

Some administration functions and macros cannot be classified with other functions, but are useful in various ways. For example:

- `_dll...` functions for use on Windows operating systems. These functions call standard C library functions such as *fopen*, *fclose*, *fread*, and *fwrite*; their arguments use the runtime library linked with the DLL.
- The *getopt* function parses command line options in a Linux environment. Most of the sample applications/utilities use this function (see [Sample Applications and Utilities](#) on [page 120](#)).
- The *BfvMemAllocFuncSet* function allows you to write your own functions to dynamically allocate and free memory instead of using the Bfv API functions to do so.
- The `sleep` macro lets you write applications that sleep for a defined period of time (in seconds). This macro is only defined for environments that do not have built-in sleep functions.

Call Control

Call control functions enable the application to set up, initiate, connect, disconnect, and perform other tasks related to the telephone network. Three forms of call control are available: Bfv high-level, low-level and BSMI-level.

Bfv Call Control

High-level Bfv call control functions simplify the process of accessing the telephone system. Some of the high-level functions call the low-level Bfv call control functions to automatically perform the low-level tasks. However, the high-level functions trade flexibility and control for ease of programming.

With the Bfv low-level call control functions, you can perform T1 robbed bit, T1/E1 PRI, T1/E1 QSIG, E1 R2, analog, BRI, SIP, and H.323 call control functionality. See *Volume 2* of the *Dialogic® Brooktrout® Bfv API Reference Manual* for the descriptions of the Bfv call control functions.

BSMI-Level Call Control

The BSMI-level call control functions are used to facilitate communications directly between the module and the telephony lines. These are the Bfv API Boston Simple Message Interface (BSMI) functions that use messages to communicate between the module and the telephone lines. The collection of messages is the interface to the telephony component of the Boston firmware and provides all the facilities for management, call control, and performance statistics monitoring. Control message naming conventions in the BSMI are descriptive of the functions they serve and make it easier to develop applications. When developing an application, you do not need to have a detailed knowledge of the protocol involved, although a general understanding of call models is beneficial. You can use one of many different signaling protocols such as T1/E1 PRI; R2; and Local Exchange Carriers (LEC) T1 Robbed Bit, Analog and BRI. See [Robbed Bit Signaling](#) on [page 266](#) - [Using the BSMI R2 Signaling Capability](#) on [page 335](#) for more information about the protocols.

Note: BSMI is not supported on QSIG, SIP, and H.323.

Typically, the BSMI is used as one component of a system. Firmware download, for example, is achieved using the call control functions of the Bfv API. Through the Bfv API, you can perform all appropriate configuration and management functions for the Brooktrout products.

BSMI is used by the Bfv call control functions to perform call processing. BSMI is a level lower than the Bfv API, providing greater flexibility.

The host communicates with the module through the Control Interface. The host application (referenced as L4) issues BSMI control messages to configure the module or to instruct it to perform a specific action, such as make a call, clear a call, or request the status of an interface. The module-issued BSMI control messages (referenced as L3) inform the host of the status of the interface, call events, or an error condition.

BSMI supports the R2 signaling protocol. Using the BSMI host to module messages, you can:

- Start and stop the R2 protocol on a particular timeslot on an E1 span.
- Block or unblock an idle B-channel (the ISDN channel that handles data).
- Place an outbound call.
- Answer an inbound call.
- Disconnect a call.
- Reject an incoming call.

BSMI module to host messages respond to the host by:

- Starting and stopping the R2 protocol.
- Blocking or unblocking the B-channel.
- Seizing the line for an incoming call.
- Alerting the host and then connecting a call.
- Clearing a request.
- Notifying the host when the remote end phone is ringing.
- Notifying the host when the call is disconnected at the remote end.
- Providing a protocol error or invalid command status message.

Media Processing

Media processing refers to the operation that is performed on the modules. Depending on the product configuration, it can include:

- Signal generation and detection
- Voice play and record
- Faxing
- File format manipulation

Signal Generation and Tone Detection

With the signal generation and tone detection functions, you can:

- Play call progress signals and generate other tone groups and tone patterns.
- Get the next call progress code.
- Enable and disable DTMF detection.
- Discard tones from a buffer.
- Wait for a tone and return it as an ASCII character or return it without disturbing the buffer.
- Play a tone for a specified time.
- Play a single frequency tone or a custom tone.
- Replace a tone in the buffer for reuse.

The module receives call progress signals generated by telcos and Private Branch Exchanges (PBXs) before, during, and after dialing. The module's call progress analysis processes then interprets them.

During call progress analysis, modules can report dial tone detection, ring-back, busy signals, remote fax tone detection, and other important information. Applications can use this information to determine their next course of action, to display the status of a call, or to track billing information. Applications can use postdialing results, such as `HUMAN` and `BUSY`, to decide what redialing strategy to use.

Modules can also generate and play DTMF and MF tone groups and single tone patterns to send to the telco or PBX.

Voice Record and Play

With the Bfv voice record and play functions, the application can:

- Open, play, and close a previously recorded prompt file.
- Record speech into an infopkt stream, a raw speech data buffer, a raw speech file, or a wave file.
- Play back speech from an infopkt stream, a raw speech data buffer, a raw speech file, or a wave file.
- Modify the volume and rate of a speech playback while it is in progress.

The voice functions allow you to write Interactive Voice Recognition systems where you can record prompts for later playback. You can also build voice mail systems for recording and playing back messages.

Fax Functions

The Bfv API provides a wealth of fax functions that allow you to control every aspect of sending and receiving V.17 or V.34 faxes.

The fax functions are divided into high-, mid-, and low-level functions. *Volume 4, Fax Processing, in the Dialogic® Brooktrout® Bfv API Reference Manual* provides a detailed description of each Bfv function.

Generally, the high-level functions simplify the process of transmitting and receiving facsimiles. Since the high-level functions incorporate many of the appropriate low-level functions to automatically perform the basic low-level tasks, applications using the high-level functions are freed to perform other tasks. For example, the high-level function *BfvFaxSend* is constructed of these mid- and low-level functions:

BfvFaxBeginSend

BfvFaxEndOfDocument

BfvFaxGetRemoteInfo

BfvFaxSendPage

BfvFaxSetLocalId

BfvFaxWaitForTraining

The high-level functions trade the maximum flexibility and control provided by the low-level functions for ease of programming.

The mid-level functions provide more flexibility and control than the corresponding high-level functions, but they require more knowledge of and attention to the basic steps involved in sending and receiving facsimiles.

The low-level functions provide the greatest flexibility and control over sending and receiving facsimiles, but they require extensive knowledge of and attention to the basic steps involved in each of these tasks. For example, applications can screen phone calls based on an ID string or NSF information with the low-level functions, but not with the high-level functions.

Both the high- and mid-level functions use only infopkt files, so the distinction between them is measured in the flexibility and control they provide. The low-level functions, however, use only raw data files.

Combining the high-, mid-, and low-level functions within the same application program is valid and useful. Need for the low-level calls depends on the degree of flexibility and functionality an application requires.

[Table 1](#) contains a partial list of the high-, mid-, and low-level functions that perform fax tasks.

Table 1. Fax Functions by Type

Type/Level	Function Names
High-Level	<i>BfvFaxPoll</i> <i>BfvFaxReceive</i> <i>BfvFaxSend</i>

Table 1. Fax Functions by Type (Continued)

Type/Level	Function Names
Mid-Level	<i>BfvFaxBegin</i> <i>BfvFaxBeginReceive</i> <i>BfvFaxBeginSend</i> <i>BfvFaxEndReception</i> <i>BfvFaxNextPage</i> <i>BfvFaxReceivePage</i> <i>BfvFaxReceivePages</i> <i>BfvFaxSendPage</i> <i>BfvFaxSetReceiveFmt</i> <i>BfvFaxTurnaround</i>
Low-Level	<i>BfvFaxBeginRaw</i> <i>BfvFaxBeginSendRaw</i> <i>BfvFaxEndOfDocument</i> <i>BfvFaxGetLocalId</i> <i>BfvFaxGetRemoteInfo</i> <i>BfvFaxNextPageRaw</i> <i>BfvFaxPageParams</i> <i>BfvFaxReceiveData</i> <i>BfvFaxReceiveFile</i> <i>BfvFaxSendData</i> <i>BfvFaxSendFile</i> <i>BfvFaxStripParams</i> <i>BfvFaxWaitForTraining</i>

In addition, the fax functions are divided into two subgroups: those that process infopkt-formatted data files and those that process ASCII or G3 data files in other formats. For fax functions that process raw ASCII or G3 data files rather than infopkt-formatted data files, see *Volume 4, Fax Processing*, in the *Dialogic® Brooktrout® Bfv API Reference Manual*.

File Format Manipulation Functions

Using the file format function calls, you can perform the following infopkt operations:

- Open and close infopkt stream files
- Find the pointer position in an infopkt stream file

-
- Look for an offset to a specific place in an infopkt file
 - Read from and write to an infopkt stream file or buffer
 - Put the last infopkt read back into the infopkt stream file
 - Create your own function to handle user-defined infopkt files

Infopkts

An infopkt is a structure, consisting of a tag and associated data, that organizes different data types (ASCII, voice, and fax data) into a single structure for transmission or reception.

The Bfv functions can process voice and fax data that is stored in files containing infopkt structures. Infopkt structures contain speech or fax formatting and control parameters, speech or fax data, or pointers to other data or infopkt files. These structures provide a flexible and easily extendable method to combine and transmit various types of data.

For voice applications, infopkts provide an easy means to build sophisticated interactive voice systems. Using infopkts, a voice application can create a master prompt file that builds all of the system's prompts out of short phrases.

This scheme:

- Reduces the amount of disk space needed for storage.
- Enables the application to build new prompts as changing demands on the system dictate.
- Simplifies the development of multilingual systems that can translate recorded prompts to other languages on-the-fly.

For fax applications, infopkts provide a flexible means to transmit a complex, computer-generated fax document.

A fax transmission consists of one or more documents. A document consists of one or more pages containing the document parameters, agreed upon by two communicating fax machines according to the T.30 protocol, that do not change. A page consists of one or more strips of data, converted from their original format to the established document format for transmission.

To a fax machine, a document is a simple object with three parameters: vertical resolution, length, and width. To a computer, a document is often more complex.

For example, fax applications might require the transmission of ASCII data in fax format. The fax module converts the ASCII data to G3 format in real-time. A file header and signature, already in G3 fax format, can accompany the ASCII data. The infopkt structure makes it easy to send a document of this type because it organizes fax and ASCII data into a single structure for transmission and reception.

The Bfv API defines and supports the infopkt types described on the following pages:

- *Tag Infopkts* on [page 41](#)
- *Data Infopkts* on [page 43](#)
- *Indirect Infopkts* on [page 43](#)
- *User-Defined Infopkts* on [page 44](#)

Tag Infopkts

Contain speech parameter structures (which describe the sample rate, coding format, and data format of the speech or indicate the end of speech playback), and fax parameter structures (which describe a strip or page of data, the line parameters, or control parameters). They are:

INFOPKT_ASCII_STRIP_PARAMETERS

Tag containing parameters for ASCII data strip.

INFOPKT_BEGINNING_OF_PAGE

Tag indicating the beginning of a new page with no new parameters from the previous page.

INFOPKT_DOCUMENT_PARAMETERS

Tag indicating new page composition parameters. An infopkt stream must begin with this infopkt type. This is used, for example, to change the resolution between pages in the middle of a fax transmission.

INFOPKT_EFF_PAGE_PARAMETERS

Tag containing enhanced fax format page parameters.

INFOPKT_END_OF_SPEECH

Tag indicating end of speech playback.

INFOPKT_FAX_HEADER

Tag specifying a header or footer to appear on all subsequent pages of a fax transmission.

INFOPKT_G3_STRIP_PARAMETERS

Tag containing parameters for G3 data strip.

INFOPKT_PAGE_PARAMETERS

Tag containing margin and padding parameters for a page.

INFOPKT_SPACE

Tag containing the spacing parameters for a fax page or a fax overlay.

INFOPKT_SPEECH_PARAMETERS

Tag containing parameters for succeeding speech infopkts.

INFOPKT_T30_PARAMETERS

Tag containing T.30 line parameters bit rate and scan time. When ECM is in use, the scan time specification has no effect.

Data Infopkts

Contain just the header and data, permitting applications to organize large files as a sequence of small data infopkts. They are:

INFOPKT_ASCII

ASCII data.

INFOPKT_G3

G3 data.

INFOPKT_PROMPT_MAP

Used only in prompt files. Contains information on how to find each of the phrases in the prompt file.

INFOPKT_SPEECH

Speech data in any of several coding formats.

Indirect Infopkts

Contain a pointer to a file that contains either raw data (ASCII, speech, or G3) or infopkts. They are:

INFOPKT_INDIR_ASCII

Pointer to an ASCII file.

INFOPKT_INDIR_DCX

Pointer to an Intel DCX fax file that contains PCX data and its own header information.

INFOPKT_INDIR_G3

Filename of a G3 file.

INFOPKT_INDIR_INFOPKT

Pointer to another infopkt stream file.

INFOPKT_INDIR_SPEECH

Pointer to a raw speech file.

INFOPKT_INDIR_TIFF

Pointer to a TIFF-F file that contains G3 data and its own header information.

INFOPKT_INDIR_WAVE

Pointer to a wave file that contains speech data and its own header information.

User-Defined Infopkts

Contain a header (the document's title, the subject of the document, or the total number of pages that the document contains) and user-defined information (document summary and statistics, etc.) useful to an application. When the Bfv API encounters these infopkt types, it ignores them. See the *BfvInfopktUser* function in *Volume 4, Fax Processing*, in the *Dialogic® Brooktrout® Bfv API Reference Manual*. They are:

INFOPKT_USER0_USER1 . . . _USER9

Infopkt containing a header and storage for user application information. A maximum of ten user-defined infopkt types (numbered 0 to 9) are included in an infopkt stream.

INFOPKT_ANNOTATION

Contains header and ASCII text, but is otherwise identical to the user types described above.

The Infopkt Stream

An infopkt stream is a file or memory buffer containing concatenated individual infopkts. The length of an infopkt stream is limited only by the file size conventions specific to an operating system. The *BfvInfopktOpen* function opens file-based infopkt streams, and the *BfvInfopktOpenMem* function opens memory-based infopkt streams.

For speech record and play applications, the first infopkt in the infopkt stream depends on the type of speech file.

If the speech file is an indexed prompt file (see [mkprompt](#) on [page 148](#) in this manual, and the *BfvPromptPlay* function in *Volume 3 of the Dialogic® Brooktrout® Bfv API Reference Manual*), an INFOPKT_PROMPT_MAP infopkt begins the infopkt stream. An INFOPKT_SPEECH_PARAMETERS infopkt begins each new speech file embedded within the infopkt stream.

If the speech file is a simple speech file, one recorded with the *BfvSpeechRecord* function, an INFOPKT_SPEECH_PARAMETERS infopkt begins the infopkt stream.

Creating an Infopkt Stream

The *mkinfopk* program, included on the distribution media, builds an infopkt stream. It has the following command syntax:

mkinfopk -o output_fname {infopkt_type arg}...

-o output_fname	Is the name of the output file.
infopkt_type	Indicates the type of infopkt which follows.
arg	Is the argument for the infopkt. Depending on the infopkt_type, it is either a file name or a dummy value.

For more detailed information on *mkinfopk*, see [mkinfopk](#) on [page 146](#) in this manual.

The *decode* program, also included on the distribution media, reads a specified infopkt stream and lists the individual infopkts within the stream. See [Sample Applications and Utilities](#) on [page 120](#) in this volume for more detailed information on how to use this utility.

Figure 2 illustrates how to create an infopkt stream to play either an indexed prompt file or a simple speech file. You must create a separate infopkt stream for each type.

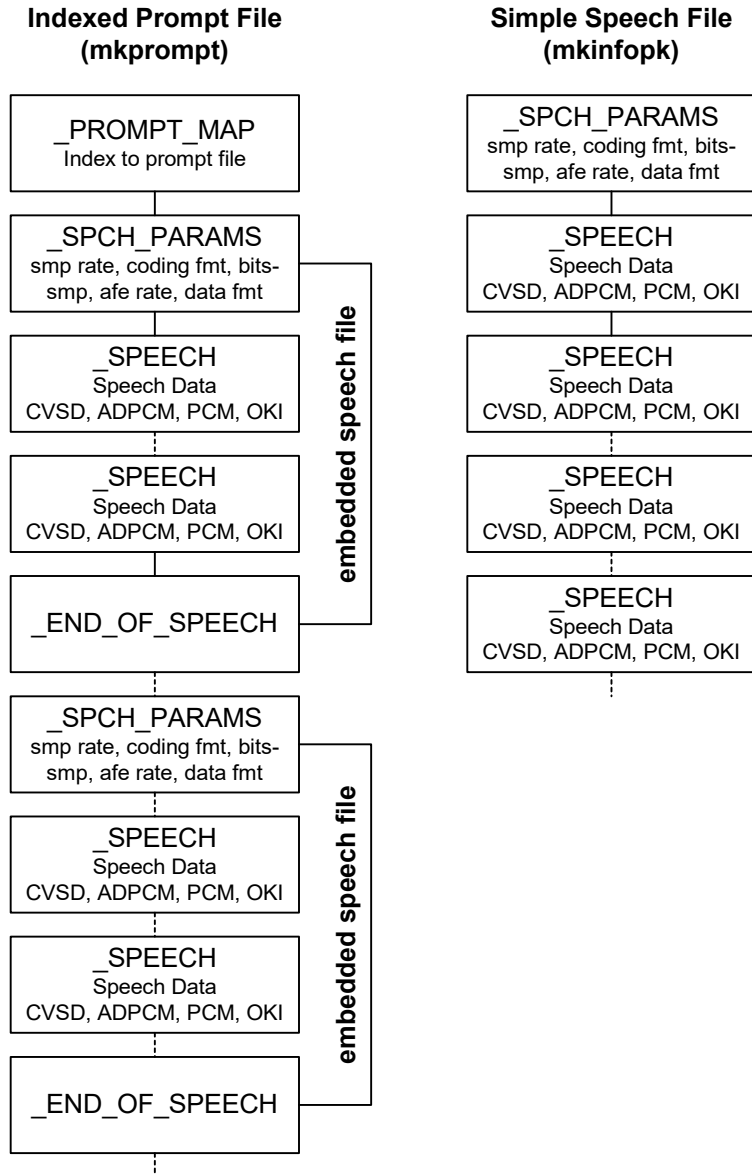


Figure 2. Flow Chart for Creating Infopkt Streams that Play Speech

The following sample shows how to create an infopkt stream file to test the fax functionality of your hardware and software:

```
mkinfoPk -o fax.ips doc 1 ascii fax.c
```

Where:

<code>fax.ips</code>	Is the name of the output infopkt stream file that contains the ASCII file <i>fax.c</i> (the sample fax application program included on your distribution CD).
<code>doc</code>	Is required as the first infopkt in a stream (1 is its argument). See mkinfoPk on page 146 for more information about <code>doc</code> .
<code>ascii</code>	Indicates that the input file <i>fax.c</i> is an ASCII formatted file.
<code>fax.c</code>	Is the input file.

For fax-receiving applications, an `INFOPKT_DOCUMENT_PARAMETERS` infopkt begins each new G3 page within it. If the application uses a nonstandard receive format (see the *BfvFaxSetReceiveFmt* function in *Volume 4, Fax Processing*, in the *Dialogic® Brooktrout® Bfv API Reference Manual*), it must also include a `G3_STRIP_PARAMETERS` infopkt.

For fax-transmitting applications, an infopkt stream must begin with an `INFOPKT_DOCUMENT_PARAMETERS` infopkt. Any type of infopkt or combination of infopkts can follow the first `INFOPKT_DOCUMENT_PARAMETERS` infopkt.

Within a fax-transmitting application, a new page is indicated when one of the following infopkt types is encountered in an infopkt stream:

```
INFOPKT_BEGINNING_OF_PAGE
INFOPKT_DOCUMENT_PARAMETERS
INFOPKT_T30_PARAMETERS
INFOPKT_FAX_HDR
```

[Figure 3](#) on [page 48](#) illustrates how to create an infopkt stream that transmits G3 data. G3 data includes MH, MR, MMR, and PCX formats.

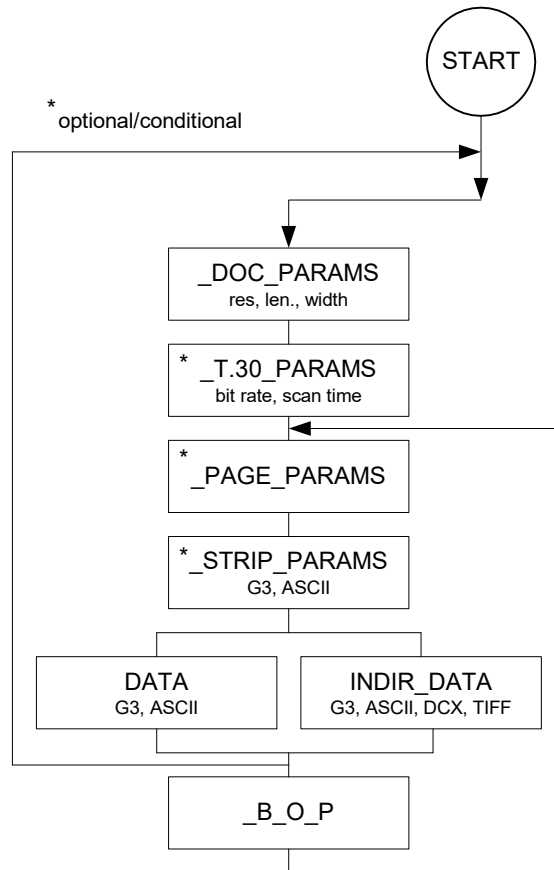


Figure 3. Creating Infopkt Streams that Transmit Facsimiles

Infopkt Structure

Every infopkt consists of a header and data. The 4-byte header consists of a type code and a length. The type code defines the infopkt type, and the length field indicates the total length of the infopkt, including the header. Data consists of raw data, formatting parameters or, in the case of indirect infopkts, pointers to other files.

The maximum length of an individual infopkt is 30,000 bytes, but Dialogic recommends limiting the size to approximately 1K. This limit affects the size of infopkts only and has no effect on the size of a fax document, since large fax documents are simply converted to multiple infopkts within an infopkt stream.

When indirect infopkts point to other infopkt stream files, the maximum nesting depth is three. All file names that occur in indirect infopkts must be 0-terminated.

The *infopkt.h* file, located in the *inc* subdirectory, contains the definitions of the infopkt structures.

See *Appendix E* in *Volume 6* of the *Dialogic® Brooktrout® Bfv API Reference Manual* for the parameter values and defaults of each tag type infopkt.

Speech Infopkt Parameters

The `INFOPKT_SPEECH_PARAMETERS` infopkt defines the speech parameters for infopkt-formatted speech files. These parameters include the coding format, the compressed sample rate, the number of bits per sample, the analog front end (afe) rate, and the data format. In infopkt streams made up of simple speech files, this infopkt begins the infopkt stream. In infopkt streams made up of indexed prompt files, it begins each new speech file embedded in the stream.

Applications use the *BfvSpeechRecord* function to record speech in infopkt format (see the *Dialogic® Brooktrout® Bfv API Reference Manual, Volume 3* for a complete description). Valid settings for the data coding format, compressed sample rate for playback, and number of bits per sample are found with the *BfvSpeechRecord* function.

Fax Infopkt Parameters

The T.30 protocol requires two communicating fax machines to agree on several transmission parameters at the beginning of a facsimile transmission. These transmission parameters include the bit rate, scan time, coding scheme, and the basic document format parameters – vertical resolution, page width, and page length. The least capable fax machine determines the values of these parameters; both fax machines adjust to the final values.

Two infopkts:

INFOPKT_T30_PARAMETERS

INFOPKT_DOCUMENT_PARAMETERS

affect the parameters that are negotiated during the T.30 protocol handshaking procedure.

The INFOPKT_T30_PARAMETERS infopkt specifies the desired values of the transmission parameters. Normally the default values are used, but INFOPKT_T30_PARAMETERS is useful, for example, for setting a lower bit rate.

The INFOPKT_DOCUMENT_PARAMETERS infopkt sets the document related parameters: vertical resolution, length, and width (only the vertical resolution is programmable). These parameters format the fax data that is sent out.

INFOPKT_DOCUMENT_PARAMETERS is required and specifies the desired resolution. If it appears in the midst of an infopkt stream, both machines might renegotiate to the new parameter values.

While document parameters define an entire fax document sent, page parameters define an entire page only, and strip parameters define horizontal strips of data within a page.

For ASCII data, there are page parameters and strip parameters. For G3 data, there are strip parameters. Because page formatting elements (top and bottom margins, etc.) are inherent in G3 data; there are no separate page parameters.

The INFOPKT_PAGE_PARAMETERS infopkt defines the ASCII page parameters, which apply only to pages. Its use is optional, and when it is not included with ASCII data infopkts, the module uses the default values (see *Appendix E* in *Volume 6 of the Dialogic® Brooktrout® Bfv API Reference Manual* for default values).

The strip parameters infopkts:

INFOPKT_ASCII_STRIP_PARAMETERS

INFOPKT_G3_STRIP_PARAMETERS

define the actual strips of data that make up a page and must precede an ASCII or G3 data type infopkt only to change the default or previously applied strip parameter values (see *Volume 6, Appendix E* in the *Dialogic® Brooktrout® Bfv API Reference Manual* for default values). Dialogic, however, recommends that you include a strip parameter infopkt whenever you define a strip, even if the default values are appropriate for the strip. Strip parameters include the basic format of the data (that is, ASCII or G3 data) and presentation parameters such as vertical resolution, width, and, in the case of ASCII, left and right margins.

Strip parameters ensure that the strip data is sent out properly and the received fax is displayed with the proper proportions. The module converts strip data to the proper vertical resolution before transmitting it.

For example, if the T.30 document resolution is set to fine resolution, and a G3 strip is in normal resolution, the firmware converts the data to fine resolution, replicating each line. Likewise, if the T.30 document resolution is set to normal resolution, and a G3 strip is in fine resolution, the firmware converts the data to normal resolution, removing every other line. If the T.30 document resolution and a G3 strip resolution are the same, the module transmits the data as-is. The vertical resolution of the strip data informs the module when to convert data and how to convert it.

When ASCII strip data is sent to the module, the current vertical resolution parameter is set to normal, even if an INFOPKT_ASCII_STRIP_PARAMETERS infopkt is inserted into the infopkt stream. Thus the resolution of any G3 strip data, encountered later in the infopkt stream, is also assumed to be normal, unless otherwise specified by an accompanying INFOPKT_G3_STRIP_PARAMETERS infopkt.

When two consecutive G3 strips of data are sent to the module, an INFOPKT_G3_STRIP_PARAMETERS infopkt must be inserted between them. Since each G3 strip data ends with an RTC marker (six consecutive end-of-line codes) that is interpreted by the module as the end of the strip of data, an INFOPKT_G3_STRIP_PARAMETERS infopkt is required to indicate the beginning of the second G3 strip.

When a single G3 strip consists of multiple G3 infopkts, do not insert an INFOPKT_G3_STRIP_PARAMETERS infopkt between the G3 infopkts.

Figure 4 illustrates how an electronic mail document (ASCII), accompanied by a cover sheet, a letterhead, and a signature, could be delivered to a fax machine.

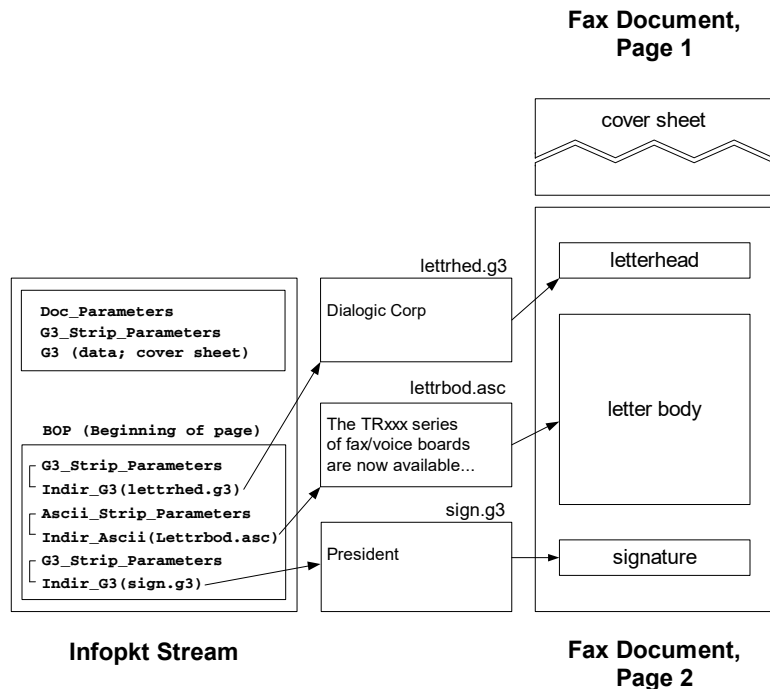


Figure 4. Conversion of a Partial Infopkt Stream to a Fax Document

Part of the infopkt stream, a file containing a series of infopkts that contains this fax example, is shown below:

```
INFOPKT_DOCUMENT_PARAMETERS
```

(Tag containing new page composition parameters; a fax infopkt stream must begin with this infopkt type)

```
INFOPKT_G3_STRIP_PARAMETERS
```

(Parameter Structure)

This two-page document contains a G3 fax document as a cover page.

INFOPKT_G3

(Cover sheet; G3 data)

It is followed by a page boundary tag,

INFOPKT_BEGINNING_OF_PAGE. The data for the cover sheet is stored in the infopkt stream.

INFOPKT_BEGINNING_OF_PAGE

(Cover sheet is on its own page)

INFOPKT_G3_STRIP_PARAMETERS

(Parameter Structure)

The second page starts with a G3 document containing the letterhead. It is stored in a separate file.

INFOPKT_INDIR_G3

(Document letterhead; G3 file name)

INFOPKT_ASCII_STRIP_PARAMETERS

(Parameter Structure)

The next strip of the second page is ASCII data which is also stored in a separate file.

INFOPKT_INDIR_ASCII

(Contents of E-Mail message; ASCII file name)

INFOPKT_G3_STRIP_PARAMETERS

(Parameter Structure)

The second page ends with G3 data, contains the signature, and is also stored in a separate file.

INFOPKT_INDIR_G3

(Signature data is in infopkt structure; G3 data)

The bulk of most infopkt files are types INFOPKT_G3 and INFOPKT_ASCII. Two files containing infopkts are concatenated.

2 - Developing Applications Using the Bfv API

This chapter describes how to develop applications with the Brooktrout Fax Software.

The chapter has the following sections:

- *[Developing a Voice Application on page 55](#)*
- *[Using Prompt Files on page 57](#)*
- *[Developing a Fax Application on page 59](#)*

Developing a Voice Application

Recording and Playing Voice

The following steps for recording and playing back speech are demonstrated:

- How to record voice
- How to play back previously recorded voice

These steps are the same for all supported operating systems.

To record and then play speech back, first select the voice channel on which you want to record your message. Then, use the *voice.c* sample program, included on your distribution CD, to record and play back speech.

Recording Voice

1. Prepare channel 2 to record your message:

```
voice -u 2 -r voice.ips
```

The command `voice` invokes the `voice.c` program, whose arguments include:

Arguments

<code>-c num</code>	Call the given number, else wait for ring.																																						
<code>-f</code>	Specify record coding format; use the number of the format or one of the following names: <table><tr><td><code>adpcm</code></td><td>1</td></tr><tr><td><code>adpcm32</code></td><td></td></tr><tr><td><code>adpcm24</code></td><td></td></tr><tr><td><code>pcm_ulaw</code></td><td>2</td></tr><tr><td><code>pcm_ulaw64</code></td><td></td></tr><tr><td><code>pcm_ulaw48</code></td><td></td></tr><tr><td><code>pcm_ulaw88</code></td><td></td></tr><tr><td><code>pcm_alaw</code></td><td>3</td></tr><tr><td><code>pcm_alaw64</code></td><td></td></tr><tr><td><code>pcm_alaw48</code></td><td></td></tr><tr><td><code>pcm_alaw88</code></td><td></td></tr><tr><td><code>g723-1</code></td><td>7</td></tr><tr><td><code>g723-1-53</code></td><td></td></tr><tr><td><code>g723-1-63</code></td><td></td></tr><tr><td><code>g729-a</code></td><td>8</td></tr><tr><td><code>sx7300</code></td><td>9</td></tr><tr><td><code>sx9600</code></td><td>10</td></tr><tr><td><code>gsm_610</code></td><td>14</td></tr><tr><td><code>gsm_660</code></td><td>15</td></tr></table>	<code>adpcm</code>	1	<code>adpcm32</code>		<code>adpcm24</code>		<code>pcm_ulaw</code>	2	<code>pcm_ulaw64</code>		<code>pcm_ulaw48</code>		<code>pcm_ulaw88</code>		<code>pcm_alaw</code>	3	<code>pcm_alaw64</code>		<code>pcm_alaw48</code>		<code>pcm_alaw88</code>		<code>g723-1</code>	7	<code>g723-1-53</code>		<code>g723-1-63</code>		<code>g729-a</code>	8	<code>sx7300</code>	9	<code>sx9600</code>	10	<code>gsm_610</code>	14	<code>gsm_660</code>	15
<code>adpcm</code>	1																																						
<code>adpcm32</code>																																							
<code>adpcm24</code>																																							
<code>pcm_ulaw</code>	2																																						
<code>pcm_ulaw64</code>																																							
<code>pcm_ulaw48</code>																																							
<code>pcm_ulaw88</code>																																							
<code>pcm_alaw</code>	3																																						
<code>pcm_alaw64</code>																																							
<code>pcm_alaw48</code>																																							
<code>pcm_alaw88</code>																																							
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<code>g723-1-53</code>																																							
<code>g723-1-63</code>																																							
<code>g729-a</code>	8																																						
<code>sx7300</code>	9																																						
<code>sx9600</code>	10																																						
<code>gsm_610</code>	14																																						
<code>gsm_660</code>	15																																						
<code>-l</code>	Loop forever, sending or receiving.																																						
<code>-n secs</code>	Specify recording time in seconds.																																						
<code>-p</code>	Play.																																						
<code>-r</code>	Record (default 10 seconds).																																						
<code>-u unitnum</code>	Use specified channel.																																						

The `-u 2` argument selects channel 2 as the recording channel and the `-r` argument places channel 2 in record mode. `Voice.ips` is the speech infopkt stream file in which to store the voice message. The channel waits for an incoming call.

-
2. Dial the phone number of the channel you selected. sure to have all the necessary cables connected for the channel to receive a call.

The `voice.c` program does not indicate when to begin recording. Begin recording when the call is connected (when you no longer hear ringing).

Playing Back the Voice Message

- **Request that a channel (0 in the example) play back your previously recorded message stored in *voice.ips*.**

1. At the system prompt, type:

```
voice -u 0 -p voice.ips
```

2. Dial the phone number of the channel. You should hear your recorded message.

Using Prompt Files

Prompt files are infopkt files that contain many individual speech phrases in a single file. Each phrase is a partial or complete prompt in infopkt format followed by an `INFOPKT_END_OF_SPEECH` infopkt with mode value 1. The prompt file starts with an `INFOPKT_PROMPT_MAP` infopkt, which contains file offsets to each individual phrase.

The Bfv API contains functions that open and close a prompt file and play individual phrases stored in it. For example, after an application opens a Brooktrout prompt file, it can call the *BfvPromptPlay* function to play any individual phrase, or it can call the *BfvPromptPlay* function multiple times to concatenate phrases and create a complete prompt.

Using Brooktrout prompt files provides two advantages. Since all of the prompts are stored in a single file, tracking and maintaining prompts is easier. And because you can combine phrases to create complete prompts, you can reduce the amount of disk space needed for overall speech storage.

Using the *mkprompt* Utility

The *mkprompt* utility converts multiple infopkt files into a Brooktrout prompt file and updates an existing Brooktrout prompt file by adding new phrases or modifying existing phrases.

When you create a Brooktrout prompt file, the *mkprompt* utility automatically assigns each infopkt file a phrase number, sequentially, in the order that you enter each file name at the command line. The *mkprompt* utility always assigns the phrase number 0 to the first infopkt file you enter.

When you update a Brooktrout prompt file, you assign a phrase number to each infopkt file you are adding to the existing Brooktrout prompt file.

Since the *mkprompt* utility cannot return phrase numbers of individual prompt files in a Brooktrout prompt file, be sure to keep your own record. You might need this information when you update your prompt file.

Both the *mkprompt* utility and the Bfv API permit you to include the text of each phrase in the Brooktrout prompt file. Create an infopkt file for each phrase in which the first infopkt in the file contains the text annotation and the remaining infopkts contain the speech that make up the phrase. The *mkprompt* utility treats the whole file as a phrase infopkt, and the *BfvPromptPlay* function skips over the annotated text.

Creating a New Prompt File

To create a new Brooktrout prompt file, at the command line type:

```
mkprompt prompt_file [phrase1.pkt phrase2.pkt...]
```

Where:

<code>prompt_file</code>	Specifies the name of the prompt file to create.
<code>phrase1.pkt,</code> <code>phrase2.pkt, ...</code>	Provides the names of the infopkt-formatted. Prompts files in the order in which you enter them at the command line.

The *mkprompt* utility automatically assigns phrase 0 to the file you enter as `phrase1.pkt` and sequentially numbers any additional files in the order that you enter them.

Updating an Existing Prompt File

To update an existing Brooktrout prompt file, at the command line type:

```
mkprompt -u phrase_num prompt_file phrase.pkt:
```

Where:

<code>-u</code>	Specifies the update command.
<code>phrase_num</code>	Provides the index number to assign the infopkt-formatted input file.
<code>prompt_file</code>	Provides the name of the prompt file.
<code>phrase.pkt</code>	Provides the name of the infopkt-formatted prompt file to add to the prompt file.

The *mkprompt* utility opens the existing Brooktrout prompt file and append the new phrase if the phrase number you specify is new or replace the old phrase whose phrase number matches the phrase number you specify.

Developing a Fax Application

Sending and Receiving a Fax

The following are the step-by-step instructions for transmitting and receiving a fax:

- How to create an infopkt stream using the *mkinfopk* utility.
- How to send a fax (an infopkt stream file) from one channel to another channel in your system.
- How to send a fax from an external fax machine to one channel in your system. This same fax is then transmitted back to the same fax machine from the same channel in your system.

Sending a Fax from One Channel to Another

- **Use the *fax* sample program to send a fax from one channel to another in your system.**

1. Prepare channel 1 to receive a fax:

```
fax -u 1 -r recfile.ips
```

fax invokes the sample fax program with the following arguments:

<code>-u</code>	Specifies that the following number is the number of the channel that receives a fax (in this case channel 1 is used).
<code>-r</code>	Places the channel (1) in receive mode.
<code>recfile.ips</code>	Creates a file, <i>recfile.ips</i> , in which to receive a fax.

The channel is now set to receive a fax and is waiting for an incoming call.

2. Send a fax from channel 0 to channel 1.

In a second window, type:

```
fax -u 0 -s wphonenum fax.ips
```

<code>-u</code>	Specifies that the following number is the number of the channel that sends a fax (in this case channel 0 is used).
<code>-s</code>	Places the channel (0) in send mode.
<code>w</code>	■ In robbed-bit T1 TDM environments, checks the signaling if <i>w</i> is the first character of the string. ■ In E&M immediate mode, causes the channel to wait and see if the T1 slot is free. ■ In E&M wink mode, causes the channel to wait for a wink signal from the remote side. ■ Anywhere else in the string, waits for dial tone.
<code>phonenum</code>	Specifies the phone number of the channel to receive the fax (in this case channel 1).
<code>fax.ips</code>	Sends the infopkt stream file <i>fax.ips</i> previously created with the <i>mkinfopk</i> utility. Refer to The Infopkt Stream on page 44

Sending a Fax to a Channel from an External Fax Machine

Use the *fax.c* sample program to send a fax from an external fax machine to a channel in your system. Then send the same fax back to the same fax machine from the same channel in your system.

1. Prepare channel 1 to receive a fax as you did in the previous example. Use the same input file name *recfile.ips* used to send a fax from one channel to another in the same system:

```
fax -u 1 -r recfile.ips
```

Channel 1 is now set to receive a fax and is waiting for an incoming call.

2. At the fax machine, insert the sample fax in the fax machine and call the channel's (channel 1) phone number in the normal way. At the tone send your sample fax.

Wait for the selected channel to receive the sample fax.

3. Send the received fax back to the fax machine:

```
fax -u 1 -s wphonenum recfile.ips
```

This time, **phonenum** is the phone number of the external fax machine.

You can examine the structure of the infopkt stream *recfile.ips* before you transmit it back to the fax machine.

```
decode recfile.ips
```

Using Bfv API Fax Functions

The following sections show how to send and receive facsimiles using the high- and low-level function calls, noninfopkt function calls, and TIFF-F function calls. It also shows how to send and receive facsimiles in MR and MMR format, access infopkt streams and TIFF-F fax files from an application, combine data on a single page using TIFF-F fax files, interpret fax status information from an application, and how to use prompt files.

The following declarations are assumed to be in effect for all examples.

```
struct args_line_admin args_admin;  
struct args_telephone args_tel;  
struct args_speech args_speech;  
struct args_fax args_fax;  
struct args_fax_t30_params args_t30;  
struct args_fax_page_params args_page;  
struct args_fax_strip_params args_strip;  
struct args_tone args_tone;  
struct args_download args_download;  
struct args_dh args_dh;  
struct args_intlimit args_intlimit;  
struct args_infopkt args_infopkt;  
struct args_tiff args_tiff;  
struct args_cc args_cc
```

Using High- and Low-Level Functions

The Bfv API library contains both high- and low-level function calls (see *Table 1 on page 38*). Several low-level function calls are combined into one high-level function to provide a quick and easy method to send or receive a facsimile.

The low-level functions, on the other hand, provide more flexibility and functionality than the higher level function calls.

Sending a Fax Using High-Level Function Calls A typical way to send a fax using the high-level fax function calls is demonstrated below. Each function is presented in sequential order, and the action it performs is described beneath it.

```
BT_ZERO(args_admin);  
args_admin.unit = unit;  
lp = BfvLineAttach(&args_admin);
```

Attaches to a free channel and gets a line pointer.

```
args_admin.config_file_name = "usrcnfig.cfg";  
BfvLineReset (lp,&args_admin);
```

Resets the channel and sets the user-configured options selected in the user-defined configuration file named, *usrcnfig.cfg*.

```
BT_ZERO(args_infopkt);  
args_infopkt.fname = name;  
args_infopkt.fmode = "r";  
ips = BfvInfopktOpen(&args_infopkt);
```

Opens the infopkt-formatted file called *name* for reading and transmission.

```
BT_ZERO(args_tel);  
args_tel.phonenum = "w7814499009";  
args_tel.call_protocol_code = CALL_PROTOCOL_FAX;  
args_tel.func = userfunc;  
args_tel.arg = userarg;  
BfvLineOriginateCall (lp,&args_tel);
```

Dials the phone number, monitors call progress, calls the user function to optionally decide when to terminate call progress.

```
BT_ZERO(args_fax);  
args_fax.s_ips = ips;  
args_fax.local_id = local_id;  
BfvFaxSend(lp,&args_fax);
```

Transmits documents based on an infopkt stream.

```
BT_ZERO(args_infopkt);  
args_infopkt.ips = ips;  
BfvInfopktClose (&args_infopkt);
```

Closes the infopkt stream file after the file is sent.

```
BT_ZERO(args_tel);  
BfvLineTerminateCall (lp, &args_tel);
```

Disconnect the call and sets the line state to IDLE.

```
BT_ZERO(args_admin);  
BfvLineDetach (lp, &args_admin);
```

Frees all memory for the attached line and closes the device.

Receiving a Fax Using High-Level Function Calls A typical way to receive a fax using the high-level fax function calls is demonstrated below. Each function is presented in sequential order, and the action it performs is described beneath it.

```
BT_ZERO(args_admin);  
args_admin.unit = unit;  
lp = BfvLineAttach(&args_admin);
```

Attaches to a free channel and gets a line pointer.

```
args_admin.config_file_name = "usrcnfig.cfg";  
BfvLineReset (lp, &args_admin);
```

Resets the channel and sets the user-configured options selected in the user-defined configuration file named *usrcnfig.cfg*.

```
BT_ZERO(args_infopkt);  
args_infopkt.fname = name;  
args_infopkt.fmode = "w";  
ips = BfvInfopktOpen(&args_infopkt);
```

Opens the infopkt-formatted file called *name* for writing.

```
BT_ZERO(args_tel);  
args_tel.timeout = 0L;  
BfvLineWaitForCall (lp, &args_tel);
```

Waits without a timeout for an incoming call and performs call screening when the call occurs.

```
BfvLineAnswer (lp, &args_tel);
```

Answers the incoming call and sets the line state to CONNECTED.

```
BT_ZERO(args_fax);  
args_fax.r_ips = ips;  
args_fax.local_id = local_id;  
BfvFaxReceive (lp, &args_fax);
```

Receives fax pages and puts them into the infopkt stream ips.

```
BT_ZERO(args_infopkt);  
args_infopkt.ips = ips;  
BfvInfopktClose (&args_infopkt);
```

Closes the infopkt stream file after the file is sent.

```
BT_ZERO(args_tel);  
BfvLineTerminateCall (lp, &args_tel);
```

Disconnect the call and sets the line state to IDLE.

```
BT_ZERO(args_admin);  
BfvLineDetach (lp, &args_admin);
```

Closes the infopkt stream file after the file is received.

```
BT_ZERO(args_admin);  
BfvLineDetach (lp, &args_admin);
```

Frees all memory for the attached line and closes the device.

Sending a Fax Using Low-Level Infopkt Function Calls One way to send a fax using the low-level infopkt fax function calls is demonstrated below. Each function is presented in sequential order, and the action it performs is listed beneath it.

```
BT_ZERO(args_admin);  
args_admin.unit = unit;  
lp = BfvLineAttach(&args_admin);
```

Attaches to a free channel and gets a line pointer.

```
args_admin.config_file_name = "usrconfig.cfg";  
BfvLineReset(lp, &args_admin);
```

Resets the channel and sets the user-configured options selected in the user-defined configuration file named *usrconfig.cfg*.

```
BT_ZERO(args_infopkt);  
args_infopkt.fname = name;  
args_infopkt.fmode = "r";  
ips = BfvInfopktOpen(&args_infopkt);
```

Opens the infopkt-formatted file called *name* for reading and transmission.

```
BT_ZERO(args_cc);  
args_cc.phonenum = "w7814499009";  
args_cc.call_protocol_code = CALL_PROTOCOL_ FAX;  
BfvCallSetup(lp, &args_cc);
```

Dials the phone number

```
BT_ZERO(args_cc);  
args_cc.call_protocol_code = CALL_PROTOCOL_ FAX;  
BfvCallWaitForComplete(lp, &args_cc);
```

Monitors call progress, calls the user function to optionally decide when to terminate call progress.

```
BT_ZERO(args_fax);  
args_fax.local_id = "Id_string";  
BfvFaxSetLocalID(lp, &args_fax);
```

Sets the local id.

```
BT_ZERO(args_fax);  
args_fax.s_ips = ips;  
BfvFaxBeginSend(lp, &args_fax);
```

Begins the handshaking procedure between the two machines.

```
BfvFaxGetRemoteInfo(lp, &args_fax);
```

Waits for the called machine to send its ID and capabilities.

```
BfvFaxWaitForTraining(lp, &args_fax);
```

Waits for the completion of the Phase B handshaking procedure.

```
for (;;)
{
    BT_ZERO(args_fax);
    args_fax.s_ips = ips;
    if ((ret = BfvFaxNextPage(lp, &args_fax)) <= 0 )
        break;
    BfvFaxSendPage(lp, &args_fax);
}
```

Loops through the infopkt stream, getting the next page and transmitting it to the driver.

```
BfvFaxEndOfDocument(lp, &args_fax);
```

Finishes up when the infopkt stream is exhausted.

```
BT_ZERO(args_infopkt);
args_infopkt.ips = ips;
BfvInfopktClose (&args_infopkt);
```

Closes the infopkt stream file after the file is sent.

```
BT_ZERO(args_tel);
BfvLineTerminateCall (lp, &args_tel);
```

Disconnect the call and sets the line state to IDLE.

```
BT_ZERO(args_admin);
BfvLineDetach (lp, &args_admin);
```

Frees all memory for the attached line and closes the device.

You can replace some low-level functions with a high-level function, for example:

BfvCallSetup
BfvCallWaitForComplete

These low level functions are replaced with the high level function *BfvLineOriginateCall*.

BfvFaxSetLocalID
BfvFaxBeginSend
BfvFaxGetRemoteInfo
BfvFaxWaitForTraining
BfvFaxSendPage
BfvFaxEndOfDocument

These low level functions are replaced with the high level function *BfvFaxSend*.

Receiving a Fax Using Low-Level Infopkt Function Calls One way to receive a fax using the low-level infopkt fax function calls is demonstrated below. Each function is presented in sequential order, and the action it performs is described beneath it.

```
BT_ZERO(args_admin);
args_admin.unit = unit;
lp = BfvLineAttach(&args_admin);
```

Attaches to a free channel and gets a BTLINE pointer.

```
args_admin.config_file_name = "usrcnfig.cfg";
BfvLineReset(lp, &args_admin);
```

Resets the channel and sets the user-configured options in the user-defined configuration file *usrcnfig.cfg*.

```
BT_ZERO(args_infopkt);
args_infopkt.fname = name;
args_infopkt.fmode = "w";
ips = BfvInfopktOpen(&args_infopkt);
```

Opens the infopkt-formatted file, name, to store the received fax.

```
BT_ZERO(args_tel);
args_tel.timeout = 0L;
BfvLineWaitForCall(lp, &args_tel);
```

Waits without a timeout for an incoming call and performs call screening when the call occurs.

```
BfvLineAnswer(lp, &args_tel);
```

Answers the incoming call and sets the line state to CONNECTED.

```
BT_ZERO(args_fax);
args_fax.local_id = "Id_string";
BfvFaxSetLocalId(lp, &args_fax);
```

Sets the local ID to transmit to the sending machine.

```
BT_ZERO(args_fax);
BfvFaxBeginReceive(lp, &args_fax);
```

Begins the Phase B handshaking procedure.

```
BfvFaxGetRemoteInfo(lp, &args_fax);
```

Waits for the remote to send its ID and capabilities.

Note: The previous phone call is terminated by the application if the remote fax's ID does not match the expected value.

```
BfvFaxWaitForTraining(lp, &args_fax);
```

Waits for the completion of the Phase B handshaking process.

```
BT_ZERO(args_fax);
args_fax.r_ips = ips;
BfvFaxReceivePages(lp, &args_fax);
```

Receives and writes the fax data to the infopkt stream file pointed to by ips.

```
BT_ZERO(args_infopkt);
args_infopkt.ips = ips;
BfvInfopktClose (&args_infopkt);
```

Closes the infopkt stream file after the file is sent.

```
BT_ZERO(args_tel);
BfvLineTerminateCall (lp, &args_tel);
```

Disconnect the call and sets the line state to IDLE.

```
BT_ZERO(args_admin);
BfvLineDetach (lp, &args_admin);
```

Frees all the memory for the attached line and closes the device.

You can replace some low-level functions with a high-level function, for example:

<i>BfvFaxBeginReceive</i>	These low level functions are
<i>BfvFaxGetRemoteInfo</i>	replaced with the high level function
<i>BfvFaxSetLocalId</i>	<i>BfvFaxReceive</i> .
<i>BfvFaxWaitForTraining</i>	
<i>BfvFaxReceivePages</i>	

See the applications in the sample application directory for more detailed information.

Sending a Fax Using Function Calls for Noninfopkt-Formatted Raw G3 Files

One way to send a fax using function calls for noninfopkt-formatted raw G3 files is demonstrated below. Each function is presented in sequential order, and the action it performs is described beneath it.

This example sends a two-page fax whose page and strip data are stored in noninfopkt-formatted files. The first page consists of a Group 3 letterhead (*ltrhd.g3*), a body in ASCII (*main.txt*), and a signature file in Group 3 (*sig.G3*). The second page is an ASCII file (*memo.txt*).

```
BT_ZERO(args_admin);
args_admin.unit = unit;
lp = BfvLineAttach(&args_admin);
```

Attaches to a free channel and gets a line pointer.

```
args_admin.config_file_name = "usrconfig.cfg";
BfvLineReset(lp, &args_admin);
```

Resets the channel and sets the user-configured options selected in the user-defined configuration file named *usrncfig.cfg*.

```
BT_ZERO(args_tel);
args_tel.phonenum = "w7814499009";
args_tel.call_protocol_code = CALL_PROTOCOL_FAX;
args_tel.func = userfunc;
args_tel.arg = userarg;
BfvLineOriginateCall(lp, &args_tel);
```

Dials the phone number, monitors call progress, calls the user function to optionally decide when to terminate call progress.

```
BT_ZERO(args_t30);
args_t30.bit_rate = BITRATE_14400;
args_t30.scan_time = SCANTIME_0;
BfvFaxT30Params(lp, &args_t30);
```

Configures the channel's maximum transmission rate. This function is optional.

```
BT_ZERO(args_page);
args_page.top_margin = 0;
args_page.bottom_margin = 0;
args_page.length = 1143;
args_page.ascii_pad = 1;
args_page.image_pad = 0;
args_page.image_break = 0;
args_page.image_margin = 0;
BfvFaxPageParams(lp, &args_page);
```

Sets the page parameters: no top or bottom margins, a page length of 1143 (normal) G3 lines, no padding of short ASCII pages, no padding of short images, no breaking of images, and no margins for images.

```
BT_ZERO(args_fax);
args_fax.resolution = RES_200H_200V;
args_fax.width = WIDTH_A4;
BfvFaxBeginSendRaw(lp, &args_fax);
```

Begins the handshaking procedure and indicates that the first page is in fine resolution and has A4 width.

```
BfvFaxGetRemoteInfo(lp, &args_fax);
```

Waits for the called machine to send its ID and capabilities.

```
BfvFaxWaitForTraining(lp, &args_fax);
```

Waits for the completion of the Phase B handshaking procedure.

```
BT_ZERO(args_strip);
args_strip.fmt = DATA_G3;
args_strip.resolution = RES_200H_100V;
args_strip.width = WIDTH_A4;
BfvFaxStripParams(lp, &args_strip);
```

Sets the G3 strip parameters for the G3 strip *ltrhd.g3*.

```
BT_ZERO(args_fax);
args_fax.fname = "ltrhd.g3";
args_fax.fmt = DATA_G3;
BfvFaxSendFile(lp, &args_fax);
```

Sends the G3 letterhead data file *ltrhd.g3* to the driver.

```
BT_ZERO(args_strip);
args_strip.fmt = DATA_ASCII;
args_strip.resolution = RES_200H_100V;
args_strip.width = WIDTH_A4;
args_strip.left_margin = 5;
args_strip.right_margin = 0;
args_strip.line_spacing = 2;
args_strip.eof_char = 0x1a;
BfvFaxStripParams(lp, &args_strip);
```

Sets the ASCII parameters for the ASCII strip *main.txt*, since it differs from the default.

```
BT_ZERO(args_fax);
args_fax.fname = "main.txt";
args_fax.fmt = DATA_ASCII;
BfvFaxSendFile(lp, &args_fax);
```

Sends the ASCII text body file *main.txt* to the driver.

```
BT_ZERO(args_strip);
args_strip.fmt = DATA_G3;
args_strip.resolution = RES_200H_100V;
args_strip.width = WIDTH_A4;
BfvFaxStripParams(lp, &args_strip);
```

Sets the G3 strip parameters for the G3 strip *sig.G3*.

```
BT_ZERO(args_fax);
args_fax.fname = "sig.g3";
args_fax.fmt = DATA_G3;
BfvFaxSendFile(lp, &args_fax);
```

Sends the G3 signature file *sig.G3* to the driver.

```
BT_ZERO(args_fax);
args_fax.resolution = RES_200H_100V;
args_fax.width = WIDTH_A4;
BfvFaxNextPageRaw(lp, &args_fax);
```

Sends an end-of-page command to the driver and indicates that another page (normal resolution and A4 width) follows.

```
BT_ZERO(args_page);
args_page.top_margin = 5;
args_page.bottom_margin = 5;
```

```
args_page.length = 1143;  
args_page.ascii_pad = 0;  
BfvFaxPageParams(lp, &args_page);
```

Sets the page parameters for this page since they differ from those of the first page.

```
BT_ZERO(args_fax);
args_fax.fname = "memo.txt";
args_fax.fmt = DATA_ASCII;
BfvFaxSendFile(lp, &args_fax);
```

Sends the ASCII text file, *memo.txt* the only file on the second page, to the driver.

```
BfvFaxEndOfDocument(lp, &args_fax);
```

Indicates to the driver that the second page is the last page of the transmission.

```
BT_ZERO(args_tel);
BfvLineTerminateCall (lp, &args_tel);
```

Disconnect the call and sets the line state to IDLE.

```
BT_ZERO(args_admin);
BfvLineDetach (lp, &args_admin);
```

Frees all the memory for the attached line and closes the device.

Receiving a Fax Using Function Calls for Noninfopkt-Formatted Raw G3 Files

One way to receive a fax using function calls for noninfopkt-formatted raw G3 files is demonstrated below. Each function is presented in sequential order, and the action it performs is described beneath it.

This example receives fax data into a buffer only and does not include instructions for further processing the contents of the buffer.

```
BT_ZERO(args_admin);
args_admin.unit = unit;
lp = BfvLineAttach(&args_admin);
```

Attaches to a free channel and gets a BTLINE pointer.

```
args_admin.config_file_name = "usrcnfig.cfg";
BfvLineReset (lp, &args_admin);
```

Resets the channel and sets the user-configured options in the user-defined configuration file *usrcnfig.cfg*.

```
BT_ZERO(args_tel);
args_tel.timeout = 0L;
BfvLineWaitForCall (lp, &args_tel);
```

Waits without a timeout for an incoming call and performs call screening when the call occurs.

```
BfvLineAnswer (lp, &args_tel);
```

Answers the incoming call and sets the line state to CONNECTED

```
BT_ZERO(args_fax);
args_fax.local_id = "Id_string";
BfvFaxSetLocalID(lp, &args_fax);
```

Sets the local ID to transmit to the sending machine.

```
BT_ZERO(args_fax);
BfvFaxBeginReceive(lp, &args_fax);
```

Begins the Phase B handshaking procedure.

```
BfvFaxGetRemoteInfo (lp, &args_fax);
```

Waits for the remote to send its ID and capabilities.

Note: The previous phone call is terminated by the application if the remote fax machine's ID does not match the expected value.

```
BfvFaxWaitForTraining (lp, &args_fax);
```

Waits for the completion of the Phase B handshaking process.

```
do
{
    BT_ZERO(args_fax);
    args_fax.buf = buf;
    args_fax.size = size;

    /* receive data into buffer */
    if (BfvFaxReceiveData(lp, &args_fax) <= 0)
        break;

    /* Process buffer contents */
    Process(buf);
}
```

Keeping track of the resolution of each page is the application's responsibility.

When the function returns a 0 at exit from the loop, the application must determine, from the value of `args_fax.expect_another`, if there is another page to receive.

```
BfvFaxEndReception(lp, &args_fax);
```

Call this function when there are no more pages to receive.

```
BT_ZERO(args_tel);
BfvLineTerminateCall (lp, &args_tel);
```

Disconnect the call and sets the line state to IDLE.

```
BT_ZERO(args_admin);
BfvLineDetach (lp, &args_admin);
```

Frees all the memory for the attached line and closes the device.

Sending a Fax Using Calls for TIFF-F Files

One way to send a fax using function calls for TIFF-F files is demonstrated below. Each function is presented in sequential order, and the action it performs is described beneath it.

```
BT_ZERO(args_admin);  
args_admin.unit = unit;  
lp = BfvLineAttach(&args_admin);
```

Attaches to a free channel and gets a line pointer.

```
args_admin.config_file_name = "usrconfig.cfg";  
BfvLineReset(lp, &args_admin);
```

Resets the channel and sets the user-configured options selected in the user-defined configuration file named *usrconfig.cfg*.

```
BT_ZERO(args_tiff);  
args_tiff.fname = name;  
args_tiff.fmode = "r";  
tp = BfvTiffOpen(&args_tiff);
```

Opens the TIFF-F file name for reading and transmission.

```
BT_ZERO(args_tel);  
args_tel.phonenum = "w7814499009";  
args_tel.call_protocol_code = CALL_PROTOCOL_FAX;  
args_tel.func = userfunc;  
args_tel.arg = userarg;  
BfvLineOriginateCall(lp, &args_tel);
```

Dials the phone number, monitors call progress, calls the user function to optionally decide when to terminate call progress.

```
BT_ZERO(args_fax);  
args_fax.local_id = "Id_string";  
BfvFaxSetLocalID(lp, &args_fax);
```

Sets the local id.

```
BT_ZERO(args_fax);  
args_fax.s_tp = tp;  
BfvFaxBeginSendTiff(lp, &args_fax);
```

Begins the Phase B handshaking procedure.

```
BfvFaxGetRemoteInfo(lp, &args_fax);
```

Waits for the called machine to send its ID and capabilities.

```
BfvFaxWaitForTraining(lp, &args_fax);
```

Waits for the completion of the Phase B handshaking procedure.

```
for (;;)
{
    BT_ZERO(args_fax);
    args_fax.s_tp = tp;
    args_fax.combine = 0;
    if (BfvFaxNextPageTiff(lp, &args_fax) <= 0)
        break;
    BT_ZERO(args_fax);
    args_fax.s_tp = tp;
    if (BfvFaxSendPageTiff(lp, &args_fax) < 0)
        break;
}
```

Loops through the TIFF-F file, getting the next page and sending it to the driver.

```
BfvFaxEndOfDocument(lp, &args_fax);
```

Finishes up when the TIFF data is exhausted.

```
BT_ZERO(args_tiff);
args_tiff.tp = tp;
BfvTiffClose(&args_tiff);
```

Closes the TIFF-F file after the file is received.

```
BT_ZERO(args_tel);
BfvLineTerminateCall (lp, &args_tel);
```

Disconnect the call and sets the line state to IDLE.

```
BT_ZERO(args_admin);
BfvLineDetach (lp, &args_admin);
```

Disconnect the call and sets the line state to IDLE.

```
BT_ZERO(args_admin);
BfvLineDetach (lp, &args_admin);
```

Frees all the memory for the attached line and closes the device.

Receiving a Fax Using Calls for TIFF-F Files

One way to receive a fax using function calls for TIFF-F files is demonstrated below. Each function is presented in sequential order, and the action it performs is described beneath it.

```
BT_ZERO(args_admin);
args_admin.unit = unit;
lp = BfvLineAttach(&args_admin);
```

Attaches to a free channel and gets a BTLIN pointer.

```
args_admin.config_file_name = "usrcnfig.cfg";  
BfvLineReset(lp, &args_admin);
```

Resets the channel and sets the user-configured options in the user-defined configuration file named *usrcnfig.cfg*.

```
BT_ZERO(args_tiff);  
args_tiff.fname = name;  
args_tiff.fmode = "w";  
tp = BfvTiffOpen(&args_tiff);
```

Opens the TIFF-F file *name* to store the received fax.

```
BT_ZERO(args_tel);  
args_tel.timeout = 0L;  
BfvLineWaitForCall (lp, &args_tel);
```

Waits without a timeout for an incoming call and performs call screening when the call occurs.

```
BfvLineAnswer (lp, &args_tel);
```

Answers the incoming call and sets the line state to CONNECTED.

```
BT_ZERO(args_fax);  
args_fax.local_id = "Id_string";  
BfvFaxSetLocalID(lp, &args_fax);
```

Sets the local ID to transmit to the sending machine.

```
BT_ZERO(args_fax);  
BfvFaxBeginReceive(lp, &args_fax);
```

Begins the Phase B handshaking procedure.

```
BfvFaxGetRemoteInfo(lp, &args_fax);
```

Waits for the remote to send its ID and capabilities.

```
BfvFaxWaitForTraining(lp, &args_fax);
```

Waits for the completion of the Phase B handshaking process.

```
do  
{  
    BT_ZERO(args_fax);  
    args_fax.r_tp = tp;  
}  
while (BfvFaxRcvPageTiff(lp, &args_fax) > 0);
```

Receives and writes the fax data to the TIFF-F file pointed to by *tp*.

```
BfvFaxEndReception(lp, &args_fax);
```

After the last page is received, waits for the T.30 handshaking confirmation sequence to complete.

```
BT_ZERO(args_tiff);
args_tiff.tp = tp;
BfvTiffClose(&args_tiff);
```

Closes the TIFF file after the file is received.

```
BT_ZERO(args_admin);
BfvLineDetach (lp, &args_admin);
```

Frees all the memory for the attached line and closes the device.

Receiving and Storing a Fax in MMR or MR Format

Receiving an Infopkt-Formatted Fax and Storing it in MMR Format

A typical way to receive a fax that is made up of infopkts and store it in MMR format is demonstrated below. Each function is presented in sequential order, and the action it performs is described beneath it.

When MR or MMR facsimiles are received in infopkt format, the data format type specification is automatically included through the `g3strpkt` infopkt structure. Retransmission of these infopkt files is done the same way transmission of MH files is done (see *Sending a Fax Using Low-Level Infopkt Function Calls on page 65* for detailed instructions).

The application controls the format of images received from the channel; this format is independent of the format of data received by the channel.

```
BT_ZERO(args_admin);
args_admin.unit = unit;
lp = BfvLineAttach(&args_admin);
```

Attaches to a free channel and gets a line pointer.

```
args_admin.config_file_name = "usrconfig.cfg";
BfvLineReset(lp, &args_admin);
```

Resets the channel and sets the user-configured options selected in the user-defined configuration file named *usrconfig.cfg*.

```
BT_ZERO(args_infopkt);
args_infopkt.fname = name;
args_infopkt.fmode = "w";
ips = BfvInfopktOpen(&args_infopkt);
```

Opens the infopkt-formatted file called *name* to store the received fax.

```
BT_ZERO(args_tel);
args_tel.timeout = 0L;
BfvLineWaitForCall (lp, &args_tel);
```

Waits without a timeout for an incoming call and performs call screening when the call occurs

```
BfvLineAnswer (lp, &args_tel);
```

Answers the incoming call and sets the line state to CONNECTED.

```
BT_ZERO(args_fax);
args_fax.fmt = FMT_MMR_ALIGN_MSB;
BfvFaxSetReceiveFmt (lp, &args_fax);
```

Sets the format used to pass the received fax data from the channel to the computer. In this example, MMR data format – byte aligned, most significant bit first – is specified. See the *fmt* parameter description for detailed information on all of the data format types that are available through *BfvFaxSetReceiveFmt*.

```
BT_ZERO(args_fax);
args_fax.r_ips = ips;
args_fax.local_id = local_id;
BfvFaxReceive (lp, &args_fax);
```

Receives fax pages and puts them into the infopkt stream *ips*.

```
BT_ZERO(args_infopkt);
args_infopkt.ips = ips;
BfvInfopktClose (&args_infopkt);
```

Closes the infopkt stream file after the file is received.

```
BT_ZERO(args_tel);
BfvLineTerminateCall (lp, &args_tel);
```

Disconnect the call and sets the line state to IDLE.

```
BT_ZERO(args_admin);
BfvLineDetach (lp, &args_admin);
```

Frees all the memory for the attached line and closes the device.

Receiving a Noninfopkt-Formatted Fax and Storing It in MR Format

A typical way to receive a fax that is made up of noninfopkts and store it in MR data format is demonstrated below. Each function is presented in sequential order, and the action it performs is described beneath it.

```
BT_ZERO(args_admin);
args_admin.unit = unit;
lp = BfvLineAttach(&args_admin);
```

Attaches to a free channel and gets a BTLIN pointer.

```
args_admin.config_file_name = "usrcnfig.cfg";  
BfvLineReset(lp, &args_admin);
```

Resets the channel and sets the user-configured options in the user-defined configuration file *usrcnfig.cfg*.

```
BT_ZERO(args_tel);  
args_tel.timeout = 0L;  
BfvLineWaitForCall(lp, &args_tel);
```

Waits for the detection of an incoming call.

```
BfvLineAnswer(lp, &args_tel);
```

Answers the phone line by going off-hook.

```
BT_ZERO(args_fax);  
args_fax.fmt = FMT_MR_UNALIGN_MSB;  
BfvFaxSetReceiveFmt(lp, &args_fax);
```

Sets the format used to pass the received fax data from the channel to the computer. In this example, MR data format – byte unaligned, least significant bit first – is specified.

See the *fmt* parameter description for detailed information on all of the data format types that are available through *BfvFaxSetReceiveFmt*.

```
BT_ZERO(args_fax);  
args_fax.local_id = "Id_string";  
BfvFaxSetLocalID(lp, &args_fax);
```

Sets the local ID to transmit to the sending machine.

```
BT_ZERO(args_fax);  
BfvFaxBeginReceive(lp, &args_fax);
```

Begins the Phase B handshaking procedure.

```
BfvFaxGetRemoteInfo(lp, &args_fax);
```

Waits for the remote end to send its ID and capabilities.

Note: The previous phone call is terminated by the application if the remote fax machine's ID does not match the expected value.

```
BfvFaxWaitForTraining(lp, &args_fax);
```

Waits for the completion of the Phase B handshaking process.

```
do  
{  
    BT_ZERO(args_fax);  
    args_fax.buf = buf;  
    args_fax.size = size;
```

```

        /* receive data into buffer */
        if (BfvFaxReceiveData(lp, &args_fax) <= 0)
            break;

        /* Process buffer contents */
        Process(buf);
    }

```

Keeping track of the resolution and the data format (previously set by *BfvFaxSetReceiveFmt*) of each page, is the application's responsibility.

When the function returns a 0 at exit from the loop, the application must determine, from the value of *args_fax.expect_another*, if there is another page to receive.

```
BfvFaxEndReception(lp, &args_fax);
```

Call this function when there are no more pages to receive.

```
BT_ZERO(args_admin);
BfvLineDetach (lp, &args_admin);
```

Frees all the memory for the attached line and closes the device.

Sending a Noninfopkt-Formatted Fax Stored in MMR Format

One way to send a fax using function calls for noninfopkt-formatted raw G3 files is demonstrated below. Each function is presented in sequential order, and the action it performs is described beneath it.

This example sends a one-page fax whose page and strip data are stored in noninfopkt-formatted files. The page consists of a Group 3 document (*mmrdoc.g3*) that is stored on disk in MMR format.

The application controls the format of images sent to the channel; this format is independent of the format of data transmitted by the channel.

```
BT_ZERO(args_admin);
args_admin.unit = unit;
lp = BfvLineAttach(&args_admin);
```

Attaches to a free channel and gets a line pointer.

```
args_admin.config_file_name = "usrcnfig.cfg";
BfvLineReset(lp, &args_admin);
```

Resets the channel and sets the user-configured options selected in the user-defined configuration file named *usrcnfig.cfg*.

```
BT_ZERO(args_tel);
args_tel.phonenum = "w7814499009";
args_tel.call_protocol_code = CALL_PROTOCOL_FAX;
```

```
args_tel.func = userfunc;
args_tel.arg = userarg;
BfvLineOriginateCall(lp, &args_tel);
```

Dials the phone number, monitors call progress, and calls the user function to optionally decide when to terminate call progress.

```
BT_ZERO(args_t30);
args_t30.bit_rate = BITRATE_14400;
args_t30.scan_time = SCANTIME_0;
BfvFaxT30Params(lp, &args_t30);
```

Configures the channel's maximum transmission rate. This function is optional.

```
BT_ZERO(args_page);
args_page.top_margin = 0;
args_page.bottom_margin = 0;
args_page.length = 1143;
args_page.ascii_pad = 1;
BfvFaxPageParams(lp, &args_page);
```

Sets the page parameters: no top or bottom margins, a page length of 1143 (normal) G3 lines, and no padding of short ASCII pages, no padding of short images, no breaking of images, and no margins for images.

```
BT_ZERO(args_fax);
args_fax.resolution = RES_200H_100V;
args_fax.width = WIDTH_A4;
BfvFaxBeginSendRaw(lp, &args_fax);
```

Begins the handshaking procedure and indicates that the first page is in normal resolution and has A4 width.

```
BfvFaxGetRemoteInfo(lp, &args_fax);
```

Waits for the called machine to send its ID and capabilities.

```
BfvFaxWaitForTraining(lp, &args_fax);
```

Waits for the completion of the Phase B handshaking procedure.

```
BT_ZERO(args_strip);
args_strip.fmt = FMT_MMR_ALIGN_MSB;
args_strip.resolution = RES_200H_100V;
args_strip.width = WIDTH_A4;
BfvFaxStripParams(lp, &args_strip);
```

Sets the G3 parameters for the G3 document *mmrdoc.g3*, since the data format differs from the default (MH).

```
BT_ZERO(args_fax);
args_fax.fname = "mmrdoc.g3";
args_fax.fmt = FMT_MMR_ALIGN_MSB;
BfvFaxSendFile(lp, &args_fax);
```

Sends the G3 document data file *mmrdoc.g3* stored on disk in MMR format, to the driver.

```
BfvFaxEndOfDocument(lp, &args_fax);
```

Indicates to the driver that this page is the last page of the transmission.

```
BT_ZERO(args_tel);  
BfvLineTerminateCall (lp, &args_tel);
```

Disconnect the call and sets the line state to IDLE.

```
BT_ZERO(args_admin);  
BfvLineDetach (lp, &args_admin);
```

Frees all the memory for the attached line and closes the device.

Accessing an Infopkt Stream from an Application

The function calls *BfvFaxSendPage* and *BfvFaxNextPage* are typically used in a loop. Both read infopkts from the infopkt stream for processing.

BfvFaxSendPage reads infopkts and processes them in a loop. When data (embedded or indirect) or strip parameter type infopkts are encountered, the indicated parameters and data format (ASCII or G3) commands are sent to the channel, the data is downloaded to the driver buffer, and the next infopkt is read. If any other type of infopkt is encountered, the current position in the infopkt stream remains unchanged, and the function returns to the calling routine.

BfvFaxNextPage reads infopkts from the current position in the infopkt stream. All consecutive infopkts of a new-page type,

```
INFOPKT_PAGE_PARAMETERS,  
INFOPKT_T30_PARAMETERS,  
INFOPKT_BEGINNING_OF_PAGE, or  
INFOPKT_DOCUMENT_PARAMETERS,
```

are processed, and relevant data and a single end-of-page command are sent to the channel. If no infopkts of a new-page type are found, the current position in the infopkt stream remains unchanged, and the function returns to the calling routine.

A program to read a file containing infopkts could look like this:

```
BT_ZERO(args_infopkt);  
args_infopkt.fname = "filename";  
args_infopkt.fmode = "r";  
ips = BfvInfopktOpen(&args_infopkt);
```

```
for (;;)
{
    BT_ZERO(args_infopkt);
    args_infopkt.ips = ips;
    args_infopkt.i_mode = INDIR_MODE_FOLLOW;
    if ((ip = BfvInfopktGet(&args_infopkt)) == NULL)
        break;
    process(ip);
}
```

With the flag set to `INDIR_MODE_FOLLOW` or `INDIR_MODE_FOLLOW_NOUSER`, *BfvInfopktGet* follows indirect infopkt links automatically, so the user sees only the actual data. Other flags let the user examine a file without following indirect infopkt links. See the *BfvInfopktUser* function in *Volume 4, Fax Processing*, in the *Dialogic® Brooktrout® Bfv API Reference Manual* for detailed information on how to access user-defined infopkts when using the `INDIR_MODE_FOLLOW_NOUSER` flag.

Sending a TIFF-F Fax File Within an Infopkt Stream

Transmitting a fax stored as a TIFF-F file is accomplished using the TIFF-F fax routines, as described earlier in this chapter, or using an infopkt of type `INFOPKT_INDIR_TIFF` within an infopkt stream.

TIFF files contain resolution and width parameters for each page. However, the same rules that apply to ordinary fax data streams also apply to fax data streams that contain `INDIR_TIFF` infopkts:

- A `DOCUMENT_PARAMETERS` infopkt must be the first infopkt in the stream, but the resolution specified by (the first page) of the TIFF file takes precedence over the resolution specified by the `DOCUMENT_PARAMETERS` infopkt. This rule is in effect only if `INDIR_TIFF` occurs right after the `DOCUMENT_PARAMETERS` or other infopkt types indicating beginning-of-page.
- To insert a page break between the last page of the first file and the first page of the second file, one of the infopkt types that indicate a beginning-of-page (see *Accessing an Infopkt Stream from an Application* on page 82) must be present between `INDIR_TIFF` infopkts.
- ASCII or G3 data are combined on a single page with the first or last page of a TIFF file by constructing the stream with no new-page type infopkt between the ASCII or G3 data and the `INDIR_TIFF` infopkt. G3 strip parameter packets might be

required (see *Infopkts on page 40*); the resolution and width of the G3 data in a TIFF file is always specified in the TIFF file, not in the G3 strip parameter packet.

Combining Data on a Single Page Using TIFF-F Fax Files

Two methods exist for combining data on a single page using TIFF-F files.

The first method uses an infopkt of type `INFOPKT_INDIR_TIFF` to embed a TIFF-F file in an infopkt stream, as described in *Sending a TIFF-F Fax File Within an Infopkt Stream on page 83*.

The second method uses the normal TIFF sending functions to transmit a TIFF file that is preceded or succeeded by other G3 or ASCII files that are transmitted with either the noninfopkt-formatted raw data fax functions or the TIFF file fax functions.

To use the second method successfully, the user must pay attention to the combine argument of the *BfvFaxNextPageTiff* function (see the function description in *Volume 4, Fax Processing*, in the *Dialogic® Brooktrout® Bfv API Reference Manual* for detailed information). When the combine argument is 1, the page information for the TIFF G3 data is sent to the channel without beginning a new page (do not call the *BfvFaxStripParams* function to do this).

For example, a program to send a page composed of ASCII combined with G3 from a TIFF file that is combined with G3 from a raw file could look like this:

```
/* Set up call prior */

/* Begin fax transmission, normal resolution, */
/* normal width */

BT_ZERO(args_fax);
args_fax.resolution = RES_200H_100V;
args_fax.width = WIDTH_A4;
BfvFaxBeginSendRaw(lp, &args_fax);
BfvFaxGetRemoteInfo(lp, &args_fax);
BfvFaxWaitForTraining(lp, &args_fax);
BT_ZERO(args_fax);
args_fax.fname = "ascii_file";
args_fax.fmt = DATA_ASCII;
BfvFaxSendFile(lp, &args_fax);
BT_ZERO(args_tiff);
args_tiff.fname = "tiff_file";
args_tiff.fmode = "r";
```

```

tp = BfvTiffOpen(&args_tiff);
/* No BfvFaxStripParams call is needed here, */
/* due to a combine value of 1 */

BT_ZERO(args_fax);
args_fax.s_tp = tp;
args_fax.combine = 1;
BfvFaxNextPageTiff(lp, &args_fax);
BT_ZERO(args_fax);
args_fax.s_tp = tp;
BfvFaxSendPageTiff(lp, &args_fax);
BT_ZERO(args_tiff);
args_tiff.tp = tp;
BfvTiffClose(&args_tiff);

```

```

/* A call to BfvFaxStripParams must be done */
/* here for combination with previous G3 */
/* data (from TIFF file)*/

```

```

BT_ZERO(args_strip);
args_strip.fmt = DATA_G3;
args_strip.resolution = RES_200H_100V;
args_strip.width = WIDTH_A4;
BfvFaxStripParams(lp, &args_strip);

BT_ZERO(args_fax);
args_fax.fname = "g3_file";
args_fax.fmt = DATA_G3;
BfvFaxSendFile(lp, &args_fax);
BfvFaxEndOfDocument(lp, &args_fax);

```

The previous example set the page resolution and width to fixed, predetermined values. A slight variation permits you to use the resolution and width values stored in the TIFF page for both the page and the data strip within the page. The TIFF routines retain these values until the data strip is sent.

```

/* set up call prior */

BT_ZERO(args_tiff);

args_tiff.fname = "tiff_file";
args_tiff.fmode = "r";
tp = BfvTiffOpen(&args_tiff);
BT_ZERO(args_fax);
args_fax.s_tp = tp;
BfvFaxBeginSendTiff(lp, &args_fax);
BfvFaxGetRemoteInfo(lp, &args_fax);
BfvFaxWaitForTraining(lp, &args_fax);
BT_ZERO(args_fax);
args_fax.fname = "ascii_file";
args_fax.fmt = DATA_ASCII;
BfvFaxSendFile(lp, &args_fax);
/* No BfvFaxStripParams call is needed here, */

```

```

/* due to a combine value of 1 */
BT_ZERO(args_fax);
args_fax.s_tp = tp;

args_fax.combine = 1;
BfvFaxNextPageTiff(lp, &args_fax);
BT_ZERO(args_fax);
args_fax.s_tp = tp;
BfvFaxSendPageTiff(lp, &args_fax);
BT_ZERO(args_tiff);
args_tiff.tp = tp;
BfvTiffClose(&args_tiff);

/* A call to BfvFaxStripParams must be done */
/* here for combination with previous G3 */
/* data (from TIFF file)*/

BT_ZERO(args_strip);
args_strip.fmt = DATA_G3;
args_strip.resolution = RES_200H_100V;
args_strip.width = WIDTH_A4;
BfvFaxStripParams(lp, &args_strip);
BT_ZERO(args_fax);
args_fax.fname = "g3_file";
args_fax.fmt = DATA_G3;
BfvFaxSendFile(lp, &args_fax);
BfvFaxEndOfDocument(lp, &args_fax);

```

Accessing a TIFF-F File from an Application

Although applications can directly read and write TIFF-F files with the Bfv API, some knowledge of TIFF-F file format is useful.

The *BfvTiffOpen* and *BfvTiffClose* functions open and close TIFF-F files, respectively.

The *BfvTiffReadIFD* and *BfvTiffReadImage* functions read an opened TIFF file. *BfvTiffReadIFD* calls a user-supplied function repeatedly with IFD entry information stored in an IFD (Image File Directory) for a particular page. The application can use *fseek* to move to locations in the TIFF file as directed by the tags (using the `TIFF_FP(tp)` macro to get the file pointer) and *BfvTiffReadRes* to help determine the resolution, as is often needed. The *BfvTiffReadImage* function puts data into a user-supplied buffer until the end of the page is reached.

A sample program to read a TIFF-F file follows:

```

int main (int argc, char **argv)
{
    int my_ifd_func();
    TIFF *tp;

```

```

        unsigned char buf[1024];
        int n;
        struct args_tiff args_tiff;

        BT_ZERO(args_tiff);
        args_tiff.fname = "filename";
        args_tiff.fmode = "r";
        tp = BfvTiffOpen(&args_tiff);
        for (;;)
        {
            args_tiff.tp = tp;
            args_tiff.func = my_ifd_func;
            args_tiff.arg = NULL;
            if (BfvTiffReadIFD(&args_tiff) <= 0)
                break;
            args_tiff.buf = buf;
            args_tiff.size = sizeof(buf);
            if ((n = BfvTiffReadImage(&args_tiff)) <= 0)
                break;
            process_image(buf,n);
        }

        args_tiff.tp = tp;
        BfvTiffClose(&args_tiff);

        return 0;
    }

    int my_ifd_func(TFILE *tp, struct ifd_field *ifd_field,
        char *arg)
    {
        /* Does nothing, just returns */
        return (0);
    }
}

```

The *BfvTiffWriteImage* and *BfvTiffWriteIFD* functions are used to write a new TIFF file. *BfvTiffWriteImage* receives data from a user-supplied buffer until the end of the page is reached. The *BfvTiffWriteIFD* function is called repeatedly with IFD entry information. The application can use *fseek* to move to locations in the TIFF file to determine the proper offsets to use for a given tag (using the `TIFF_FP(tp)` macro to get the file pointer) and *BfvTiffWriteRes* to help write the resolution as is often needed.

The presence of a certain set of tags is required to produce a valid TIFF-F file; *BfvTiffWriteIFD* automatically takes care of the `STRIPOFFSETS` and `STRIPBYTECOUNTS` tags, but the application is responsible for all other tags.

A sample program to write a TIFF-F file follows:

```
int main (int argc, char **argv)
```

```

{
    TFILE *tp;
    unsigned char buf[1024];
    int n;
    struct ifd_field ifd_field;
    struct args_tiff args_tiff;

    BT_ZERO(args_tiff);
    args_tiff.fname = "filename";
    args_tiff.fmode = "w";
    tp = BfvTiffOpen(&args_tiff);
    for (;;)
    {
        if ((n = get_image_data(buf)) <= 0)
            break;
        args_tiff.tp = tp;
        args_tiff.buf = buf;
        args_tiff.size = n;
        if (BfvTiffWriteImage(&args_tiff) != 0)
            break;
    }

    args_tiff.tp = tp;
    args_tiff.buf = NULL;
    args_tiff.size = 0;
    BfvTiffWriteImage(&args_tiff);
    args_tiff.ifd_field = &ifd_field;
    while (determine_next_ifd(&ifd_field) > 0 &&
        BfvTiffWriteIFD(&args_tiff) == 0);
    args_tiff.ifd_field = NULL;
    BfvTiffWriteIFD(&args_tiff);
    args_tiff.tp = tp;
    BfvTiffClose(&args_tiff);
}

```

Determining Fax Status Information from an Application

An in-progress fax transmission or reception has a number of attributes that an application might find useful to access. The application can use this information, for example, to update a fax status information screen.

The information available to an application includes:

- Remote Fax Node Parameters
 - ◆ Remote ID string
 - ◆ Remote NSF/NSC/NSS
 - ◆ Remote SSP/PWD/SUB
- Transfer Mode
 - ◆ Transmitting
 - ◆ Receiving

■ Current Page Transmission Parameters

- ◆ Bit rate
- ◆ Scan time
- ◆ ECM mode
- ◆ Compression format
- ◆ Resolution
- ◆ Width

This section explains how an application can access this information.

Remote Fax Node Parameters

The *BfvFaxGetRemoteInfo* returns the remote ID, remote NSF/NSC/NSS values, and remote SSP/PWD/SUB values to the application.

Transfer Mode

The application normally keeps track of its own operation mode (transmitting or receiving), but it can also use the `LINE_DCS` (see *Volume 4, Fax Processing*, in the *Dialogic® Brooktrout® Bfv API Reference Manual*) macro to get this information.

Current Page Transmission Parameters

The application can use the `LINE_DCS` macro to access information about the currently transmitting page; this information might change between pages. The information includes the bit rate, scan time, ECM mode, compression format, resolution, and width. The values for the first page are available after the *BfvFaxWaitForTraining* function has returned. Each time renegotiation or retraining occurs, the Bfv API updates the values available from the `LINE_DCS` macro.

3 - Debugging

This chapter describes how to use the debugging tools.

The chapter has the following sections:

- *[Bfv API Debug Mode on page 91](#)*
- *[BfvDataFSK on page 91](#)*
- *[BfvLineDumpStructure on page 91](#)*
- *[Dump History on page 92](#)*
- *[Parsed Command Information on page 97](#)*
- *[Utility Programs for Debugging on page 99](#)*
- *[BSMI Debugging on page 100](#)*

Bfv API Debug Mode

Some components of the Bfv API have their own unique debug functions to produce debug information relevant only to that component. The output from these functions is combined to provide a unified debug output if desired.

The following table lists the component, function name and reference:

By calling one of the *DebugModeSet* functions with an appropriate value, the application can cause the Bfv API to print various status and debugging information to the standard output. This output includes commands sent and received and state information. The application can change the debug mode output function to redirect the output to a file or nonstandard display by calling one of the *DebugFuncSet* functions.

The time-stamped output of the Bfv API debug mode is generated until another call to one of the *DebugModeSet* functions disables it.

BfvDataFSK

BfvDataFSK is issued at any point during the execution of an application to monitor the T.30 protocol procedure and to retrieve FSK data. Issuing *BfvDataFSK* and reviewing the FSK data after each function call is very useful as a debugging tool. See *Volume 4, Fax Processing*, in the *Dialogic® Brooktrout® Bfv API Reference Manual* for detailed information on *BfvDataFSK*.

BfvLineDumpStructure

The *BfvLineDumpStructure* function (*Volume 1*, in the *Dialogic® Brooktrout® Bfv API Reference Manual*) dumps the contents of the BTLINE structure into a file. It writes each element of the line structure individually.

Use this call to create error report logs (along with the contents of Dump History) and to track changing states of the line.

Dump History

Dump History (*dh*) is a stand-alone utility, which displays to the screen a log of the interactions between the driver and channels, modules, and applications. The size is specified at driver configuration time. The default is a buffer size of approximately 1MB for the entire driver.

To use the Dump History utility, you must enable the debug option during installation of the driver. See the *Dialogic® Brooktrout® Fax Products SDK Installation and Configuration Guide* section on *History Logging* for your operating system for instructions on how to enable the debug option. Once enabled, driver operations are automatically logged with a time stamp in a circular buffer.

Using dump history, in any form, can have an effect on the system. Enabling history at all, regardless of the configured size, slows the operation of the driver down to a small degree. If the history size is set to a very large value (e.g., a number of megabytes), then actually getting the history slows the system down in ways that affects any other Bfv API applications running.

This is because of several factors:

- The driver requires time to copy history memory. All other standard driver operations are temporarily suspended.
- The history functions require time to parse the history memory into readable form.
- High disk load while writing the many megabytes of history data in a short time.

Invoking Dump History

You can invoke Dump History from within an application or directly from the command line.

- From within an application use the *BfvHistoryDump...* functions. See *Volume 1* of the *Dialogic® Brooktrout® Bfv API Reference Manual* for detailed information on how to use the *BfvHistoryDump...* functions.
- As a stand-alone utility, as follows:

```
dh [-C] [-f] [-r [-b]] [-R file offset] [-P pktver] [-H hdr_dir] module channel
```

-C	Clear the history buffer, do not print entries.
-f	Display history output continuously until the program is terminated by the user. This option works best only on lightly loaded systems. For best speed, use the options -r -b, then later interpret the output using -R.
-r	Do a raw dump of the uninterpreted history data in ASCII form.
-b	Do dump as binary, for use with -r.
-R file offset	Indicates that history interpretation is to be done using a previously obtained binary history file. (Such files are obtained from system crash dumps or by processing the output of <code>dh -r -b</code>). A filename and starting offset in hex must be specified. When -R is used, the module and channel numbers are not required on the command line.
-P	For use with -R. Forces the packet version to the specified value (0 or 1).
-H hdr_dir	Read from the directory <code>hdr_dir</code> to create name tables. Used if modifications were made to commands or additional command header files are available after compilation and distribution of the program.
module 1 channel 1	Main driver history.
module 0	The most recent application corresponding to channel as an ordinal unit.
module FE	The fixed application history corresponding to channel as index.

Normally all history is collected in a single main history buffer accessible as module 1 channel 1. During driver configuration, you can choose a number of physical history buffers and application history buffers. If these are non-zero, additional history buffers are created, and appropriate items are logged in those buffers rather than the main log.

Under almost all conditions, the history should be configured for 0 physical histories and 0 application histories, and the only module/channel combination which should be dumped is module 1 and channel 1.

Dump History (*dh*) is supplied with the device driver and resides in *boston/driver/<platform>/user* on all systems.

The information provided by Dump History is very useful to Dialogic Technical Services and Support in identifying and solving problems.

Interpreting the Output

The output from Dump History consists of a status header line at the beginning followed by command logging lines. The output looks similar to this:

```
Hist Mod 1/Chan 1 Mon Jun 9 14:02:27 2003, Windows, V4700/B13/P5111/I25:
14:02:20.28.0000827C "PIPR: GDI_PHY_PIPR_MsgReceive (01 01 02 01)"
14:02:20.28.0000827D "PIPR: fnGenericPIPR_MsgReceive (01 01 02 01) NQ = 1 HQ = 0"
14:02:20.28.0000827E "PIPR: fnGenericPIPR_MsgReceive BufID = 518 (01 01 02 01)
    exit: norml msg"
14:02:20.28.0000827F "PIPR: GDI_PHY_PIPR_MsgReceive (01 01 02 01) exit:
    GDI_STS_OK"
14:02:20.28.00008280 "M2: Sending pkt len 000F prio 00 to (01 01 02 01)"
    Pkt bytes: 0D 00 04 01 01 02 01 02 02 FE 01 02 00 02 02
    L4 (01) ADMIN (02) FINISH (02) FIRMWARE_DOWNLOAD
14:02:20.28.00008281 "M2: Packet transferred successfully."
14:02:20.28.00008282 "UTL: GDI_UTL_BufferFree BufID = 518"
14:02:20.48.00008283 "PCI - Checking for intr/pkts"
14:02:20.48.00008284 "M2: Rcvd pkt len 0023, dest (01 01 01 01)"
14:02:20.48.00008285 "UTL: GDI_UTL_BufferAllocWait BufID = 519 all/user =
    240/172"
14:02:20.48.00008286 "M2: Queued incoming packet"
    Pkt bytes: 23 00 64 01 01 01 01 01 02 01 18 00 08 06 15
                0A 03 06 04 00 01 01 02 01 06 0B 02 00 20 03 00
                04 0C 01 C8 00
    L1A (01) ADMIN (08) EVENT (06) FLOW_CONTROL_STATUS
    (0A) CREDIT_INFO [03: Fix Uns List Unitless] LIST:
        (04) SUPPORTED_ADDRESS [00: Fix Uns Byte Unitless] 01 01 02 01
        (0B) FREE_BYTES [02: Fix Uns Long Unitless] 00032000
        (0C) FREE_PACKETS [01: Fix Uns Short Unitless] 00C8
14:02:20.48.00008287 "PIPR: GDI_PHY_PIPR_MsgSend"
14:02:20.48.00008288 "PIPR: fnPIPR_MsgSend BufID = 519 s(01 01 02 01) d(01 01 01
    01) NQ = 0 HQ = 0"
14:02:20.48.00008289 "PIPR: fnGenericPIPR_MsgReceive (01 01 01 01) NQ = 0 HQ = 1"
14:02:20.48.0000828A "PIPR: fnGenericPIPR_MsgReceive BufID = 519 (01 01 01 01)
    exit: high msg"
14:02:20.48.0000828B "UTL: GDI_UTL_BufferFree BufID = 519"
14:02:20.48.0000828C "PIPR: GDI_PHY_PIPR_MsgSend exit"
14:02:20.48.0000828D "PCI - Checking for packets to send"
14:02:20.52.0000828E "PCI - Checking for intr/pkts"
14:02:20.52.0000828F "M2: Rcvd pkt len 0017, dest (02 02 FE 01)"
14:02:20.52.00008290 "UTL: GDI_UTL_BufferAllocWait BufID = 51A all/user =
    240/172"
```

```

14:02:20.52.00008291 "M2: Queued incoming packet"
  Pkt bytes:      17 00 24 02 02 FE 01 01 01 02 01 0C 00 08 03 04
                  79 01 00 00 04 09 01 00 00
  LE (01) ADMIN (08) EVENT (03) FW_DOWNLOAD_FINISHED
    (79) CURRENT_STREAM [01: Fix Uns Short Unitless] 0000
    (09) DOWNLOAD_RESULT [01: Fix Uns Short Unitless] 0000
14:02:20.52.00008292 "PIPR: GDI_PHY_PIPR_MsgSend"
14:02:20.52.00008293 "PIPR: fnPIPR_MsgSend BufID = 51A s(01 01 02 01) d(02 02
  FE 01) NQ = 0 HQ = 0"
14:02:20.52.00008294 "PIPR: Dispatch (GDI_SIG_MSGPEND): Wakeup SlpHd = 88"
14:02:20.52.00008295 "PIPR: GDI_PHY_PIPR_MsgSend exit"
14:02:20.52.00008296 "PCI - Checking for packets to send"
14:02:20.52.00008297 "PIPR: GDI_APL_PIPR_MsgReceiveWait (02 02 FE 01) Awake
  SlpHd = 88 Rt = GDI_STS_OK"
14:02:20.52.00008298 "PIPR: GDI_APL_PIPR_MsgReceive (02 02 FE 01)"
14:02:20.52.00008299 "PIPR: fnGenericPIPR_MsgReceive (02 02 FE 01) NQ = 1 HQ = 0"
14:02:20.52.0000829A "PIPR: fnGenericPIPR_MsgReceive BufID = 51A (02 02 FE 01)
  exit: norml msg"
14:02:20.52.0000829B "PIPR: GDI_APL_PIPR_MsgReceive (02 02 FE 01)
  exit: GDI_STS_OK"
14:02:20.52.0000829C "PIPR: GDI_APL_PIPR_MsgReceiveWait exit: GDI_STS_OK"
14:02:20.52.0000829D "UTL: GDI_UTL_BufferFree BufID = 51A"
14:02:20.52.0000829E Ioctl ret #B0A
14:02:20.53.0000829F Ioctl MILL_SESSION_DESTROY #B0B

```

Status Header Line

Lists the module and channel number, the date and time the history was dumped, the platform name, the driver version, build number, PIPR version number (internal version), and ioctl interface version number (internal communication mechanism).

Event Logging Lines

Contain information about each packet being sent or received and other interactions taking place within the driver and between the driver and its applications.

Event Logging Line Format

The event logging lines report the time the event occurred and describe the particular event.

Timing Information

Timing information is reported first and usually takes the form:

`hr:min:sec.fracts.sequence`

For example: 11:12:25.512934.0000827C

The `sequence` field counts each event added to the history entries and is unique over all history buffers. Gaps in the sequence numbers occur when events occur in other history buffers. Sequence numbers are displayed in hex and wrap at 0xffffffff.

Event Descriptions

Each line gives information about some event that took place within the driver. If a packet is sent or received, a line specifies the direction. All or some of the packet data are displayed, and as much as possible is parsed and displayed.

Interactions between applications and the driver are usually via `ioctl` commands, which are shown starting with the word `ioctl`.

Parsed Command Information

In both Bfv API debug mode output and dump history output, commands appear parsed. In the history output, it appears after a message saying whether the packet was sent or received, the packet data, and a length value preceded by the letter 'L'. In Bfv API debug mode output, it appears after a '>' (greater than symbol) or '<' (less than symbol); indicating sent or received.

The commands in a packet appear one at a time (usually just one per packet). Indented under each command are all of its tagged values (often just called tags). List tags contain their own tags which are further indented. For example.

```
(01) ADMIN (08) EVENT (03) FW_DOWNLOAD_FINISHED
      (79) CURRENT_STREAM [01: Fix Uns Short Unitless] 0000
      (09) DOWNLOAD_RESULT [01: Fix Uns Short Unitless] 0000
```

First is the facility value in parentheses followed by the name of the facility (01 and ADMIN). Next is the command verb value followed by the name of the command verb (08 and EVENT). Last is the command specifier value followed by the name of the command specifier (03 and FW_DOWNLOAD_FINISHED).

The example command contains 2 tags directly within it. Looking at the first one, the line describing a tag starts with the tag ID value and tag name (79 and CURRENT_STREAM). Next, in brackets, is the tag type value (01) and its meaning (fixed unsigned short unitless). At the end of the tag it either says LIST: if the type is a list, give the value of the tag if it is a simple integer value (0000 in this case), give a string value if it is an array of chars, or give a list of numbers if it is an array of other integers.

Utility Programs for Debugging

The following sample applications/utilities are available to help you in debugging your applications by giving you information about modules in the system such as the firmware, driver, connections, etc.

btver

btver gives you version information for the driver, Bfv API, boot ROM firmware, control processor firmware, and DSP firmware. See *btver* on [page 122](#) for more information.

connlist

To find out the currently established call switching connections, use the *connlist* program, which is described in detail in *connlist* on [page 123](#).

feature -q

With the *-q* option of the *feature* program, you can query the set of features loaded on your module. *feature* is described in *feature* on [page 139](#).

modinfo

To find out about the active hardware and software modules on your system as seen by the driver, use the *modinfo* program, which is described in *modinfo* on [page 149](#).

shoparam

shoparam is a stand-alone utility that displays the contents of the line structure and all of the parameter values contained in the user and read-only configuration files. See *shoparam* on [page 151](#) for more detailed information.

BSMI Debugging

The debug output is controlled using the regular mechanisms provided with the Bfv API. Included in the debug output are diagnostic strings including BSMI message tracing and network layer tracing.

BSMI Message Tracing

vttty

The `vttty` program displays layer 2, 3, and 4 messages (depending on user settings).

Command Syntax

```
vttty [-m <mod>] {-v}
```

Arguments

-m <mod>	Use specified module (default 2)
-v	Turn on Bfv API debugging mode

Included in the debug output is a diagnostic message showing the message identifier of all messages sent to the firmware from the application, and all messages sent from the firmware to the application.

Running a Layer 2 Trace

The diagnostic trace function allows you to trace Layer 2 messages entering and leaving the framer (See [Table 5](#) on [page 110](#) for command line syntax). The trace function displays link layer protocol messages only, such as ISDN Q.931. The trace display resembles a simple protocol analyzer, with the message type decoded and its direction shown.

A trace shows Layer 2 messages being passed over the links, and provides some protocol and routing information. It also displays the received/transmitted message Information Frame in hexadecimal format. This hexadecimal string contains Layer 2 ISDN frame headers.

The trace information is embedded within the Bfv API debug output, see [Figure 5](#) for a sample output. See [Table 2](#) for report heading information:

Ch#	Time	Direct	SAPI	TEI	C/R	Type	N(s)	N(r)	P/F	Size
03	23B4	Xmit	3F	7F	0	teiReqst			0	0008
		FCFF030F23B501FF								
03	23BB	Rcvd	3F	7F	1	teiAssgn			0	0008
		FEFF030F23B502C1								
03	23BB	Xmit	00	60	0	SABME			1	0003
03	23C1	Rcvd	00	60	0	UA			1	0003
03	278D	Xmit	00	60	0	Setup	00	00	0	0018
		00C1000008010105040288901801812C0735353532303030								
03	27C2	Rcvd	00	60	1	Prdng	00	01	0	000B
		02C1000208018102180189								
03	27C2	Xmit	00	60	1	RR		01	0	0004
03	27DD	Rcvd	00	60	1	Alrtng	01	01	0	0008
		02C1020208018101								
03	27DD	Xmit	00	60	1	RR		02	0	0004
03	27FB	Rcvd	00	60	1	Connct	02	01	0	0008
		02C1040208018107								
03	27FB	Xmit	00	60	0	ConAck	01	03	0	0008
		00C102060801010F								
03	280C	Rcvd	00	60	0	RR		02	0	0004
03	2E43	Xmit	00	60	0	Discct	02	03	0	000C
		00C104060801014508028090								
03	2E5E	Rcvd	00	60	1	Rlease	03	03	0	0008
		02C106060801814D								
03	2E5E	Xmit	00	60	0	RelCom	03	04	0	0008
		00C106080801015A								
03	2E71	Rcvd	00	60	0	RR		04	0	0004
00	09F3	Rcvd	00	00	1	Prdng	00	01	0	000E
		0201000208028001021803A98381								

Figure 5. Level 2 Trace Example

Table 2. Trace Report Values

Value	Meaning
Ch#	Lapdid number. This is an even number where 0=Port A 2=Port B
Time	Hexadecimal timestamp incremented at 1 ms intervals.
Direct	Direction of frame; possible values are Xmit (transmitted by module) and Rcvd (received by module).
SAPI	Service Access Point Identifier which identifies the type of D-channel signaling performed; possible values are 00 (ISDN call control) or 63 (management procedures).
TEI	Terminal Endpoint Identifier that identifies a particular endpoint device.
C/R	Command/Response bit that identifies the frame as either a command (C) or response (R); possible values vary depending on whether the module is performing user side or network side signaling. For user side, 0 indicates a command and 1 indicates a response. For network side, 0 indicates a response and 1 indicates a command.
Type	Q.921 UNKNI message frame for unknown information frames or the Q.931 message contained in the I (information) frame.
N(s)	Sequence number assigned to the frame sent by the transmitting device.
N(r)	Expected sequence number of the next frame to be received from the transmitting device.
P/F	Poll/final bit which indicates the device is polling for a response from the other end, sending a final frame in response to a command, or neither. Possible values are 1 (polling for response or responding to command) or 0 (not polling or unsolicited response). If the message frame is a command (based on the C/R bit), this is a Poll bit; if the message frame is a response, this is a Final bit.
Size	Number of bytes in frame (shown in hexadecimal).

Understanding Trace Hexadecimal Strings

The hexadecimal string displayed in the trace consists of the following components:

- Information (I) Frame header

Note: A trace displays hexadecimal strings for I Frame messages only. Supervisory (S Frame) messages, such as Receiver Ready (RR), and Unnumbered (U Frame) messages, such as SABME and UA, are not displayed in hexadecimal format.

- Message header
- Information Elements (IEs)

Interpreting the I Frame Header

The I Frame header contains Layer 2 routing and packet transaction information. The first four bytes of the hexadecimal string comprise the I Frame header.

```
03 278D  Xmit  00  60  0  Setup  00  00  0  0018
00C10000 08010105040288901801812C0735353532303030
└──────────┘
      I Frame
      Header
```

Figure 6 compares the general format for an I Frame against the I Frame for an example SETUP message, and illustrates the following points:

- The trace automatically removes the 0x7E byte flags (binary 01111110) that normally indicate the start and end of the frame and the Frame Check Sequence (FCS) values.
- A trace automatically interprets and displays I Frame header elements such as the SAPI and TEI (refer to [Table 2](#) for descriptions of these elements).
- A value of 0 in the shaded bit position identifies the frame as an I Frame.

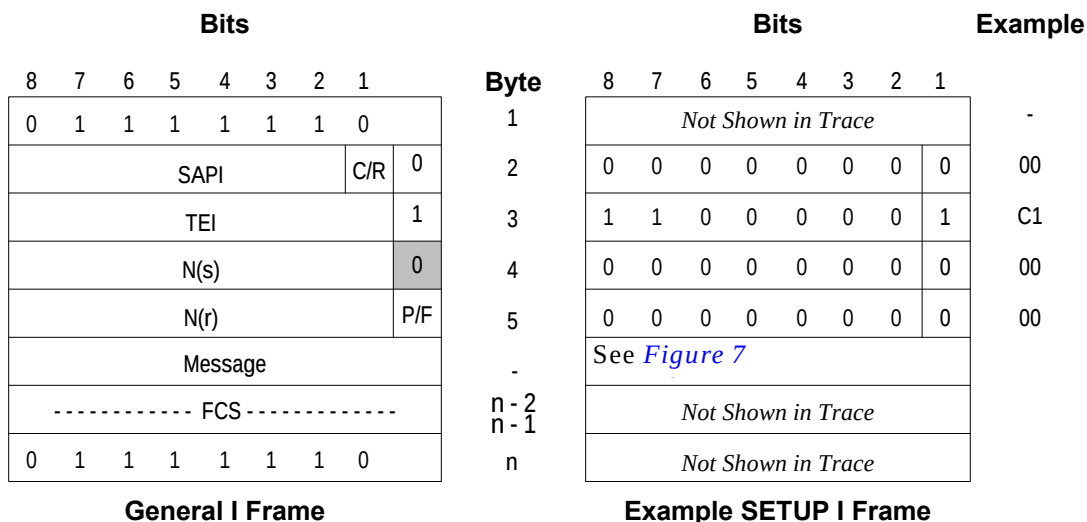
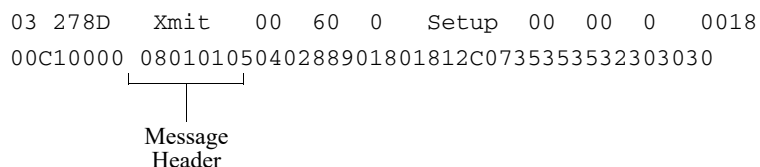


Figure 6. I Frame Formats

Interpreting the Message Header

The Message header starts at byte offset 5 of the hexadecimal string. For Q.931 call control messages, this header identifies the D-channel message and references the call for which the message applies.



[Figure 7](#) compares the general structure for a Message header against the example SETUP message header, and illustrates the following points:

- The Protocol Discriminator value is 0x08 for all Q.931 call control messages.
- Modules assign 1-byte call reference values for Q.931 messages, so the call reference length is always 0x01 and the third byte in the Message header contains the call reference value (0x0001 in the example).

- The Message type value 0x05 identifies the D-channel message as a SETUP; refer to [Table 3](#) on [page 105](#) for possible Q.931 message type values.

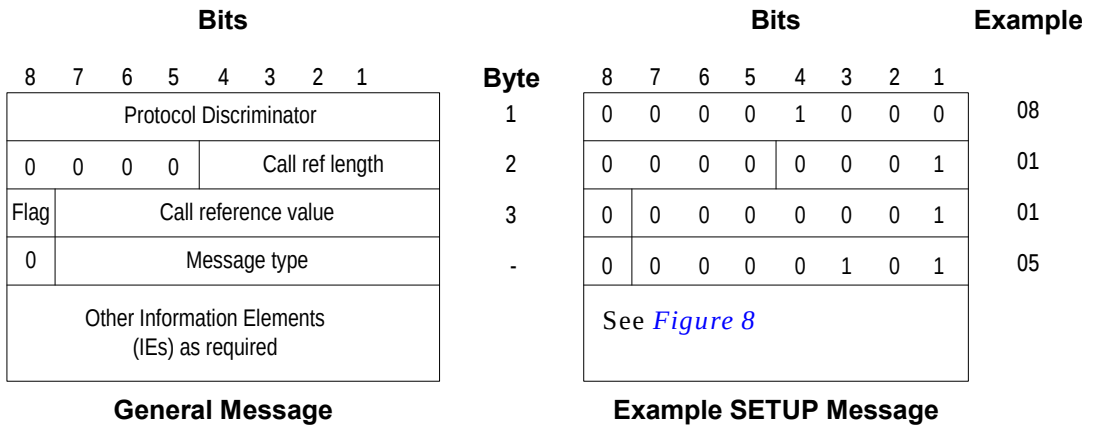


Figure 7. Message Structures

Table 3. Q.931 Message Types

Message Type Bits	Hex	Message
0 0 0 0 0 0 0 1	01	Alerting
0 0 0 0 0 0 1 0	02	Call Proceeding
0 0 0 0 0 1 1 1	07	Connect
0 0 0 0 1 1 1 1	0F	Connect Acknowledge
0 0 0 0 0 0 1 1	03	Progress
0 0 0 0 0 1 0 1	05	Setup
0 0 0 0 1 1 0 1	0D	Setup Acknowledge
0 1 0 0 0 1 0 1	45	Disconnect
0 1 0 0 1 1 0 1	4D	Release
0 1 0 1 1 0 1 0	5A	Release Complete
0 1 0 0 0 1 1 0	46	Restart
0 1 0 0 1 1 1 0	4E	Restart Acknowledge

Table 3. Q.931 Message Types (Continued)

Message Type Bits	Hex	Message
0 1 1 1 1 0 1 1	7B	Information
0 1 1 0 1 1 1 0	6E	Notify
0 1 1 1 1 1 0 1	7D	Status
0 1 1 1 0 1 0 1	75	Status Enquiry

Interpreting Information Elements

For Q.931 call control messages, the first Information Element (IE) starts at byte offset 9 in the hexadecimal string. Each message can contain several IEs of either fixed (single byte) or variable length.

```

03 278D  Xmit  00 60 0  Setup  00 00 0 0018
00C10000080101050 4028890 180181 2C0735353532303030

```

Figure 8 on [page 107](#) compares the general IE format against the first IE contained in the example SETUP message, and illustrates the following points:

- A value of 0 in the shaded bit position indicates a variable-length IE; a value of 1 in that position indicates a single byte IE.

Note: Single byte IEs are commonly used for locking codeset shifts. Locking shift IEs appear only after all variable-length IEs within the message. Refer to the Bellcore Technical Reference TR-TSY-000268 for more information on the structure and use of single byte IEs and codeset shifts.

- The IE identifier value 0x04 indicates a Bearer Capability IE; refer to [Table 4](#) on [page 107](#) for possible IE identifier values. IEs appear in messages in ascending order according to their identifier number.
- The 2-byte length of the IE value indicates that it contains only the required structures for a Bearer Capability IE.
- The IE contents indicate an information transfer capability of unrestricted digital information (0x88) and a transfer rate/mode equal to 64 kbps/circuit mode (0x90).

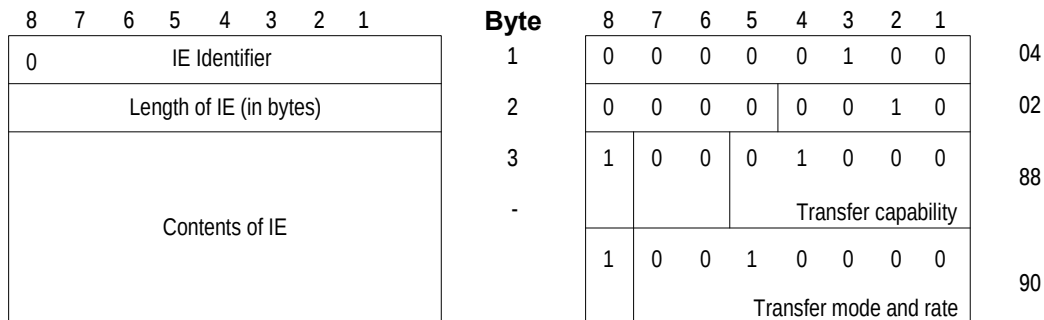


Figure 8. IE Formats

Table 4. Q.931 Information Element Identifiers

IE Identifier Bits	Hex	Information Element
0 0 0 0 0 1 0 0	04	Bearer capability
0 0 0 0 1 0 0 0	08	Cause
0 0 0 1 0 1 0 0	14	Call state
0 0 0 1 1 0 0 0	18	Channel identification
0 0 0 1 1 1 1 0	1E	Progress indicator
0 0 1 0 1 1 0 0	2C	Keypad
0 0 1 1 0 1 0 0	34	Signal
0 1 0 0 0 0 0 0	40	Information rate
0 1 0 0 0 0 1 0	42	End-to-end transit delay
0 1 0 0 0 0 1 1	43	Transit delay selection and indication
0 1 0 0 0 1 0 0	44	Packet-layer binary parameters
0 1 0 0 0 1 0 1	45	Packet-layer window size
0 1 0 0 0 1 1 0	46	Packet size
0 1 1 0 1 1 0 0	6C	Calling party number
0 1 1 0 1 1 0 1	6D	Calling party subaddress

Table 4. Q.931 Information Element Identifiers (Continued)

IE Identifier Bits	Hex	Information Element
0 1 1 1 0 0 0 0	70	Called party number
0 1 1 1 0 0 0 1	71	Called party subaddress
0 1 1 1 1 0 0 0	78	Transit network selection
0 1 1 1 1 0 0 1	79	Restart indicator
0 1 1 1 1 1 0 0	7C	Low-layer compatibility
0 1 1 1 1 1 0 1	7D	High-layer compatibility

For additional information on Layer 2 and ISDN message headers and processing, refer to the following documents:

- CCITT Recommendation I.441
- Bellcore Technical References TR-TSY-000268 and TR-TSY-000793

VTTY Tracing Feature

The VTTY tracing provides access to control processor internals and diagnostic tracing information. Two VTTY applications are provided:

- *VTTY Console Commands on page 110*
- *VTTY Tracer GUI on page 111* (Windows operating systems only)

These applications allow users to enter commands that enable tracing capabilities or retrieve any saved information blocks.

Each trace message is prefixed with a time stamp in the same format used for the call control trace files and the Bfv API application debug log files.

The time reference for this time stamp is the local host time.

VTTY Console Commands

Specifically, you can use the VTTY commands to show argument information, tracing information, access memory locations and get help. [Table 5](#) shows the command syntax and describes the action that is performed. (See [Table 2](#) on [page 102](#) for a definition of “lapdid”.

Table 5. VTTY Commands

Command	Meaning
> help > ?	Returns a menu of supported commands with syntax. Same as help.
The following commands are show commands that return information specific to the arguments.	
> s ?	Returns a menu of supported show command arguments with syntax.
> s tmr	Returns a display of ISDN timer structures.
> s lap <lapdid#>	Returns a display of ISDN lap structures for provided <i>lapdid</i> .
> s <lapdid#>	Returns a display of Layer 2 statistics for the provided <i>lapdid</i> .
> s q931 <lapdid#>	Returns a display of Q931 information for the provided <i>lapdid</i> .
> s pump <lapdid#>	Returns a display of pump channel information for the provided <i>lapdid</i> .
The following commands are trace commands that enable/disable tracing information specific to the arguments.	
> t ?	Returns a menu of supported trace command arguments with syntax.
> t	Toggles the tracing of Bfv API messages between layers 3 and 4.
> t smi	Same as t, toggles the tracing of Bfv API messages between layers 3 and 4.
> L <lapdid#> <level>	Enables Layer 2 tracing for level=1 or disables tracing for level=0 on the provided <i>lapdid</i> .
> l <level>	Enables Layer 2 tracing for level=1 or disables tracing for level=0 on the provided <i>lapdid</i> .
> ww <address> <value>	Writes the 16-bit form of the value provided to the address provided.
> wl <address> <value>	Writes the 32-bit form of the value provided to the address provided.
> dump <address> <length>	Returns a display of the memory contents starting at the provided address.

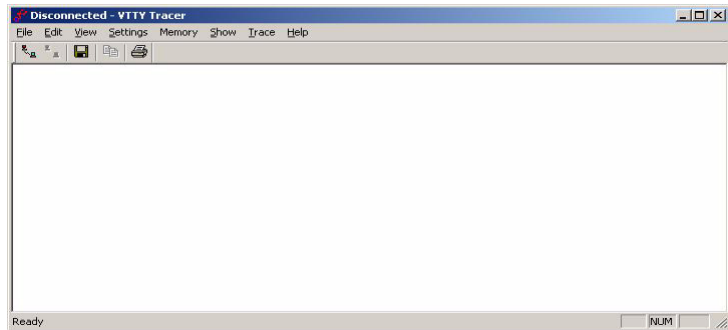
VTTY Tracer GUI

The VTTY Tracer graphical user interface provides access to control processor internals and diagnostic tracing information from a Windows environment.

- **To start the VTTY Tracer, enter:**

```
vttty_tracer.exe
```

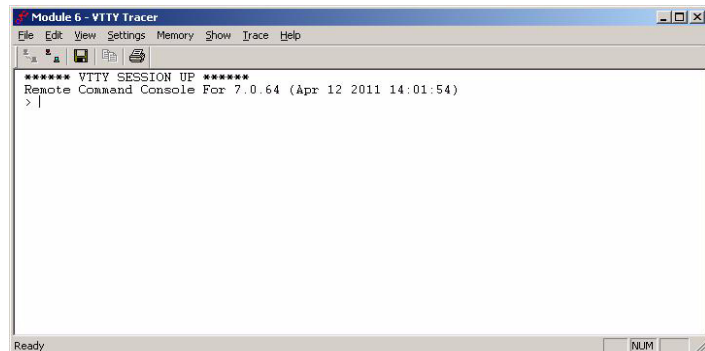
The *VTTY Tracer* screen is displayed.



Setting Output Options

From the main screen, you can change the tracer output options.

1. Click *Settings | Options*. The *Tracing Options* dialog is displayed.



Set where the trace output is saved, either screen or file. The maximum file size is set in MBytes. Once a file reaches its maximum size, the tracing output loops back to the beginning of the file. The user always has the number of MBytes of information entered.

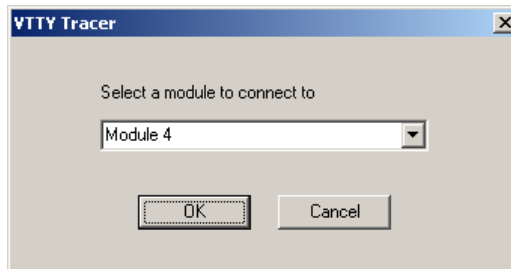
You can change these options any time during execution (while tracing or before connecting to a module).

2. Click *OK* to save your tracing options.

Connecting to a Module

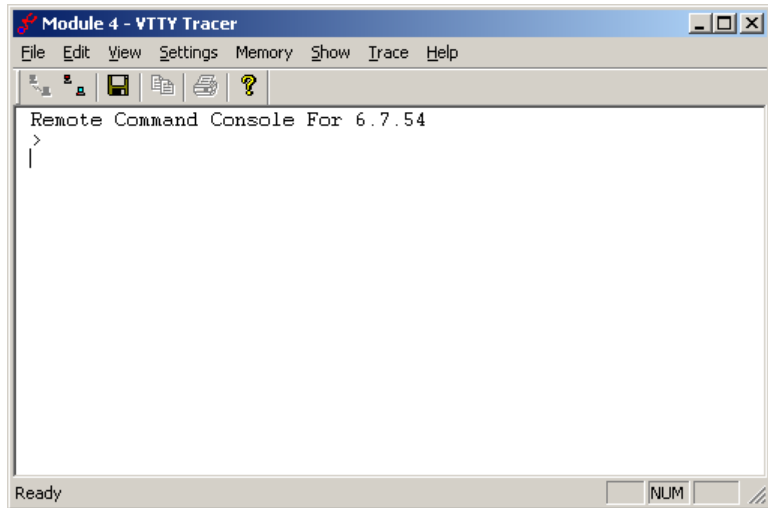
➤ Use the File menu to connect to a module:

1. Click *File*, *Connect to Module* or click on the button  to connect to the module. The *VTTY Tracer* dialog is displayed.



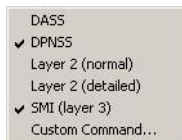
2. Choose an available module from the drop down box. The *Select a module to connect to* drop down box contains a list of all modules available on the system that are traced.

-
- Click **OK**. The connection confirmation is displayed in the VTTY Tracer screen:



Using the Trace Menu

When you choose options from the *Trace* menu, multiple items are selected. When complete the selected items are checked, for example:



When *Trace | Custom Command* is selected, the Custom Command dialog is displayed:



Use the Custom Command only under the guidance of Dialogic Technical Services and Support.

Using the Memory Menu

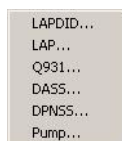
Click the *Memory* menu to read and write predefined sizes of memory.



Using the **Memory** menu requires extreme care. Reading or writing memory to the module can cause irreparable harm. This menu should never be used unless explicitly directed to do so by Dialogic Technical Services and Support. See [Getting Technical Support](#) on [page 19](#).

Using the Show Menu

The *Show* menu causes a set of predefined variables to be displayed:

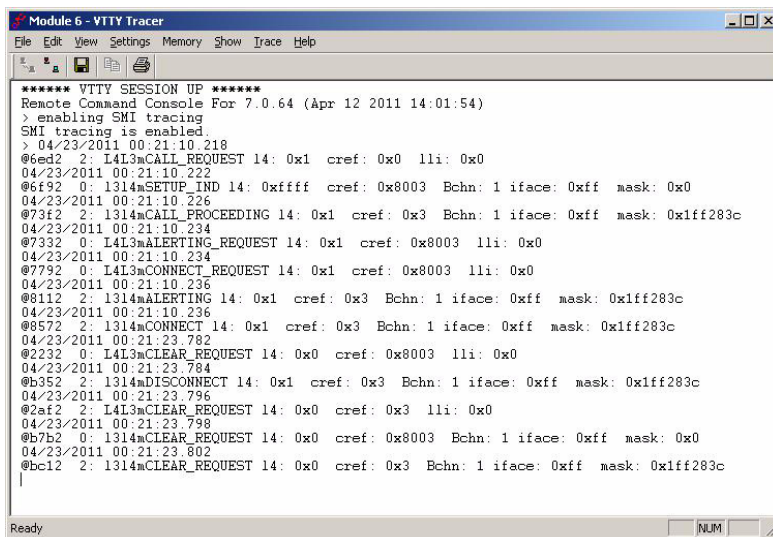


Show Menu Options	Information About
LAPDID	The specified D-channel
LAP	The LAP-D (Q.921) protocol
Q931	The Q.931 ISDN call control protocol
DASS	The DASS-2 call control protocol
DPNSS	The DPNSS-1 call control protocol
Pump	Internal operations

Each menu choice produces a dialog box where another selection is made from a drop down box. Once selected, output is displayed in the main trace screen. See [Figure 9](#).

VTTY Trace Results

Figure 9 shows the results of a VTTY trace.



```
Module 6 - VTTY Tracer
File Edit View Settings Memory Show Trace Help

***** VTTY SESSION UP *****
Remote Command Console For 7.0.64 (Apr 12 2011 14:01:54)
> enabling SMI tracing
SMI tracing is enabled.
> 04/23/2011 00:21:10.218
@6ed2 2: I4L3mCALL_REQUEST 14: 0x1 cref: 0x0 lli: 0x0
04/23/2011 00:21:10.222
@6f92 0: I314mSETUP_IND 14: 0xffff cref: 0x8003 Bchn: 1 iface: 0xff mask: 0x0
04/23/2011 00:21:10.226
@73f2 2: I314mCALL_PROCEEDING 14: 0x1 cref: 0x3 Bchn: 1 iface: 0xff mask: 0x1ff283c
04/23/2011 00:21:10.234
@7332 0: I4L3mALERTING_REQUEST 14: 0x1 cref: 0x8003 lli: 0x0
04/23/2011 00:21:10.234
@7792 0: I4L3mCONNECT_REQUEST 14: 0x1 cref: 0x8003 lli: 0x0
04/23/2011 00:21:10.236
@8112 2: I314mALERTING 14: 0x1 cref: 0x3 Bchn: 1 iface: 0xff mask: 0x1ff283c
04/23/2011 00:21:10.236
@8572 2: I314mCONNECT 14: 0x1 cref: 0x3 Bchn: 1 iface: 0xff mask: 0x1ff283c
04/23/2011 00:21:23.782
@2232 0: I4L3mCLEAR_REQUEST 14: 0x0 cref: 0x8003 lli: 0x0
04/23/2011 00:21:23.784
@b352 2: I314mDISCONNECT 14: 0x1 cref: 0x3 Bchn: 1 iface: 0xff mask: 0x1ff283c
04/23/2011 00:21:23.796
@2af2 2: I4L3mCLEAR_REQUEST 14: 0x0 cref: 0x3 lli: 0x0
04/23/2011 00:21:23.798
@b7b2 0: I314mCLEAR_REQUEST 14: 0x0 cref: 0x8003 Bchn: 1 iface: 0xff mask: 0x0
04/23/2011 00:21:23.802
@bc12 2: I314mCLEAR_REQUEST 14: 0x0 cref: 0x3 Bchn: 1 iface: 0xff mask: 0x1ff283c

Ready NUM
```

Figure 9. VTTY Trace Example

Each trace message (for both screen and trace file output) is prefixed with a time stamp in the same format used for the call control trace files and the Bfv API application debug log files.

The time reference for this time stamp is the local host time.

Call Tracer

Dialogic provides a Call Tracer command line utility that collects call trace information in an active system. The output is intended for Dialogic Technical Services and Support, but it is important that all users know how to use Call Tracer to create the output file, if Dialogic Technical Services and Support personnel request it. The Call Tracer utility can be started before or after starting the client application. If you want to trace the initialization section of the client application, start the Call Tracer before the client application.

For information on how to start the Call Tracer, type *brktcctrace - ?*. Exit the Call Tracer application by typing 'q' or 'Control-C', or by closing the command console window. The Call Tracer application reads trace filter settings from a text configuration file called *filtersettings.cfg*. The output is logged to a file name of your choosing.

For log information internal to the Call Tracer, the application maintains its own log file that is located in the current working directory of the application. The tracer logs all warning, error and panic level messages by default.

The Call Tracer utility, and a sample configuration file, can be found in the *\Brooktrout\Boston\utils\winnt\bin* directory when installing the Brooktrout SDK, or in the *\Brooktrout\bin* directory when installing just the Dialogic® Brooktrout® Runtime Software.

Note: Any paths in the command line argument that contain spaces should be added between double quotes.

The Brooktrout Tracer opens ports 4010 and 4020 - 4024 by default. BFV applications such as modinfo, features, and btver may be blocked by a Firewall. On Windows, a Firewall dialog box will pop-up informing the user that the application is attempting to open a port.

➤ **To disable the Firewall dialog box perform one of the following:**

- Open ports 4010 and 4020 to 4024: Firewall configuration
- Allow the applications using BFV to open the ports: Firewall configuration
- On Windows, disable the Brooktrout Tracer by setting "Number of Client Ports" to 1 on the Brooktrout Configuration Tool Advanced Mode in the menu Options, Tracer Connection Settings. This option is not desired since it will prevent tracing BFV application at runtime

Command Syntax

```
brktcctrace [-o <path_string>] [-i <path_string>] [-t  
<duration/sec>] [-x <max_file_size>] [-n <max_num_files>]  
[-h]
```

Arguments

-o <path_string>	Fully qualified path of the output file including the log file name. example: c:\Brooktrout\brktlog.txt. The path containing spaces should be added between double quotes. This is a mandatory parameter. The Brooktrout Tracer will create an individual file for each process using the Brooktrout API and append the client port number to the end of the log file name. In the example above the log files would be named c:\Brooktrout\brktlog_xxxx.txt.
-i <path_string>	Path of the input filter configuration file including the file name. Path containing spaces should be added between double quotes.
-t <duration/sec>	Time duration for the Call Tracer to stay connected to the clients in seconds (default zero-infinite).
-x <max_file_size>	Maximum size of the log file in Megabytes (default 1000).
-n <max_num_files>	Maximum number of log files to create (default 1).
-h	Help

An example of using *brktcctrace* to create a call trace file:

```
brktcctrace -x 10 -n 5 -o c:\Brooktrout\brktlog.txt
```

In this case, the logging application creates the brktlog_xxxx.txt file where xxxx represents the client port number. Since a configuration file was not specified, the logging process enables the default log settings. While the logging process continues, logs are always written to the specified file until the log file size reaches 10MB then the log file is cleared and relogging starts all over again and saves the log file for example, as brktlog_xxxx.txt.1, brktlog_xxxx.txt.2, brktlog_xxxx.txt.3, etc.

Logging stops after the fifth log file has been created. When the logging application records the fifth logging file, it then starts overwriting the first log file for example, *brktlog_xxxx.txt*.

Configuration File Format

This section describes the filter configuration file format. Because the configuration file is optional, there is no default.

Lines that start with a '#' character are comments only.

The filter settings are not case sensitive. All the settings are printed in upper case for uniformity. If the configuration file has multiple entries for the same filter settings, the first setting is used.

For example: If the filter settings configuration file contains the following:

```
BFV.BFV.DEBUG_PRINT_CMD    = OFF
BFV.BFV.DEBUG_PRINT_INTR   = OFF
BFV.BFV.DEBUG_MON           = OFF
BFV.BFV.DEBUG_ERR           = OFF
BFV.BFV.DEBUG_PRINT_CMD     = ON
```

then BFV.BFV.DEBUG_PRINT_CMD=OFF setting is used.

The following configuration file is an example of what Dialogic Technical Services and Support would request.

```
#####
# Filter Settings Configuration File
#####
DEFAULT_CONFIG = AUTO
#-----
# BFV Valid Values:
# ON
# OFF
# Default: ON, except for DEBUG_ENTRY_EXIT_ARGS
#-----
BFV.BFV.DEBUG_PRINT_CMD           = ON
BFV.BFV.DEBUG_PRINT_INTR          = ON
BFV.BFV.DEBUG_MON                  = ON
BFV.BFV.DEBUG_ERR                  = ON
BFV.BFV.DEBUG_DEBUG                = ON
BFV.BFV.DEBUG_ACCULIB              = ON
BFV.BFV.DEBUG_ENTRY_EXIT           = ON
BFV.BFV.DEBUG_ENTRY_EXIT_ARGS     = ON
#-----
# ECC Valid Values:
# NONE
# ERROR
# WARNING
# BASIC
# VERBOSE
# Default: VERBOSE
#-----
ECC.ECC_API                        = VERBOSE
ECC.ECC_HOST_MODULE                = VERBOSE
ECC.ECC_INTERNAL                   = VERBOSE
ECC.ECC_IP_STACK                   = VERBOSE
ECC.ECC_L3L4                       = VERBOSE
ECC.ECC_L4L3                       = VERBOSE
```

4 - Sample Applications and Utilities

This chapter describes the sample applications and utilities that come as part of the Dialogic® Brooktrout® SDK.

Dialogic includes a large collection of sample application programs and utilities with the Bfv API.

Sources for the sample applications are located in either *boston/bfv.api/app.src* or *boston/bfv.api/bapp.src*, except for a few applications whose directories are stated in the text. The application makefiles appear and compilation is performed in either *boston/bfv.api/<platform>/app.src* or *boston/bfv.api/<platform>/bapp.src* (where *<platform>* represents the name of the operating system in use). Many of the *bapp.src* program executables are also distributed in this directory.

The chapter has the following sections:

- An alphabetical list of all the sample applications
- [Compiling Sample Applications Using Microsoft Developer Studio Project Files on page 165](#)
- [Compiling Sample Applications Using Makefiles on page 167](#)

Note: Source code to utilities such as *vttty* are provided for convenience to allow application re-building if desired. Sample applications are provided to demonstrate how to use Bfv API functionality.

boardmon

The *boardmon* program monitors the condition of a module. It provide Ethernet link status (as determined by *BfvBoardNotify*) when monitoring a board with an enabled Ethernet interface. It displays the module temperature, the status of the Ethernet port and monitors one or more telephony spans on the module. It reports the state of the signaling bits and alarms. It also counts the errors (e.g., framing errors, CRC errors, clock slips, BPVs) on each span. It only works with spans that are configured for robbed-bit signaling and where telephony connections have been made.

The spans are numbered starting from 1 which is the first interface on a module.

The *btcall.cfg* file is the user-defined configuration file. The *boardmon* application is found in the *bapp.src* directory.

Command Syntax

```
boardmon [-m <mod>] [-s <span>] [-d] [-v] [-h]
```

Arguments

-m <mod>	Use specified module (default 2)
-s 	Use specified span only [1-4] (default all)
-d	Enable program debug mode
-v	Enable Bfv API debug mode
-h	Help

While the program is running you can press a key to reset the error counters or to quit. When running under Linux, you must press *Enter* after pressing the key.

1	Reset error counters for span 1
2	Reset error counters for span 2
3	Reset error counters for span 3
4	Reset error counters for span 4
r or R	Reset error counters for all spans
q or Q	Quit the program

Sample *boardmon* Output

Board Temperature: 105.8F 41.0C

Ethernet link 0: UP

Mod 0x03 Span 1 No Alarms FRM 000000 CRC 000000 BPV 000000 SLIP 000000

Ch: 1 5 9 13 17 21 Last reset: 11/30 14:42:11

In: a.aa .aaa aaaa .aaa aa.a aaa.
b.bb .bbb bbbb .bbb bb.b bbb.
c.cc .ccc cccc .ccc cc.c ccc.
d.dd .ddd dddd .ddd dd.d ddd.

Out: A.AA .AAA AAAA .AAA AA.A AAA.
B.BB .BBB BBBB .BBB BB.B BBB.
C.CC .CCC CCCC .CCC CC.C CCC.
D.DD .DDD DDDD .DDD DD.D DDD.

Mod 0x03 Span 2 LOS FRM 000002 CRC 000000 BPV 000000 SLIP 000001

Ch: 1 5 9 13 17 21 Last reset: 11/30 14:42:11

In: aa..
bb..
cc..
dd..

Out: AA..
BB..
CC..
DD..

btver

The *btver* program displays version information for the driver, Bfv API, boot ROM firmware, control processor firmware, and DSP firmware.

The *btcall.cfg* file is the user-defined configuration file. The *btver* program is found in the *bapp.src* directory. The Bfv API debug mode is turned on.

Command Syntax

btver [-m <mod>] [-v]

Arguments

- m <mod> Display version info for specified module only.
 Otherwise display for all.
- v Turn on Bfv API debug mode.

The firmware must be downloaded on a module to obtain its control processor and DSP information.

connlist

The *connlist* program lists currently established call switching connections. Full-duplex connections are always reported as a pair of simplex TRANSMIT-ONLY connections, where the source and destination endpoints are swapped for the two halves of the full-duplex connection.

The Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *connlist* program is found in the *bapp.src* directory.

Command Syntax

connlist [-m <mod>] [-v]

Arguments

- m <mod> Module number for connections (default 2).
- v Turn on Bfv API debug mode.

A receive connection between two given points is identical to a transmit connection between those same points with the source and destination interchanged.

csend

This application uses low-level, noninfopkt, raw-data, fax-sending routines to send facsimiles. It allows sending a single fax page either in standard fax format from an MH/MSB G3 data file or in an enhanced fax format (e.g. JPEG, JBIG) from an EFF data file. T.30 holdup is used to check on the receiver's capabilities before deciding which type of file to send.

The Bfv API debug mode is turned on. The *btcall.cfg* file is the configuration file. The *csend* program is found in the *app.src* directory.

Command Syntax

```
csend [-u <unitno>] [-v] [-c <EFF_file>] [-g <G3 file>]  
[-e <EFF_OPT>] <phone number>
```

Arguments

- c <EFF_file> Color or other enhanced fax file, default: *c1.jpg*
- e <EFF_opt> Enhanced Fax Format (EFF) options, default: 3
OR together the following hex values:
 - 1 = JPEG Enable
 - 2 = Full Color
 - 4 = Default tables
 - 8 = 12-bit (vs. 8-bit)
 - 10 = No Subsampling
 - 20 = Custom Illuminant
 - 40 = Custom Gamut
 - 100 = JBIG
 - 200 = JBIG L0
 - 400 = Lossless Color
 - 800 = MRC
 - 1000 = MRC
 - 2000 = MRC
 - 4000 = Plane Interleave
 - 8000 = Page Length Strip
- g <G3 file> Black and white MH file, default: *eagle.301*
- <phone number> The telephone number to call.
- u <unitnum> Use specified channel number.
- v Turn on Bfv API debug mode.

deact

The *deact* program deactivates a hardware module, marking it as dead.

Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *deact* program is found in the *bapp.src* directory.

Command Syntax

deact [-a] [-s] <mod>

Arguments

-a	Deactivate all modules on the board containing <mod>.
-s	The value of <mod> is interpreted as a cPCI slot number.
<mod>	Module to deactivate.

debug_control

The *debug_control* utility allows a user to selectively turn on logging remotely in an application. The application gives the user the ability to control all the debug options available in the *BfvDebugModeSetAdv* function.

The *debug_control* utility is in the *bapp.src* directory.

Command Syntax

```
debug_control [-v] [-u chan]
               [-d debug_mode [-f file1 [-f file2 [-m max_size]]]]
               [-F 0|1 [-a]]
               [-c debug_type[,...] [-l level] [-L file]]
```

Arguments

-d	=	Set Bfv API debug mode, optionally with file or files with limit.
debug_mode	=	Numerical debug mode value used with BfvDebugModeSet (for example, 255)
-F	=	Turn on function entry/exit debugging, optionally with arguments: 1 - enable 0 - disable -a - use arguments
-c	=	Turn on call control debugging, optionally with level or file.
debug_type	=	api 1314 1413 int host ip (one or more)
level	=	none error warning basic verbose(default)
-v	=	Turn on local Bfv API debug mode.

At least one of -d, -F, or -c is required.

The -u option is required for -d and -F.

decode

The *decode* program reads a specified infopkt stream file and lists the individual infopkts that comprise the stream.

The *decode* program is found in the *app.src* directory.

Command Syntax

```
decode [-f] <filename>
```

Arguments

<code>-f</code>	Directs <i>decode</i> to follow indirect infopkts and decode the contents, instead of listing file names.
<code><filename></code>	The name of the infopkt stream file to read.

Example

```
decode filename.ips
```

dfax

The *dfax* program uses the low-level Intel DCX fax transmitting and receiving routines to send and receive facsimiles.

The Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *dfax* program is found in the *app.src* directory.

Command Syntax

```
dfax [-u <unitno>] [-v] -s <phoneno> <dcx_file>
```

or

```
dfax [-u <unitno>] [-v] -r <dcx_file>
```

Arguments

<dcx_file>	The name of the file to send or receive.
-r	Receive mode.
-s <phoneno>	Send to the specified phone number.
-u <unitno>	Channel number.
-v	Turn on Bfv API debug mode.

divert

The *divert* program waits for an incoming call from a caller. It then can do several things:

1. Set the -d num option to divert the call to a given number by calling *BfvLineDivert()* and providing a reason specified by option -e. Option -s specifies the second channel to use to divert the call. If option -s is not specified, Bfv picks the first available channel number on the same module of the channel number specified in option -u.
2. If option -d is not specified and the incoming call was diverted, option -j rejects the diverted call. If option -j is not specified, the software accepts the call. The call is determined to be diverted by the value of *args_cc.cres.redir_reason* being equal to something other than *DIVERT_NONE* (0).

-
3. If the incoming call is accepted, the *divert* program uses the `speed_infopkt_file` recording and playing routines to record and play speech.

Command Syntax

```
divert [-d <num>] [-e <reason>] -j [-u <unitnum>]  
[-s <unitnum>] <infopkt_file> -p -r
```

Arguments

-d <num>	Divert to a given number, or else wait for a diverted-to call.
-e <reason>	If -d, then -e specifies the <code>redir_reason</code> (default is <code>DIVERT_UNCONDITIONAL</code>).
-j	If waiting for a diverted-to call, reject the diverted-to call.
-u <unitnum>	Use specified unit number.
-s <unitnum>	Use specified unit number for second LP.
<infopkt_file>	Name of the file to play or record to.
-p	Play.
-r	Record. The default is 10 seconds.

dlfax

The *dlfax* program uses the highest level infopkt sending routines to send facsimiles and the dialing database functions to implement dialing restrictions. If you include the `-c` argument, the application uses the dialing restrictions of the specified country. If you include the `-l` argument, the application only lists the contents of the dialing database; it does not dial.

Note: If you change the country code between runs of the *dlfax* sample, it can cause incorrect blacklisting of phone numbers. When you change the country code, delete the *dialdb* file created by the sample.

Pre-blacklisted numbers do not work when using the `-c` option. To test this functionality, set the country code in the user-defined configuration file to the desired country.

Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *dlfax* program is found in the *app.src* directory.

Command Syntax

dlfax -l

or

dlfax [options] <infopkt file>

Arguments

-c <ccode>	Use the dialing restrictions for the country specified by ccode. This value must be one of the numeric values listed in <i>ccode.h</i> .
<infopkt file>	The name of the infopkt file to send.
-l	List contents of the dialing database.
-s <num>	Send to given phone number.
-u <unitnum>	Use specified channel number.

dstrip

The *dstrip* program writes out individual PCX pages from a DCX file. The first page is written to *g3data.301*, the second to *g3data.302*, and so on, until all of the pages are written.

This utility converts image data stored in a DCX file to raw PCX format. It is typically used with one of the utilities described in *Appendix A, G3 Legacy Utilities on page 404*.

The *dstrip* program is found in the *app.src* directory.

Command Syntax

dstrip [-o <output_base>]<dcx_filename>

Arguments

<dcx_filename>	Specify the name of the DCX file to convert to PCX format.
-o <output_base>	Use the supplied output file name base to form file names instead of “g3data”.

eccllvoice

The *eccllvoice* program is used to record and play speech and establish the call using the low-level call control function. It uses the speech-infopkt-file recording and playing routines to record and play speech. Recording continues for a maximum of ten seconds or the time specified in the `-n` option.

eccllvoice uses low-level call control function calls as shown in the following table.

In Place Of:	Functions Used:
<i>BfvLineOriginateCall</i>	<i>BfvCallSetup</i> <i>BfvCallWaitForComplete</i>
<i>BfvLineWaitForCall</i>	<i>BfvCallRingDetect</i> <i>BfvCallWaitForSetup</i>
<i>BfvLineAnswer</i>	<i>BfvCallAccept</i> <i>BfvCallWaitForAccept</i>
<i>BfvLineTerminateCall</i>	<i>BfvCallDisconnect</i> <i>BfvCallWaitForRelease</i>

Pressing # on the telephone keypad immediately terminates playback or recording.

Pressing the following keys on the telephone keypad affect the speed and volume at which the application plays back speech:

- 1 = increases the gain
- 2 = decreases the gain
- 3 = increases the speed
- 4 = decreases the speed

The Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *eccllvoice* program is found in the *app.src* directory.

Command Syntax

eccllvoice [options] infopktfile

Arguments

`-c <num>` Call given number, else wait for ring.

-f	Specify record coding format ¹ ; use the number of the format or one of the following names. If there is no number specified, you must use the name.
	adpcm 1
	adpcm32
	adpcm24
	pcm_ulaw 2
	pcm_ulaw64
	pcm_ulaw48
	pcm_ulaw88
	pcm_alaw 3
	pcm_alaw64
	pcm_alaw48
	pcm_alaw88
-l	Loop forever, sending or receiving.
-n <secs>	Specify recording time in seconds.
-p	Play.
-r	Record (default 10 seconds).
-u <unitnum>	Use specified channel.
-v	Enable Bfv API debug mode.

1. Not all coding formats and rate combinations are available on all products.

Requires one -p or -r argument.

fax

The *fax* program uses the highest level infopkt file fax sending and receiving routines to send or receive facsimiles. The local ID is specified from the command line. This application is very similar to *faxhl.c*.

Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *fax* program is found in the *app.src* directory.

Command Syntax

fax [options] <infopktfile>

Arguments

<infopktfile>	Name of the file to send or receive.
-l <string>	Set local ID.
-L	Loop for testing.
-r	Receive a fax.
-s <num>	Send to given phone number.
-u <unitnum>	Use specific channel number.
-w	On receive, do not wait for ring.

Requires one -s or -r argument.

faxhl

The *faxhl* program uses the highest level infopkt file fax sending and receiving routines to send or receive facsimiles. The local ID is specified from the command line. This application is very similar to *fax.c*.

The Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *faxhl* program is found in the *app.src* directory.

Command Syntax

faxhl [options] <infopktfile>

Arguments

<infopktfile>	Name of the file to send or receive.
-l <string>	Set local ID.
-r	Receive a fax.
-s <num>	Send to given phone number.
-u <unitum>	Use specific channel number.
-v	Turn on Bfv API debug mode.

Requires one -s or -r argument.

faxll

The *faxll* program uses the low-level non-infopkt raw data fax sending and receiving routines to send or receive facsimiles. This application uses the *BfvFaxSendFile* function, so 128-byte Brooktrout headers are not permitted. It also uses the user function feature of *BfvLineOriginateCall* to print call progress values.

Use the `-g` or `-a` argument to specify that the next raw data file contains G3 or ASCII data, respectively. If a file contains fine resolution data, use the `-F` argument. Use the `-b` argument to specify a page break.

The Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *faxll* program is found in the *app.src* directory.

Note: The routine contains an example of handling SIP 183 Session Progress message headers on an outgoing call. This capability can be implemented by compiling *faxll.c* with a `SIP_PROGRESS_HEADERS` compiler directive. A non-zero return from the call progress function (`cp_func`) causes a User Interrupt, which terminates the call with a SIP CANCEL message.

Command Syntax

```
faxll [-u <unitno>] [-v] [-h] [-H] -s <phoneno>  
      [-F] [-E #] [-g] [-a] [-p] <file> [-b] ...
```

or

```
faxll [-u <unitno>] [-v] [-p] -r <g3file> ...
```

Arguments

-a <files>	The following files are raw ASCII text.
-b	Beginning of page.
-E #	The next page is an enhanced fax format page. # specifies the format, which is created by ORing the following hex values: 1 = JPEG Enable 2 = Full Color 4 = Default tables 8 = 12bit (vs. 8bit) 10 = No Subsampling 20 = Custom Illuminant 40 = Custom Gamut 100 = JBIG 200 = JBIG L0 400 = Lossless Color 800,1000,2000 = MRC 4000 = Plane Interleave 8000 = Page Length Strip
-F	The next page is fine resolution, otherwise normal.
-g	The following files are raw G3 data (default).
<g3file>	Name of the file to receive.
-h/-H	Insert a page header.
-p	The following files are raw PCX data or receive PCX data.
-r	Receive mode.
-s <phoneno>	Send to the specified phone number.
-u <unitno>	Unit number.
-v	Turn on Bfv API debug mode.

When sending, the application can mix G3 and ASCII files.

Example

```
-g f1 f2 -a f3 -b -g f4
```

When receiving, each file name in the list receives a page of G3 data, so make sure that enough file names appear in the list to accommodate all pages of incoming data.

faxml

The *faxml* program uses the mid-level infopkt file fax sending and receiving routines to send or receive facsimiles.

The Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *faxml* program is found in the *app.src* directory.

Command Syntax

```
faxml[-u <unitno>][-v] -s <phoneno> <infopktfile>  
or
```

```
faxml[-u <unitno>][-v] -r <infopktfile>
```

Arguments

<infopktfile>	Name of the file to send or receive.
-r	Receive mode.
-s <phonenum>	Send to the specified phone number.
-u <unitno>	Unit number.
-v	Turn on Bfv API debug mode.

faxp

The *faxp* program uses the highest level infopkt file fax polling routines to send and/or receive facsimiles. You must specify if the program is going to call (-c) or answer (-a) and one send file (-s), one receive file (-r), or one of both.

This program performs the ordinary sending and receiving functions and all possible polling variations. The local ID is specified from the command line.

Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *faxp* program is found in the *app.src* directory.

Command Syntax

faxp [options]

Arguments

-a	Answer.
-c <num>	Dial given phone number.
-L	Loop for testing.
-r <infopktfile>	File to receive, if permitted.
-s <infopktfile>	File to send, if permitted.
-u <unitnum>	Use specified channel.

Requires one -c or -a argument and one -s or -r argument or one of each.

faxpml

The *faxpml* program uses the medium level infopkt file fax polling routines to send and/or receive facsimiles. It performs ordinary sending and receiving functions and all polling variations. Users can enter the local ID at the command line.

Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file.

The *faxpml* program is found in the *app.src* directory.

Command Syntax

faxpml [**options**]

Arguments

-a	Answer.
-c <num>	Dial given phone number.
-L	Loop for testing.
-r <infopktfile>	File to receive, if permitted.
-s <infopktfile>	File to send, if permitted.
-u <unitnum>	Use specified channel number.

Requires one -s or one -r argument or one of each, and one -c or one -a argument.

feature

The *feature* program manipulates feature set data on the product. It can query or download feature set data.

Feature set data contains licensing information specific to a given module. The licensing information contains information as to what features the user can access, how many channels are available, etc.

The *btcall.cfg* file is the user-defined configuration file. The *feature* program is found in the *bapp.src* directory. Bfv API debug mode is turned on.

Command Syntax

feature [-m <mod>] <action>

Arguments

-m <mod> Apply action to specified module (default 2)

Actions

-d <feature_file> Download ASCII feature file.

-b Perform download using binary file.

-q Query loaded feature set.

-v Enable Bfv API debug mode.

feature downloads ASCII license (feature) files by default, and can also accept binary data if you use the -b option.

When using the -q option, the value displayed for each feature on an SR140 module reflects the total value of this feature for all active SR140 licenses, not for the module itself. It may be higher than a single module can utilize.

firm

The *firm* program is used to download firmware. The firmware consists of several types, by number, which must be downloaded in the proper sequence. *firmload* automatically takes care of identifying the proper files and downloading them in the correct sequence.

The sequence is as follows:

Type 2 (PROC_APP) = Control processor firmware

Type 1 (DSP_APP) = DSP firmware

Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *firm* program is found in the *bapp.src* directory.

Command Syntax

```
firm [-m <mod>] [-c <conf_spec>] -t <type>  
      <firmware_file>
```

Arguments

-m <mod>	Module number to download to (default 2)
-c <conf_spec>	Firmware configuration specification number. This is reported by <code>modinfo -c</code> . This only applies to PROC_APP firmware. The current meaning of the configuration specification value is the number of channels for which to configure the firmware.
-t <type>	Type of firmware downloaded: 1 = DSP_APP 2 = PROC_APP
<firmware_file>	Firmware file to download

If the download type is 2 (PROC_APP), the driver attempts to reestablish communications with the destination module. If the module was previously marked as dead, it might become usable again.

firmload

The *firmload* program is used to download a complete set of standard firmware files to all hardware modules on all boards in a system. The standard firmware files are listed in Chapter 9 in the *Dialogic® Brooktrout® Fax Products SDK Installation and Configuration Guide* and in the *Dialogic® Brooktrout® Release Notes*. Downloads, by default, are attempted for all hardware modules in the range 2 through 0xFD.

The firmware consists of several types, by number, that must be downloaded in the proper sequence. *firmload* automatically takes care of identifying the proper files and downloading them in the correct sequence.

The sequence is as follows:

Type 2 (PROC_APP) = Control processor firmware

Type 1 (DSP_APP) = DSP firmware

Bfv API debug mode is turned on. The *firmload* program is found in the *bapp.src* directory.

This program was previously a script/batch file. The old version of *firmload* is supplied as *firmload_old* in the *bapp.src* directory.

Command Syntax

```
firmload [-c <conf_spec>] [-d] [-b 0|1] [-q] [-e] <firmware_dir> [<module_num> ...]
```

Arguments

- | | |
|----------------|--|
| -c <conf_spec> | Firmware configuration specification number. This is reported by <code>modinfo -c</code> . This only applies to PROC_APP firmware. The current meaning of the configuration specification value is the number of channels to configure the firmware. |
| -d | Do not download the DSP firmware. |
| -q | Quiet mode (disable Bfv API debug mode). |
| -e | Stop on the first error. Normally download continues through the entire available module sequence. |

<firmware_dir>	Directory containing firmware files.
<module_num>	Optional module numbers that have firmware downloaded.

The *firmload* program ensures that the required firmware files exist in the specified directory before beginning a download. There are multiple possibilities for some of the firmware file names. These are listed in the following list in search order. *firmload* looks for the following:

Firmware Type	file name
PROC_APP	cp.bin
DSP_APP	dsp1000.hex, dsp1000_ld.hex, dsp1000_v34.hex

Example

firmload Brooktrout/Boston/fw

font

The *font* program downloads ASCII fonts for fax transmission and reports on font download status.

Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *font* program is found in the *bapp.src* directory.

Command Syntax

font [-m <mod>] [-q] [-d] [-v]

Arguments

-m <mod>	Use specified module (default 2).
-q	Report download status of fonts.
-d	Download fonts as specified in <i>btcall.cfg</i> .
-v	Turn on Bfv API debug mode.

One of -q or -d is required. When -d is specified, fonts are downloaded as specified in the user-defined configuration file, *btcall.cfg*. Up to seven user fonts (0 - 6) and a default font (255) are downloaded.

ipstrip

The *ipstrip* program removes the infopkt header from the G3 or speech data in a specified infopkt stream file and writes each page of the converted data to a file, *g3data.30x*. The first page of converted data is written to the file *g3data.301*, the second to the file *g3data.302*, the third to the file *g3data.303*, and so on until the entire infopkt stream file is converted. All speech data is placed in a single file.

This utility is typically used to convert:

- Received fax data previously converted to an infopkt stream file back to G3 format.
(See *mkinfopk* on page 146 for detailed information on how to create an infopkt stream file.)
- Speech data recorded using the *BfvSpeechRecord* function to a raw speech format.

The *ipstrip* program is found in the *app.src* directory.

Command Syntax

```
ipstrip [-h] [-o <output_base>] <filename>
```

Arguments

- | | |
|------------------|---|
| -h | Puts a 128-byte Brooktrout header at the beginning of each file. The addition of this header causes the utility to store the resolution and width of each page within the file. |
| -o <output_base> | Uses supplied output file name base to form file names instead of “g3data”. |
| <filename> | Specifies the name of the infopkt stream file, consisting of packetized G3 or speech data, to convert to raw format. |

ivr

The *ivr* program is a small, interactive, voice-response system that permits users to receive a fax, send a fax, record a message and play a message. It uses the speech infopkt-file recording and playing routines and the highest level infopkt-file fax receiving and sending routines.

The *ivr_msg* subdirectory contains all prerecorded files, and all new files are created in that subdirectory.

Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *ivr* program is found in the *app.src* directory.

Command Syntax

ivr [options]

Arguments

-L	Loop for testing.
-u <unitnum>	Channel number.

mkdcx

The *mkdcx* program creates a DCX file from a collection of raw PCX files.

The *mkdcx* program is found in the *app.src* directory.

Command Syntax

mkdcx -o <dcx_file> <pcxfilename>...

Arguments

-o <dcx_file>	The name of the output DCX file.
<pcxfilename>	The name of one of the input raw PCX files that is one of the pages of the DCX file. Any number of pcxfilename arguments are permitted.

mkinfofk

The *mkinfofk* program builds an infopkt stream file.

The *mkinfofk* program is found in the *app.src* directory.

Command Syntax

```
mkinfofk -o <output_fname> [-i <input_fname>]
{infopkt_type arg}...
```

Arguments

-o <output_fname>	The name of the output file.
-i <input_fname>	Infopkt type specifications are included in the file <i>input_fname</i> instead of the command line.
infopkt_type	The type of infopkt that follows.
arg	An argument value for the infopkt. Depending on the corresponding <i>infopkt_type</i> , <i>arg</i> is either a file name or a dummy value.

The type *infopkt_type* is indicated by one of the following:

<i>Data type</i>	ascii, g3, speech, and annot. For data and indirect infopkts, the required argument is a file name. The annot type infopkt accepts either an ASCII file name or the argument @. If you pass <i>mkinfofk</i> the argument @, <i>mkinfofk</i> prompts you to enter a text annotation.
<i>Indirect type</i>	indir[infopkt], indirascii, indirg3, indirtiff, indirspeech, indirdcx, and indirwave. Note: The word <i>indir</i> is an abbreviation for <i>indirinfopkt</i> .

Tag type

doc, g3_strip, ascii_strip, page, t30, bop, spi, eos, fax_hdr, eff.

For tag (parameter setting) infopkts, the argument value is not normally used, and *mkinfo* inserts a dummy value. When you do not specify an argument value, *mkinfo* uses hard-coded default parameter values. If, however, *arg* is the @ character, *mkinfo* prompts you to enter parameter values for this infopkt. (The bop type does not have parameters, so the @ character has no effect on it.) When you specify the fax_hdr infopkt type, *mkinfo* prompts you for the label format text, whether or not @ is used.

For a fax infopkt stream file, the first infopkt type must be doc.

For a speech infopkt stream file, the first infopkt type must be spi.

At least one space must be inserted between each command line argument. If you enter *mkinfo* at the command line without specifying any arguments, *mkinfo* displays a list of all the possible arguments.

An example of using *mkinfo* to create an infopkt stream file:

```
mkinfo -o faxstrem.ips doc @ g3_strip 1  
indirg3 letthead.g3 g3_strip 1 indirg3 salute.g3  
ascii_strip 1 ascii lettbody.asc g3_strip 1  
indirg3 signatur.g3
```

Note: 1 is a dummy value when it follows the g3_strip and other infopkts.

mkprompt

The *mkprompt* program creates or updates a prompt file from infopkt files that contain individual phrases. When updating, you can specify an existing phrase number or a new phrase number.

The *mkprompt* program is found in the *app.src* directory.

Command Syntax

```
mkprompt <prompt_file> [<phrase_ipkt>]...
```

or

```
mkprompt -u <phrase_num> <prompt_file> <phrase_ipkt>
```

Arguments

<prompt_file>	Name of the prompt file to create or update.
<phrase_ipkt>	Name(s) of the infopkt file(s) to use for creating or updating the prompt file.
-u <phrase_num>	Update phrase_num; otherwise create.

mktiff

The *mktiff* program creates a TIFF-F file from a collection of raw G3 files in MH/MSB format with EOLs byte-aligned.

The *mktiff* program is found in the *app.src* directory.

Command Syntax

```
mktiff -o <tiff_file> <g3filename>...
```

Arguments

-o <tiff_file>	Is the name of the output TIFF-F file.
<g3filename>	Is the name of one of the input raw G3 files that is one of the pages of the TIFF file. Any number of <i>g3file</i> name arguments are permitted.

If a raw G3 file has a 128-byte Brooktrout header (it is a btG3 file), *mktiff* uses the resolution, width, and number of scan lines from the header when storing the information for that page in the new TIFF file. If it does not encounter a header or if the number of scan lines is 0, *mktiff* counts the actual number of scan lines in the G3 input file.

modinfo

The *modinfo* program lists information about active hardware and software modules maintained within the driver.

The *modinfo* program is found in the *bapp.src* directory.

Command Syntax

modinfo [-p] [-c] [-s] [-h] [-H] [-a] [mod]

Argument

-p	List PCI configuration information.
-c	List firmware configuration options.
mod	Module number whose status is to be printed, otherwise all.
-s	Display cPCI slot and CPU information.
-h	Display hardware resource information.
-H	Display the hardware information reported by the firmware.
-a	Use all previous options.

Each module found is listed, along with whatever hardware or channel information is available. If a module has been marked by the driver as dead, it is listed as **DEAD**.

Configuration values shown when using the *-c* option is supplied to *firm* or *firmload* applications when downloading type 2 (PROC_APP) firmware.

Configuration information from the module's PCI configuration space is displayed using the *-p* option.

The number of channels listed is the total number of channels supported by the module, including its administrative channel. The total is normally one more than the number of work channels, which are mapped into ordinal channel numbers.

For example, a module with 48 work channels is listed as having 49 channels, and a module with no work channels (no firmware downloaded) is listed as having one channel.

playp

The *playp* program waits for a call, then plays the specified sequence of phrases from the specified prompt file until the sequence completes or until the user presses the # key.

Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *playp* program is found in the *app.src* directory.

Command Syntax

```
playp [-u <unit>] <prompt_file> <phrase_num>...
```

Arguments

-u <unit>	Channel number (default is 0).
<prompt_file>	File from which to play the phrase.
<phrase_num>	Phrase number or numbers to play.

rtp

The *rtp* program displays the information in received RTP/RTCP events on an individual channel or on all channels. The *btcall.cfg* file is the configuration file. The *rtp* program is found in the *app.src* directory.

Command Syntax

```
rtp [-u <unitno>] [-v]  
or  
rtp [-a] [-v]
```

Arguments

-a	All Channels
-u <unitno>	Channel number.
-v	Turn on Bfv API debug mode.

shoparam

The *shoparam* program displays the contents of the line structure and the parameter values set in a user-defined configuration file and the read-only parameters taken from the country configuration file.

shoparam does not work unless a module is present in the system, the driver is installed and running, and the firmware is loaded and running on the module.

The *shoparam* program is found in the *app.src* directory.

Command Syntax

shoparam <user configuration filename>

Argument

<user configuration filename>	Specifies the name of the user-defined configuration file. The name used by many other applications is <i>btcall.cfg</i> .
-------------------------------	--

telreset

The *telreset* program resets the telephony configuration state so that new telephony and call switching information is loaded using *BfvLineReset*.

The *telreset* program is found in the *bapp.src* directory.

Bfv API debug mode is turned on.

Command Syntax

telreset [-m <mod>]

Arguments

-m <mod> Use the specified module (default 2).

telsave

The *telsave* program writes telephony parameters to a module's Non-Volatile RAM (NVRAM).

Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The file *callctrl.cfg* is the call control configuration file. The *telsave* program is found in the *bapp.src* directory.

Command Syntax

telsave [-m <mod>] [-v] -s

Arguments

-m <mod> Write to the specified module (default 2)
-v Turn on Bfv API debug mode
-s Save telephony data (required)

tfax

The *tfax* program uses the low-level TIFF-F file fax sending and receiving routines to send and receive facsimiles.

Bfv API debug mode is turned on. The *btcall.cfg* file is the configuration file. The *tfax* program is found in the *app.src* directory.

Command Syntax

```
tfax [-u <unitno>] [-v] -s <phoneno> <tiff_file>  
or  
tfax [-u <unitno>] [-v] -r <tiff_file>
```

Arguments

-r	Receive mode.
-s <phoneno>	Send mode.
<tiff_file>	Name of the file to send or receive.
-u <unitno>	Channel number.
-v	Turn on Bfv API debug mode.

tiffdump

The *tiffdump* program displays the contents of a TIFF-F file. It displays each of the image file directory entries.

Bfv API debug mode is turned on. The *tiffdump* program is found in the *app.src* directory.

Command Syntax

```
tiffdump [-d] <tiff_filename>
```

Arguments

<tiff_filename>	Specifies the name of the TIFF-F file to display.
-d	Causes <i>tiffdump</i> to list all of the ways the file deviates from the TIFF-F specification and to indicate whether Brooktrout Fax Software tolerates the deviation.

tones

The *tones* program uses DTMF detection routines to detect and display incoming touchtones and DTMF generation and single frequency tone generation routines to produce touchtones and other tones.

Bfv API debug mode is turned on. The *btcall.cfg* file is the configuration file. The *tones* program is found in the *app.src* directory.

Command Syntax

tones [options]

Arguments

-c <num>	Call given number, else wait for ring.
-g	Get tones and display them.
-p	Play tones (0,1,2,3,4,5,6,7,8,9,*,#,A,B,C,D).
-r	Turn on RFC 2833 support for detecting incoming DTMF digits as RTP events. This parameter is only valid for the SR140.
-u <unitnum>	Channel number.
-v	Turn on Bfv API debug mode.

Requires one -p or -g argument.

transfer

The *transfer* program waits and accepts an incoming call from one caller (caller A). It then calls *BfvLineTransfer()* to transfer the call to another caller (caller C). During the transfer or while the transfer is occurring, the program can run in one of two modes, supervised and unsupervised. In supervised mode, the program does not complete the transfer until it receives a response from caller C. If the response is 1 (ACCEPT), *transfer* completes the transfer. If the response is 2 (REJECT), *transfer* cancels the call transfer. In unsupervised mode, *transfer* completes the call transfer right away.

The *btcall.cfg* file is the user configuration file and the file *callctrl.cfg* is the call control configuration file.

Command Syntax

transfer [**options**] **phonenum**

Arguments

phonenum	Caller C's telephone number.
options	
-u unit_num	The unit number to use. Value: 0 to (max channels-1)
-m b_channel_mode	The channel mode used if the protocol can do 1 or 2 B-channel transfer. If the protocol supports both, transfer capability is LINE_XFER_ALL. Different protocols support different modes: ■ Hookflash supports 1 B-channel transfers. ■ RLT supports 2 B-channel transfers. ■ ETSI BRI, ETSI PRI, T1-ISDN, NTT BRI, and NTT PRI support both 1 and 2 B-channel transfers. Values: 1, 2 Note: This option does not support the LINE_XFER_TWO_CHAN_NEEDS_NAILUP transfer capability.
-s SUPERVISED mode prompt_file	Turns on supervised mode. Value: a valid prompt file.
-t line_state	Transfer is completed at different states while making the call in unsupervised mode. In supervised mode, the only state allowed to complete transfer is BST_CONNECTED. Value: BST_DIAL_COMPLETE BST_ALERTING BST_CONNECTED
-h	If transfer capability is LINE_XFER_TWO_CHAN, this option puts the first call on the first bchannel on hold before making the enquiry call on the second bchannel.

Examples

➤ **Assume that 110 is phone number of caller C.**

1. Transfer using 1 B-channel and complete transfer at the state BST_DIAL_COMPLETE:

```
transfer.exe -u 0 -m 1 -t bst_dial_complete w110
```

2. Transfer using 1 B-channel and complete transfer at the state BST_ALERTING:

```
transfer.exe -u 0 -m 1 -t bst_alerting w110
```

3. Transfer using 2 B-channels and complete transfer at the state BST_DIAL_COMPLETE:

```
transfer.exe -u 0 -m 2 -t bst_dial_complete w110
```

4. Transfer using 2 B-channels in supervised mode, and ACCEPT the transfer. Caller C presses 1 to accept the call:

```
transfer.exe -u 0 -m 2 -t bst_connected -s  
prompt_file.pkt w110
```

transferll

The *transferll* program performs the same function as *transfer* except that *transfer* uses high-level call control and *transferll* uses low-level call control.

If transfer capability is

LINE_XFER_TWO_CHAN_NEEDS_NAILUP (where the *BfvCallSWConnect()* function connects the two channels), the high level *BfvLineTransfer()* transfer function performs this switch connection automatically. However the low level *BfvCallWaitTransferComplete()* transfer function provides an option to choose the `disable_auto_sw_connect` field of the `args_cc` struct. If `disable_auto_sw_connect` is true, the application must perform the switch connection.

The following list shows the call control functions used by *transfer* and *transferll*.

transfer	transferll
BfvLineTransfer	BfvCallHold and BfvCallWaitForHold BfvCallSetup and BfvCallWaitForComplete BfvCallTransferComplete and BfvCallWaitTransferComplete
BfvLineHold	BfvCallHold and BfvCallWaitForHold
BfvLineRetrieve	BfvCallRetrieve and BfvCallWaitForRetrieve

Command Syntax

transfer [options] phonenum

Arguments

phonenum	Caller C's telephone number.
options	
-u unit_num	The unit number to use. Value: 0 to (max channels-1)

<code>-m b_channel_mode</code>	The channel mode used if the protocol can do 1 or 2 B-channel transfer. If the protocol supports both, transfer capability is <code>LINE_XFER_ALL</code>. Different protocols support different modes: ■ Hookflash supports 1 B-channel transfers. ■ RLT supports 2 B-channel transfers. ■ ETSI BRI, ETSI PRI, T1-ISDN, NTT BRI, and NTT PRI support both 1 and 2 B-channel transfers. Values: 1, 2 Note: This option does not support the <code>LINE_XFER_TWO_CHAN_NEEDS_NAILUP</code> transfer capability.
<code>-s SUPERVISED mode prompt_file</code>	Turns on supervised mode. Value: a valid prompt file.
<code>-t line_state</code>	Transfer is completed at different states while making the call in unsupervised mode. In supervised mode, the only state allowed to complete transfer is <code>BST_CONNECTED</code>. Value: <code>BST_DIAL_COMPLETE</code> <code>BST_ALERTING</code> <code>BST_CONNECTED</code>
<code>-d</code>	Disables auto switch connection if transfer capability is <code>LINE_XFER_TWO_CHAN_NEEDS_NAILUP</code>.
<code>-h</code>	If transfer capability is <code>LINE_XFER_TWO_CHAN</code>, this option puts the first call on the first bchannel on hold before making the enquiry call on the second bchannel.

trombone

The *trombone* program sets up a two channel (trombone) call transfer. After the two channels have been connected together, the *trombone* program records speech from one of the callers. The program starts by waiting for an inbound call on the primary channel. When an inbound call is detected, the application answers and plays a welcome voice prompt to the caller. After playing the

welcome prompt, the program dials an outbound call on the secondary channel. When the outbound call on the secondary channel is answered, the program connects the two parties together with a full duplex connection and records speech from the primary caller. The program terminates the trombone call when either the recording session reaches a maximum timeout value or either of the callers hangs up.

The *trombone* program is found in the *app.src* directory.

The *trombone* program is a multithreaded application that needs to link to the Osi library in addition to a Boston library. To build the *trombone* program, go to the *bfv.api*/*<OS>/app.src* directory and run the make utility with a command line argument of “others”. For example:

(Windows OS)

```
c:\Brooktrout\boston\bfv.api\winnt\app.src> nmake others
```

(Linux OS)

```
[root@RedHat9 bapp.src]$ make others
```

Command Syntax

trombone [options]

Arguments

-p <unitnum>	Primary channel number (Required).
-s <unitnum>	Secondary Channel number (Required).
-w <infopkt_file>	Welcome prompt file name (Required).
-r <infopkt_file>	Recorded file name (Required).
-n <secs>	Record timeout in seconds [10 secs (Default)].
-d <num>	Phone number to dial [“1234” (Default)].
-v <num>	Enable or disable debugging [0-Off, 1-On (Default)].

tstrip

The *tstrip* program writes individual G3 pages from a TIFF-F file. By default, the first page is written to *g3data.301*, the second to *g3data.302*, and so on, until all of the pages are written. This utility converts received fax data stored in a TIFF-F file to raw G3 format. It is typically used with one of the utilities described in *Appendix A, G3 Legacy Utilities on page 404*.

The *tstrip* program is found in the *app.src* directory. Bfv API debug mode is turned on.

Command Syntax

tstrip [-h] [-o <output_base>] [-r] <tiff_filename>

Arguments

-h	Adds a 128-byte Brooktrout header to the beginning of each file. This header includes the resolution, width, and number of scan lines.
-o <outbase>	Uses supplied output file name base to form file names instead of "g3data".
-r	Instructs <i>tstrip</i> to leave the data format as is. Normally <i>tstrip</i> forces the output into MSB format, the standard assumed by all other programs provided by Dialogic.
<tiff_filename>	Specifies the name of the TIFF-F file to convert to G3 format.

TIFF-F pages are internally constructed so that the image data is partitioned into strips. If a page that uses MMR data format is constructed in this way, there are multiple MMR end of data markers (known as EOFBs), one after each strip of data.

In such cases, the data read for different strips cannot simply be concatenated together, since an EOFB indicates an end of page. Because of this, *tstrip* treats each such strip as a separate page.

voice

The *voice* program uses the speech-infopkt-file recording and playing routines to record and play speech. Recording begins when you start speaking. Recording continues for a maximum of ten seconds or the time specified in the *-n* option.

Pressing # on the telephone keypad immediately terminates playback or recording.

Pressing the following keys on the telephone keypad affect the speed and volume at which the application plays back speech:

- 1 = increases the gain
- 2 = decreases the gain
- 3 = increases the speed
- 4 = decreases the speed

Bfv API debug mode is turned on.

The *btcall.cfg* file is the user-defined configuration file. The *voice* program is found in the *app.src* directory.

Command Syntax

voice [options] <infopktfile>

Arguments

- | | |
|-----------------------------|---|
| -c <num> | Call given number, else wait for ring. |
| -f | Specify record coding format; use the number of the format or one of the following names. If there is no number specified, you must use the name. |
| | adpcm 1 |
| | adpcm32 |
| | adpcm24 |
| | pcm_ulaw 2 |
| | pcm_ulaw64 |
| | pcm_ulaw48 |
| | pcm_ulaw88 |
| | pcm_alaw 3 |
| | pcm_alaw64 |
| | pcm_alaw48 |
| | pcm_alaw88 |
| <infopkt_file> | Name of the file to play or record. |
| -l | Loop forever, sending or receiving. |
| -n <secs> | Specify recording time in seconds. |
| -p | Play. |
| -r | Record (default 10 seconds). |
| -u <unitnum> | Use specified channel. |

Not all coding formats and rate combinations are available on all products.

Requires one **-p** or **-r** argument.

voiceraw

The *voiceraw* program uses the raw speech data file recording and playing routines to record and play speech. Recording begins when you start speaking. Recording continues for a maximum of ten seconds or the time specified in the `-n` option.

Pressing `#` on the telephone keypad immediately terminates playback or recording.

Pressing the following keys on the telephone keypad affect the speed and volume at which the application plays back speech:

- 1 = increases the gain
- 2 = decreases the gain
- 3 = increases the speed
- 4 = decreases the speed

Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *voiceraw* program is found in the *app.src* directory.

Command Syntax

voiceraw [**options**] <**speechfile**>

Arguments

-c <num>	Call given number, else wait for ring.
-f	Specify coding format; use the number of the format or one of the following names. If there is no number specified, you must use the name. adpcm 1 adpcm32 adpcm24 pcm_ulaw 2 pcm_ulaw64 pcm_ulaw48 pcm_ulaw88 pcm_alaw 3 pcm_alaw64 pcm_alaw48 pcm_alaw88
-l	Loop forever, sending or receiving.
-n <secs>	Specify recording time in seconds.
-p	Play.
-r	Record (default 10 seconds).
<speechfile>	Name of the file to play or record.
-u <unitnum>	Use specified channel.

Not all coding formats and rate combinations are available on all products.

Requires one -p or -r argument.

wave

The *wave* program uses the speech-wave-file recording and playing routines to record and play speech. Recording begins when you start speaking. Recording continues for a maximum of ten seconds or the time specified in the -n option.

Pressing # on the telephone keypad immediately terminates playback or recording.

Pressing the following keys on the telephone keypad affect the speed and volume at which the application plays back speech:

- 1 = increases the gain
- 2 = decreases the gain
- 3 = increases the speed
- 4 = decreases the speed

Bfv API debug mode is turned on. The *btcall.cfg* file is the user-defined configuration file. The *wave* program is found in the *app.src* directory.

Command Syntax

wave [**options**] <**wavfile**>

Arguments (Options)

- | | |
|--------------|---|
| -c <num> | Call given number, else wait for ring. |
| -f | Specify record coding format; use the number of the format or one of the following names. If there is no number specified, you must use the name. |
| | pcm_ulaw 2 |
| | pcm_ulaw64 |
| | pcm_ulaw88 |
| | pcm_alaw 3 |
| | pcm_alaw64 |
| | pcm_alaw88 |
| | linear 4 |
| | linear128 |
| | linear176 |
| | linear64 |
| | linear88 |
| -l | Loop forever, sending or receiving. |
| -n <secs> | Specify recording time in seconds. |
| -p | Play. |
| -r | Record (default 10 seconds). |
| -u <unitnum> | Use specified channel. |
| <wavfile> | Name of the file to play or record. |

Requires one -p or -r argument.

Note: Using 8-bit 8 kHz and 8-bit 11 kHz Linear recording formats for .wav files can produce poor quality with extra noise. If recording in Linear format, use 16-bit 8 kHz for .wav files. These rates produce better quality recording and are much closer to the format used on the phone line.

Compiling Sample Applications Using Microsoft Developer Studio Project Files

The Brooktrout SDK CD includes Microsoft Developer Studio Project files for all samples in the *bfv.api\app.src* samples directory, including project files for the following versions of Visual Studio:

- Visual Studio 6.0
- Visual Studio .NET 2003
- Visual Studio 2005
- Visual Studio 2008
- Visual Studio 2015 (recommended)

These sample applications function only on Windows systems.

Note: In Visual Studio 6.0, the development environment was called "Workspace". In subsequent versions of Visual Studio, it is called a "Solution".

Using any of the versions of the compiler above, you can view, edit, debug, test, link, compile, and build applications combining the sample files with your product files. You can also use makefiles provided for all operating systems ([See “Compiling Sample Applications Using Makefiles” on page 167](#)).

Note: Microsoft has ended support for Visual Studio 6.0, Visual Studio .NET 2003, Visual Studio 2005, and Visual Studio 2008 compilers. Therefore, in future releases, Dialogic may not be able to support applications using the Bfv API that are compiled with these compilers.

Notes for 64-bit Sample Application Compiling

- **Windows:** You must set up the compiler environment specifically to build for amd64/x64. In this case, the makefile stores the executables in a subdirectory called x64.
- **Linux:** The makefile will normally build 32-bit applications. To build 64-bit applications, define the symbol BUILD_64 to 1 on the make command line (e.g., "make BUILD_64=1"). In this case, the makefile stores the executables in a subdirectory called 64.

Using Brooktrout Files

Dialogic has created a Visual Studio Workspace/Solution *dsp* file and *vcproj* file for each Brooktrout SDK sample in the `\Brooktrout\Boston\bfv.api\winnt\app.src` directory in the Brooktrout SDK InstallShield package. You can see all the samples from a single workspace/solution by opening these files from the compiler application.

Table 6. File Naming Conventions

Type of File	File Name	Compiler Version
workspace/solution	<i>bfv_samples.dsw</i>	6.0
workspace/solution	<i>bfv_samples.sln</i>	.NET 2003 and later
<i>dsp</i> files	<i>samplename.dsp</i>	6.0
<i>vcproj</i> files	<i>samplename.vcproj</i>	.NET 2003 and later

The individual project files have Win32 Debug and Win32 Release options included, both options link to the dynamic version of the "C" runtime library and to the dynamic version of the Bfv library (*bostdll.dll*).

The workspace/solution is constructed so that each of the individual project settings provide the include and library paths rather than being stored in the global setting for Visual Studio itself.

To create an *exe* file using Developer Studio Project files, follow the instructions sent with your Windows software product.

Note: Files for Visual Studio 2005, 2008, and 2015 are appended to include "2005", "2008", and "2015", respectively. For example, *bftdump_2005.vcproj*, *bftdump_2008.vcproj*, and *bftdump_2015.vcproj*.

Compiling Sample Applications Using Makefiles

The sample applications are distributed in source form and are compiled using the supplied makefile(s). The makefile(s) come set up to link the application programs with the Brooktrout Bfv API library.

➤ **To compile the sample applications:**

1. Change to the `/Brooktrout / Boston / bfv.api / <platform> / app.src` directory.

For the operating system you are using, substitute its name for `<platform>` in the following instructions and use the appropriate location where the installed files are located (that is, `/usr/sys` for Linux; and `C:` for Windows).

2. Compile the sample programs in the `app.src` directory using `make`.

The program name `make` is used to refer to the standard make program used with the compiler. The name of this program is `make` on all platforms except Windows; on that platform the name is `nmake`.

3. Change to the `/Brooktrout / Boston / bfv.api / <platform> / bapp.src` directory.
4. Compile the sample programs in the `bapp.src` directory using `make`.

The sample programs are compiled and executable.

Note: If you compile in the x64 environment, the makefile stores the executables in a subdirectory called `x64`.

Combining the Sample Applications

The sample applications provided with the Brooktrout SDK are combined or modified in a variety of ways to demonstrate key capabilities. Typically, combining these applications can require modifications to configuration files, such as *callctrl.cfg*.

Compatibility for Compiling

The current Brooktrout SDK is compatible with all prior Brooktrout SDK versions 6.0 and later. The Brooktrout SDK does not require recompiling of applications if they are linked to the DLL (Windows) or Shared Object (Linux) version of the Bfv library and the `BT_API_SET_VER` macro is used.

5 - Transferring Calls

This chapter describes transferring calls using the Bfv API-level and BSMI-level call control functionality.

Note: The Dialogic® Brooktrout® SR140 Fax Software does not support this functionality.

Call transfer is a method of redirecting an incoming call to an internal line or “transferring” it from one channel to another channel. The phone network can also manage and disconnect the call. Call transfer functionality is supported on the board and inside a public switch. Depending on the protocol in your network, transfers use either the same channel for transfers, such as an analog line, or use two B-channels for transfers, such as explicit call transfer in Europe.

This chapter has the following sections:

- [*Making Call Transfers Using Bfv on page 170*](#)
- [*Making Hookflash Transfers on page 173*](#)
- [*Making Two B-Channel Transfers on page 175*](#)
- [*Making Call Transfers Using QSIG on page 177*](#)
- [*Making Call Transfers Using Active Redirection \(Japan\) on page 181*](#)
- [*Making Explicit Call Transfers \(ECT\) With E1 ISDN and BRI on page 183*](#)
- [*Making Two-Channel Call Transfers \(Tromboning\) on page 185*](#)
- [*Transferring Calls Using Release Link Trunk Transfer on page 197*](#)
- [*Placing Calls on Hold Using BSMI on page 205*](#)

Making Call Transfers Using Bfv

The diagram in [Figure 10](#) illustrates a transfer using the *BfvLineTransfer* high level call transfer function.

The application issues a *BfvLineTransfer* function call to transfer a call. *BfvLineTransferCapabilityQuery* checks the transfer capability through `LINE_XFER_` queries. If the line is capable of the transfer type requested (such as transferring using two B-channels, `LINE_XFER_TWO_CHAN`), then the call is placed on hold with *BfvCallHold* while the other line is checked for availability with *BfvCallSetup*. The call is transferred and supervised until either party ends the call. *BfvLineTransferComplete* notifies the application that the lines are connected.

If all conditions follow true, the application successfully transfers the call. If a statement runs into a false condition, only the enquiry call ends.

For more information on the messages, please refer to *Volume 2* in the *Dialogic® Brooktrout® Bfv APIs Reference Manual* for specific messages.

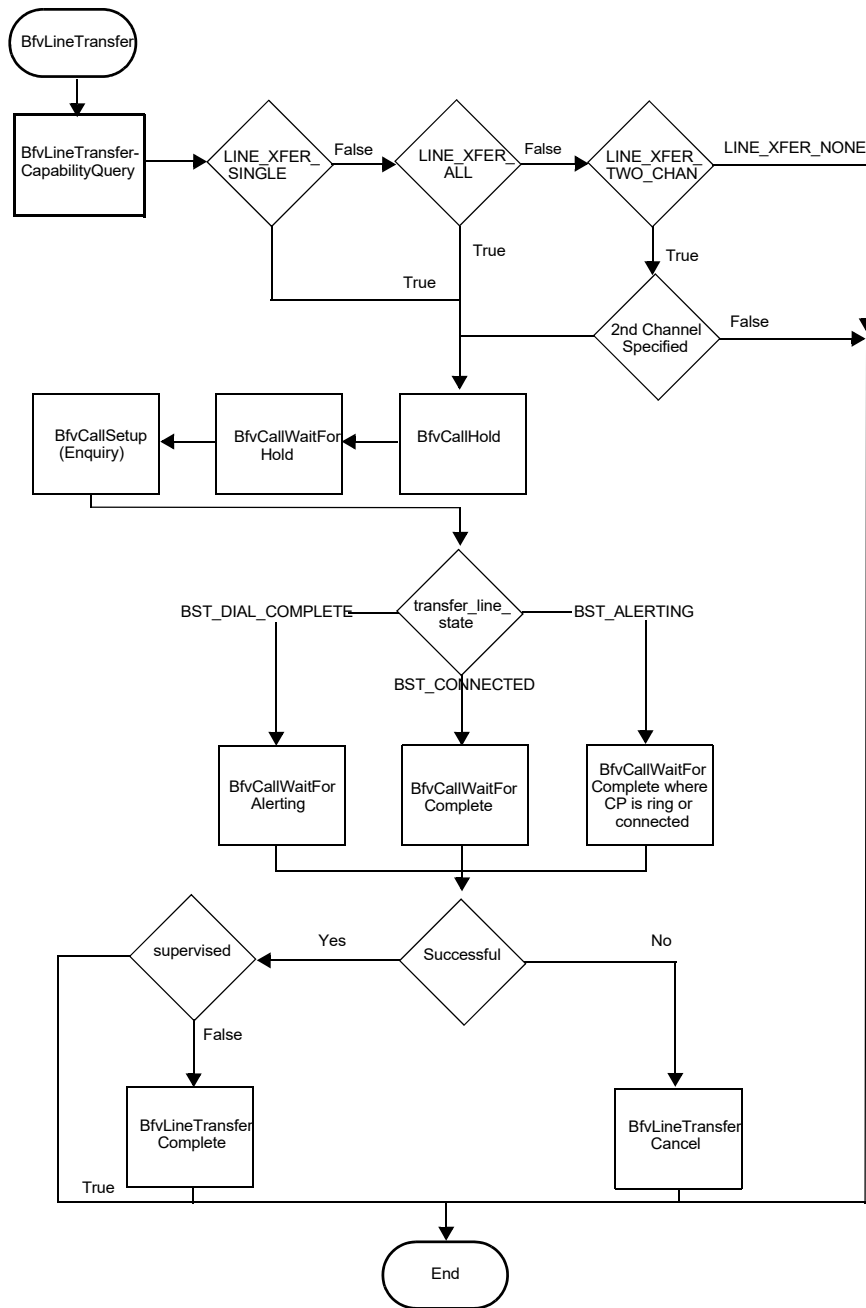


Figure 10. High-level Call Transfer using Bfv

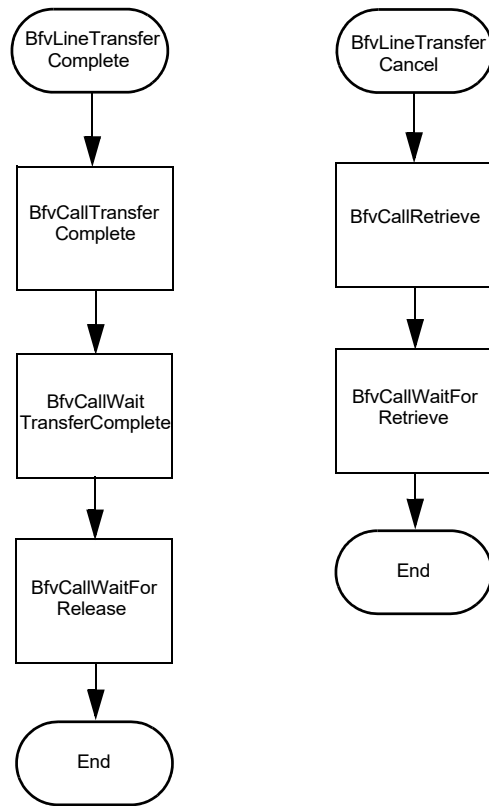


Figure 11. Low-level Call Transfer using Bfv

Making Hookflash Transfers

You can make hookflash call transfers using either the Bfv- level API or BSMI-level API. There are two types of hookflash transfers:

- Analog loop start signaling
Transfers using loop start signaling commonly use hookflash. Hookflash allows for both blind and attended transfers using the same channel. In a blind transfer, the application drops out of the call before the transfer completes. In an attended transfer, the application waits until the call is successfully connected to a new number before completing the transfer.
- T1 Robbed Bit signaling
T1 Robbed Bit hookflash transfers use E&M signaling to generate a hookflash (wink) and initiate dial tone recall to transfer a call using a single channel.

Using Bfv Applications

To configure using the Dialogic® Brooktrout® Configuration Tool (Windows only):

- Set each port's Protocol Options to T1 Robbed Bit or Analog.
- Set the Flash Hook Duration between 1 - 500. These are 10ms units.
- "Set the Protocol File for:
 - ◆ "T1 RBS:
`C:\Brooktrout\Boston\config\winkstart.lec`
 - ◆ Analog:
`C:\Brooktrout\Boston\config\analog_loopstart_us.lec`
- Set the Transfer Variant to Hookflash.
- Set the country_code in BTCall Parameters to 0010 (US).

Note: You must be in Advanced Mode in the Brooktrout Configuration Tool to configure BTCall Parameters.

To configure using configuration files:

■ Call Control Configuration File (*callctrl.cfg*)

- ◆ Set the port configuration

T1 Robbed Bit:

port_config=t1_robbed_bit

Analog:

port_config=analog

- ◆ Set the protocol file

T1 Robbed Bit:

protocol_file=C:\Brooktrout\Boston\config\winkstart.lec

Analog:

protocol_file=C:\Brooktrout\Boston\config\analog_loopstart_us.lec

- ◆ Set the transfer_variant

transfer_variant=hookflash

- ◆ Set the flash duration

flash_hook_duration=50

■ User-Defined Configuration File (*btcall.cfg*)

- ◆ Set the country code

country_code 0010

Hookflash transfer is the only T1 Robbed Bit and analog explicit call transfer method.

For more information on configuration, refer to the *Dialogic® Brooktrout® Fax Products SDK Installation and Configuration Guide*.

Using BSMI Applications

In T1 and analog BSMI applications, use *L4L3m TX_HOOKFLASH* to set the duration of the hookflash signal. The field to set the signal is:

```
L4_to_L3_struct.data.signal_duration_data = 0;
```

The duration of the hookflash signal is in milliseconds. The default value is 0, which is equal to 500ms in duration. The maximum setting is 64k.

For more information on configuration, refer to the *Dialogic® Brooktrout® Fax Products SDK Installation and Configuration Guide*.

Making Two B-Channel Transfers

When making a two B-channel transfer, the central office connects two outside calls through the central office, freeing the B-channels to take more calls. You can only transfer calls using two B-channels on T1 ISDN PRI configurations.

Bfv maintains a maximum of two calls per B-channel. Of these two calls, only one call is on hold at a time. Bfv provides a hold function using *BfvCallHold* in your application:

```
BfvCallHold(lp, &args);
```

See [page 205](#) for more information about putting a call on hold using BSMI.

To configure using the Brooktrout Configuration Tool (Windows only):

- Set each port's Protocol Options to T1 ISDN.
- Set your network's Protocol, such as AT&T PUB 41449.
- Set your network's Switch Type, such as AT&T #4 ESS.
- Set the Transfer Variant to Two B-Channel.
- Set the country_code in BtCall Parameters to 0010 (US).

Note: You must be in Advanced Mode in the Brooktrout Configuration Tool to configure BtCall Parameters.

To configure each port using configuration files:

- Call Control Configuration File (*callctrl.cfg*)
 - ◆ Set the port configuration

```
port_config=t1_isdn
```

-
- ◆ Set the protocol

`protocol=att`

- ◆ Set the transfer_variant

`transfer_variant=tbct`

- User-Defined Configuration File (*btcall.cfg*)

- ◆ Set the country code using the *btcall.cfg* configuration file:

`country_code 0010`

For more information on the *BfvCallHold* function, refer to *Volume 2*, in the *Dialogic® Brooktrout® Bfv APIs Reference Manual*. For more information on configuration, refer to the *Dialogic® Brooktrout® Fax Products SDK Installation and Configuration Guide*.

Making Call Transfers Using QSIG

ISDN QSIG

QSIG is an ISO standard that defines the ISDN signaling and control methods used to link PBXs in private ISDN networks. The standard extends the “Q” point in the ISDN logical reference model, which was established by the ITU-T in its Q.93x series of recommendations that defined the basic functions of ISDN switching systems.

QSIG is an ISDN based protocol for signaling between nodes of a Private Integrated Services Network (PISN). In particular, QSIG allows compatibility among products in a multi-vendor environment.

QSIG is only specified for the ISDN Interface Type Point-to-Point, that is you cannot configure or use this protocol in conjunction with a Point-to-Multi-Point interface.

Supplementary Services Support

QSIG supports the following supplementary services:

- Calling Line Identification Presentation (Caller ID)
- Calling Line Identification Restriction
- Advice of Charge
- Call Diversion (Unconditional, Busy and Not Responding)
- Call Transfer
- Name Identification

This service uses the same caller ID functions that are used with other protocols: *BfvLineWaitForCall* and *BfvCallWaitForSetup*.

Calling Line Identification Restriction

Uses the *BfvCallSetup* function which allows the Bfv application to set call presentation and screening on a per call basis.

Advice of Charge (AOC)

Supplementary service enables a user to receive information on the recorded charges for a call when the call is terminated.

Call Diversion

Call Diversion (Unconditional, Busy and Not Responding) contains three scenarios:

- **Originating** - the board places a call and the far end attempts to divert the call to a different destination. You can set Originating to enable or disable through the call configuration file using *enable_call_diversion* flag.
- **Served** - the board receives an incoming call and attempts to divert it. This service requires new Bfv API calls to initiate and wait for the diversion to complete. However, it is possible for a call to fail to divert. You use *BfvCallDivert*, *BFfvCallWaitForDivert*, and *BfvLineDivert*. Once the call is diverted, it is terminated. The application uses *BfvLineTerminteCall* after a successful call diversion to make sure that the call has been released completely.
- **Diverted To** - the board receives an incoming call that is diverted by another party. This service uses *call_res.redir_number* and *call_res.redir_reason* that is returned by the *BfvLineWaitForCall* or *BfvCallWaitForSetup* which notifies the Bfv application that the incoming call is being diverted from another party. As a result, the phone number of the device that diverts the call and the reason for the diversion is provided. The Bfv application has the option to refuse the diverted call through *BfvCallReject* followed by *BfvCallWaitForRelease*.

Note: You must set the QSIG control parameter, *disable_alerting* to On in order for the call to be rejected and retained by the party attempting to divert the call. If this parameter is not set to On, then the incoming call is terminated.

You can manually send an alerting message through *BfvCallSendAlerting* and is used when alerting is disabled in the call control configuration file and when the application wants to send an alerting message to the remote end rather than answer the call.

Call Transfer

QSIG supports two B-channel transfer. Both channels must support the TBCT capability. This is accomplished through the *BfvLineTransferCapabilityQuery*. However you must connect the 2-B channels so that both parties can communicate while the transfer occurs. The application then uses *BfvCallSwitchConnect* function to connect both B-channels. For further details, see *Volume 2, Bfv-Level Call Control and Call Switching, Dialogic® Brooktrout® Bfv APIs Reference Manual*.

Name Identification

This feature allows the Bfv API to see the text name of the user similar to Caller ID on an analog phone line. Therefore, if the network provides the calling party, the calling party's name is reported through *BfvLineWaitForCall/BfvCallWaitForSetup* functions located in the *res.calling_party_subaddress* field of the *args* structures.

[Table 7](#) lists the Bfv APIs that are associated with QSIG and are documented in detail in *Volume 2, Bfv-Level Call Control and Call Switching, Dialogic® Brooktrout® Bfv APIs Reference Manual*.

Table 7. Bfv APIs Associated with QSIG

BfvLineWaitforCall
BfvCallWaitForSetup
BfvCallTransferCapabilityQuery
BfvCallDivert*
BfvCallWaitForDivert*
BfvCallSendAlerting*
BfvLineDivert*

* Supported only on the QSIG protocol.

For detailed information concerning the Bfv API QSIG call control configuration functions and data structures, see *Volume 2, Bfv-Level Call Control and Call Switching, Dialogic® Brooktrout® Bfv APIs Reference Manual*.

To configure using the Brooktrout Configuration Tool (Windows only):

- Set each port's *Protocol Options* to *T1 QSIG* or *E1 QSIG*.
- Set the *Transfer Variant* to *QSIG*.

To configure using configuration files:

- Call Control Configuration File (callctrl.cfg)
- "Set the port configuration

port_config=t1_qsig BOLD

or

port_config=e1_qsig BOLD

-
- Set the transfer variant using the callctrl.cfg configuration file:

transfer_variant=qsig

- User-Defined Configuration File (btcall.cfg)

- ◆ Set the country code using the btcall.cfg configuration file:

country_code 0010 BOLD

For more information on configuration, refer to Volume 6 in the *Dialogic® Brooktrout® Fax Products SDK Installation and Configuration Guide*.

Making Call Transfers Using Active Redirection (Japan)

To transfer calls in Japan, use Active Redirection, a call transfer method that uses the central office or a PBX to transfer calls. Active redirecting is only available for T1 ISDN PRI and ISDN BRI.

To configure using the Brooktrout Configuration Tool (Windows only):

- Set port's Protocol Options to T1 ISDN or BRI.
- Set the network Protocol to JATE (Japan) INS-1500.
- Set the network Switch Type to Japan.
- Set the Transfer Variant to Jate ISDN.
- Set the country_code in BTCall Parameters to 0810.

Note: You must be in Advanced Mode in the Brooktrout Configuration Tool to configure BTCall Parameters.

To configure each port using configuration files:

■ Call Control Configuration File (*callctrl.cfg*)

- ◆ Set the port configuration

T1 ISDN:
`port_config=t1_isdn`

ISDN BRI
`port_config=bri`

- ◆ Set the protocol

T1 ISDN:
`protocol=ntt`

- ◆ Set the switch type

T1 ISDN:
`switch_type=ntt`

- ◆ Set the transfer variant

`transfer_variant=ntt`

For ISDN BRI with point-to-multipoint configuration, set:

`transfer_variant=ntt_mp`

■ User-Defined Configuration File (*btcall.cfg*)

- ◆ Set the country code

`country_code 0810`

For more information on the Brooktrout Configuration Tool, refer to the chapter on using the Brooktrout Configuration Tool in the *Dialogic® Brooktrout® Fax Products SDK Installation and Configuration Guide*.

Making Explicit Call Transfers (ECT) With E1 ISDN and BRI

Explicit call transfer (ECT) with E1 ISDN transfers calls using either a single B-channel or two B-channels using switch resources in the central office or PBX. Explicit call transfer is only used with the Bfv API, and is available on ISDN PRI (E1) and ISDN BRI.

To configure using the Brooktrout Configuration Tool (Windows only):

- Set each port's Protocol Options to E1 ISDN or BRI.
- Set your network's Protocol to the appropriate protocol type, such as EURO.
- Set your network's Switch Type to the appropriate switch type, set to Unknown/ITU conformant.
- Set the Transfer Variant to Explicit Call Transfer.
- Set the country_code in BtCall Parameters, leave as the default 0010.

Note: You must be in Advanced Mode in the Brooktrout Configuration Tool to configure BtCall Parameters.

To configure using configuration files:

- Call Control Configuration File (*callctrl.cfg*):

- ◆ Set the port configuration

E1 ISDN

```
port_config=e1_isdn
```

ISDN BRI

```
port_config=bri
```

- ◆ Set the protocol

E1 ISDN

```
protocol=EURO
```

- ◆ Set the switch type:

E1 ISDN

`switch_type=unknown`

- ◆ Set the transfer variant

`transfer_variant=etsi_exp_link`

You can also choose implicit link by setting:

`transfer_variant=etsi_imp_link`

- User-Defined Configuration File (*btcall.cfg*)

- ◆ Set the country code

`country_code 0010`

Your application should support the European Telecommunication Standards Institute (ETSI) recommendations for Explicit Call Transfer. Call hold should also be implemented in the application using *BfvCallHold*, based upon recommendations from ETSI.

For more information on *BfvCallHold*, refer to the *Volume 2, Bfv API Reference Manual*. For more information on the Brooktrout Configuration Tool, refer to the chapter on using the Brooktrout Configuration Tool in the *Dialogic® Brooktrout® Fax Products SDK Installation and Configuration Guide*.

Making Two-Channel Call Transfers (Tromboning)

This section provides information about creating a two-channel call transfer (trombone call transfer).

A two-channel call transfer occurs when an application connects a calling party to a called party with a full duplex connection, while maintaining control of both calls. For some applications, this method of transfer provides redundancy with no added benefits. However, for applications that require monitoring or functionality not provided by the caller and called resources, this method supports the additional functionality. The Bfv API manages two-channel transfer by transferring an existing call to another channel and becoming a link in the network rather than an end-point for either channel.

An application might use this method of call transfer to monitor or record the calls.



When configuring your module for two channel call transfer activities, establish a maximum of 5 channels per DSP configuration.

The section contains the following:

- [*Setting up the Two-Channel Call Transfer*](#) on [*page 186*](#)
- [*Actions During a Two-Channel Call Transfer*](#) on [*page 189*](#)
- [*Terminating the Two-Channel Call Transfer*](#) on [*page 194*](#)
- [*Disconnecting Resources*](#) on [*page 195*](#)

Setting up the Two-Channel Call Transfer

In the typical two way call transfer application, the application detects an incoming call and answers the call. The application then performs voice playback functions as needed to get a person's phone number. The application then makes an outbound call on another channel by dialing that person's phone number. When the person answers the call, the application connects the two parties together and monitors both resources to provide additional functionality.

Connecting Resources

To create the two-channel call transfer and perform the required additional functionality, the application must connect various hardware resources. These resources are network timeslots for a T1/E1 phone line or channels. Each resource has an input slot and an output slot. When connecting two resources together, one resource must be defined as the source and the other resource defined as the destination. The application should define the connection type between the two resources as “transmit” because the Brooktrout hardware maintains connections as transmits.

To understand the basic connections that exist between a channel and a network timeslot, see [Figure 12](#). These connections are defined in the Call Control Configuration File, *callctrl.cfg*. The connection lines represent both signaling and voice data. When making a two-channel call transfer, the application should modify only the voice data connections.

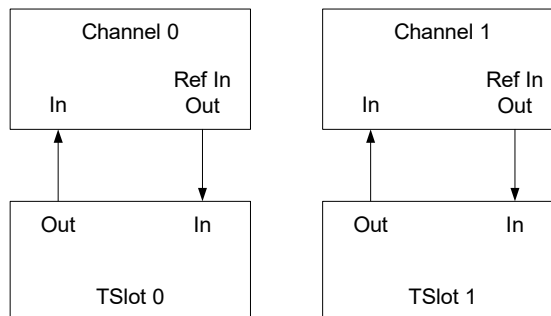


Figure 12. Connections for Standard (non-transfer) Operation

In [Figure 12](#) the output slot of Channel 0 transmits to the input slot of network timeslot 0, while the output slot of network timeslot 0 transmits to the input slot of Channel 0. The connections between Channel 0 and network timeslot 0 create a full duplex voice session. Also observe that the output slot of Channel 1 transmits to the input slot of network timeslot 1, while the output slot of network timeslot 1 transmits to the input slot of Channel 1.

Use the following Bfv functions to access or change the connection information that is maintained on the Brooktrout hardware:

- *BfvCallSWClearConns()*
Clears resource connection
- *BfvCallSWConnect()*
Connects or disconnects resources
- *BfvCallSWGetConns()*
Queries for connection information
- *BfvCallSWGetInfo()*
Provides information about the two-channel call

Refer to *Volume 1 of the Dialogic® Brooktrout® Bfv APIs Reference Manual* for further information on these functions.

The *BfvCallSWConnect()* and *BfvCallSWGetConns()* functions are the most important when setting up a two-channel call transfer. Before creating a two-channel call transfer, call the *BfvCallSWGetConns()* function to get the current connection data that is stored on the module. Store this information in a local variable where it is easily accessed and restored when disconnecting the two-channel call transfer (see [Terminating the Two-Channel Call Transfer](#) on [page 194](#)).

When connecting and disconnecting resources, the application should primarily use the *BfvCallSWConnect()* function. The application must provide all connections with a source and destination resource. The following rule is very important when connecting resources together:



A source resource can have many destination resources, but a destination resource must have only one source resource.

Before setting up the two-channel call transfer, remove any existing connections that conflict with the transfer connections.

For a model of a two-channel call transfer, see [Figure 13](#). The OUT label represents the source of the connection and the IN label represents the destination of the connection.

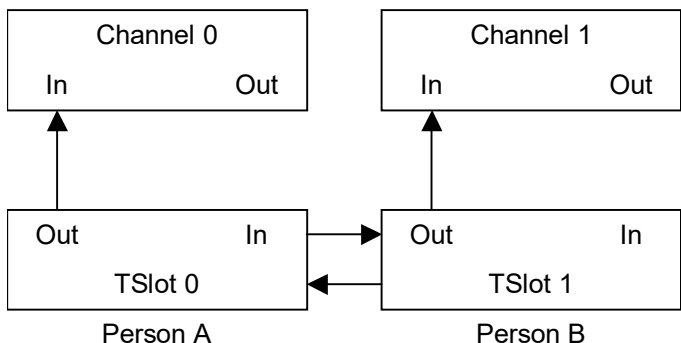


Figure 13. Connections for a Two-Channel Call Transfer

The application removed the connection from the output of Channel 0 to the input of network timeslot 0 and the connection from the output of Channel 1 to the input of network timeslot 1. If these connections were not removed from [Figure 12](#) on [page 186](#), then Channel 0 and network timeslot 1 would both be attempting to transmit to the input of network timeslot 0, violating the above rule.

The application created a new connection from the output of network timeslot 0 to the input of network timeslot 1 and another new connection from the output of network timeslot 1 to the input of network timeslot 0 (see [Figure 13](#) on [page 188](#)).

Using this model, Channel 0 can record the data that is being transmitted from the network timeslot 0 and Channel 1 can record the data that is being transmitted from network timeslot 1.



Because of firmware requirements, when disconnecting resources, you have to invert the source and destination resources. For example, if you made a transmit connection from the output slot of Channel 0 to the input slot of network timeslot 0, where Channel 0 is the source, then you must disconnect from the input slot of network timeslot 0 to the output slot of Channel 0.

Actions During a Two-Channel Call Transfer

After the application establishes a two-channel call transfer, it can then perform voice playback and voice recognition. In addition to these voice actions, the application must provide secondary functionality. For example, when performing voice recognition during a two-channel call transfer, the application must configure echo cancellation in the channel correctly. Also, when performing voice playback to a caller, the application must place the other caller on hold to avoid providing the original caller with two source inputs.

Performing Echo Cancellation

When a caller sends voice data over the telephone network, the network reflects some of that data back as input data. The reflected data is then mixed in with valid input voice data from the remote end. The channel that is connected to the telephone network must be able to remove the reflected data from the valid input data. This technique is called echo cancellation.

When a channel performs echo cancellation, it analyzes the voice data that is being sent out into the telephone network. When this voice data reappears as input voice data (the echo), the channel removes the data from the input data stream. What is left is the actual voice data from the remote end.

When the application creates a two-channel call transfer, the two callers become sources of the reflected data. The application must configure the echo cancellation portion of the channel resource to remove repeated input from two sources by using its reference signal slot (the channel's reference number is 1). The application must also call the *BfvSpeechEchoCancelControl()* function to configure the channel to accept an input reference signal on slot #1.

For example, your application could create a connection from the output slot of Network resource 1 to the reference input slot of Channel 0 by calling the *BfvSpeechEchoCancelControl()* and *BfvCallSWConnect()* functions with the following arguments:

```
BT_ZERO(speech_args);
/* Configure channel to get its input reference signal from slot #1. */
Speech_args.echoc_op = ECHOC_OP_ALT_INPUT_ENABLE;
BfvSpeechEchoCancelControl (lp, &speech_args);

BT_ZERO(args);
args.conn_mode = CALL_SW_TRANSMIT_ONLY_DEF;
args.src_port_class = CALL_SW_PORT_NETWORK_DEF;
args.src_port_unit = 0;
args.src_stream = 0;
args.src_slot = 1; /* Network Timeslot */
args.dest_port_class = CALL_SW_PORT_CHANNEL_DEF;
args.dest_port_unit = 2; /* Logical DSP Channel Number */
args.dest_stream = 0;
args.dest_slot = 1; /* Reference Slot */
BfvCallSWConnect (lp, &args);
```

Figure 14 on *page 191* shows the connections required to configure the echo canceller on a channel during a two-channel call transfer while recording from person A. The application sends the final echo cancelled data up to the host for recording.

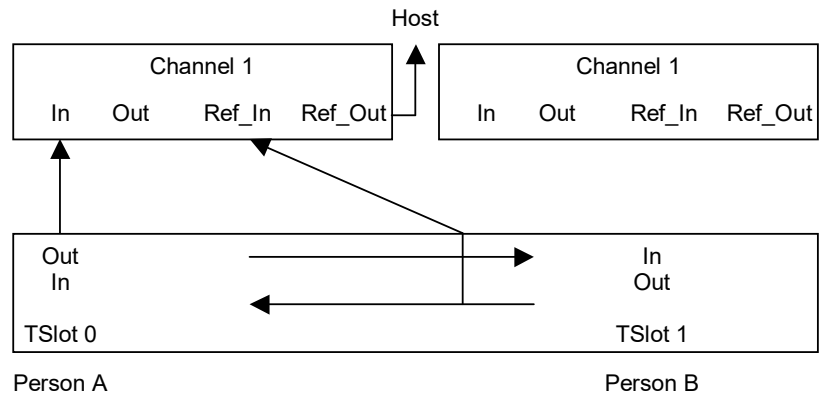


Figure 14. Required Connections for Echo Cancellation

Playing Back Voice Recordings

Voice playback during a two-channel call transfer can take one of two forms.

- The voice application plays voice data to both callers.

In this example, the channel playing the voice data is the source resource and the network timeslots are the destination resources. Before voice playback begins, to avoid providing the destinations with more than one source, the application must disconnect the full duplex connection between the network timeslots. This ensures that the two network timeslots can't transmit data to each other. After voice playback has ended, the application can re-establish the full duplex connection between the network timeslots. Refer to [Figure 15](#) for this situation.

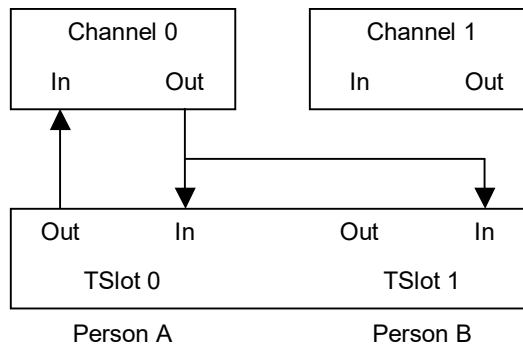


Figure 15. Playing Voice Data To Two-Channels

- The voice application plays data to one caller while placing the other caller on hold.

In this example, the channel transmits voice data to just one network timeslot. The application must place the other network timeslot on hold (disconnecting it from the full duplex connection using the *BfvCallSWConnect()* function) to avoid providing the network timeslot that is receiving the playback data with a second source input.

When your application terminates the full duplex connection to the network timeslot that is not receiving playback data, the caller might hear noise. To prevent this, connect the input slot of the network timeslot to the output slot of an idle channel (an idle

channel provides silence generation on its output). After playback is completed, the application can re-establish the full duplex connection.

The following series of illustrations demonstrate the changes.

In [Figure 16](#), there is a full duplex connection between channel 0 and TSlot 0, and another full duplex connection between channel 1 and TSlot 1.

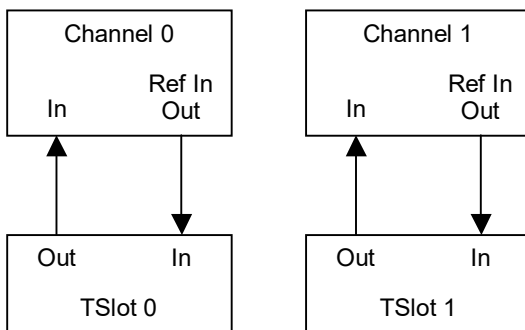


Figure 16. Standard Full Duplex Connections

In [Figure 17](#), there is a full duplex connection between TSlot 0 and TSlot 1. Channel 0 is recording the caller on TSlot 0. The application connected TSlot 1 to the reference input of channel 0 to provide echo cancellation.

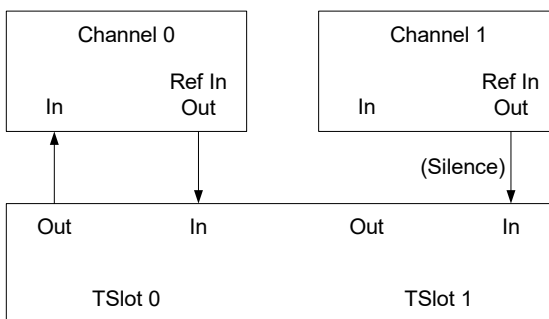


Figure 17. A Connected Two Channel Transfer.

In [Figure 18](#), the application places the caller on TSlot 1 on hold and generates silence from channel 1 to the caller. There is a full duplex connection between channel 0 and TSlot 0. Channel 0 is playing a voice prompt to the caller on TSlot 0 while recording.

To take the caller on TSlot 1 off hold, re-establish the connections as they were in [Figure 17](#).

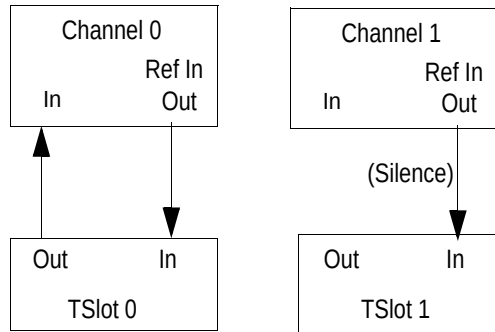


Figure 18. Playing the Voice Prompt and Generating Silence

Terminating the Two-Channel Call Transfer

Your application terminates a two-channel call transfer by deleting the transfer connections and restoring the original connections that were present before the two-channel call transfer was created. After terminating a two-channel call transfer, the application remains connected to the original caller (person A). There are three situations that cause a two-channel call transfer to be terminated:

- If the application detects a recognition signal that indicates terminating the two-channel call.
The application hangs up on the called party (person B) while staying connected to the original calling party (person A).
- The called party hangs up.
The application alerts the original caller of the remote hang up, then disconnects the called party while staying connected to the original calling party.
- The original calling party hangs up.
The application should then disconnect the entire two-channel call transfer and wait for another call.

Disconnecting Resources

When disconnecting resources, invert the source and destination resources so that the firmware responds appropriately.



Due to firmware requirements, when disconnecting resources, invert the source and destination resources. For example, if you made a transmit connection from the output slot of Channel 0 to the input slot of network timeslot 0, where Channel 0 is the source, then you must disconnect from the input slot of network timeslot 0 to the output slot of Channel 0.

Use the *BfvCallSWConnect()* function to disconnect connections. For example, if you made a transmit connection from the output of Channel 0 to the input of network timeslot 0, where Channel 0 is the source, then you must disconnect from the input of network timeslot 0 to the output of Channel 0. The following source code shows this:

```
BT_ZERO(args);
args.conn_mode = CALL_SW_DISCONNECT_DEF;
args.src_port_class = CALL_SW_PORT_NETWORK_DEF;
args.src_port_unit = 0;
args.src_stream = 0;
args.src_slot = 0; /* Network Timeslot */
args.dest_port_class = CALL_SW_PORT_CHANNEL_DEF;
args.dest_port_unit = 2; /* Logical DSP Channel Number */
args.dest_stream = 0;
args.dest_slot = 0;
BfvCallSWConnect (lp, &args);
```

If the application stored the original connection information in a local variable before creating the two-channel call transfer, restore the original connection by calling the *BfvCallSWConnect* function and referencing the local variable.

If the application configured the echo canceller in the channel to get an input reference signal from slot #1, then the application must remove this configuration to return the echo canceller to its default behavior. To do this the application must call the *BfvSpeechEchoCancelControl()* function (See Volume 3 of the *Dialogic® Brooktrout® Bfv APIs Reference Manual*). For example:

```
BT_ZERO(speech_args);  
speech_args.echoc_op = ECHOC_OP_ALT_INPUT_DISABLE;  
BfvSpeechEchoCancelControl (lp, &speech_args);
```

Calling the *BfvLineReset()* function on a channel also returns the echo canceller to its default behavior.

Transferring Calls Using Release Link Trunk Transfer

Release Link Trunk is an explicit call transfer method for Nortel DMS-250 switches and is only available for Nortel switches.

The host application initiates the Release Link Trunk (RLT) action, but the call transfer is completed within the public switch network. Calls come into the network on two B-channels. Each channel maintains its call and both channels are busy. When the call finishes, the B-channels are torn down.

Using Bfv Applications

Set Release Link Trunk in your Bfv application using the *BfvLineTransfer* argument:

```
args.lp_second_Channel
```

A zero (0) value transfers calls over a single channel, such as an analog line.

To configure using the Brooktrout Configuration Tool (Windows only):

- Set each port's Protocol Options to T1 ISDN.
- Set the Protocol to Northern Telecom NIS A211-1.
- Set the Switch Type to Northern Telecom DMS-250.
- Set the Transfer Variant to Release Link Trunk (DMS-100 or DMS-250).
- Set the country_code in BTCall Parameters to 0010 (US).

Note: You must be in Advanced Mode in the Brooktrout Configuration Tool to configure BTCall Parameters.

To configure using configuration files:

- Call Control Configuration File (*callctrl.cfg*)
 - ◆ Set the port configuration:

```
port_config=t1_isdn
```

-
- ◆ Set the protocol
`protocol=nortel`
 - ◆ Set the switch type
`switch_type=nti_dms250`
 - ◆ Set the transfer variant
`transfer_variant=rlt`
 - User-Defined Configuration File (*btcall.cfg*)
 - ◆ Set the country code
`country_code 0010`

Using BSMI Applications

Call transfer RLT functionality in BSMI is set in *L4L3CALL_REQUEST*:

```
rlt_service = 1
```

Note: RLT is set with any non-zero value.

The destination number is set in *L4L3CALL_REQUEST*:

```
data.call_req_data.redirect_num.num_digits = 0;
```

RLT functionality in BSMI requires the D-channel set in *L4L3mENABLE_PROTOCOL* with:

```
'switch_type' = IISDNstDMS-250
```

and

```
'variant' = IISDNvarNORTEL_CUSTOM
```

If the switch does not support RLT, an *L3L4mALERTING* message is received and the call is not transferred.

Applications with RLT release PRI-ISDN circuits after call transfer occurs, releasing corresponding circuits. A call comes through a B-channel, the caller requests a number and that call is transferred

from one B-channel to the other B-channel. The calls are connected inside the public switch and remain active. The application tears down both B-channels and releases the link.

Calls without RLT are extended and subsequently bridged to a third party. The third party maintains the call and circuits are kept active and in service. This ties up additional circuits on the PRI trunk.

For more information on *L4L3mENABLE_PROTOCOL* and *L3L4mALERTING*, please refer to *Volume 5*, in the *Dialogic® Brooktrout® Bfv APIs Reference Manual*.

Call Control Sequence Diagrams

The charts below describe call transfer using BSMI, both with the RLT functionality of the DMS-250, and the traditional method (for purposes of comparison).

In both calls, the board receives an incoming call and determines that the call needs to be rerouted to an alternate destination. In the non-RLT call transfer configuration, the application initiates an outbound call to the reroute destination and uses the TSI matrix to pass incoming data from the call originator to the reroute destination.

Non-RLT Call Transfer

Network	Host	
SETUP	====> <i>L3L4mSETUP_IND</i> (B1) The host receives the incoming call on B-channel #1. From the <i>IISDN_CALLED_PARTY</i> info, the host determines that this call needs to be rerouted to an alternate branch office. ====> <i>L3L4mSET_TSI</i> The host cross-connects B1 and B2. src=<i>IISDNtsiLINE_A+1</i> dst=<i>IISDNtsiLINE_A+2</i> Map B1 to B2 on Span A. src=<i>IISDNtsiLINE_A+2</i> dst=<i>IISDNtsiLINE_A+1</i> Map B2 to B1 on Span A. <td></td>	
SETUP	<=== <i>L4L3mCALL_REQUEST</i> (B2) The host initiates the outbound call on B2, where the call transfer occurs.	

Network	Host	
ALERTING	==> <i>L3L4mALERTING</i> (B2)	Call setup completes normally on B-channel #2 with receipt of ALERTING and CONNECT.
CONNECT	==> <i>L3L4mCONNECT</i> (B2)	
ALERTING	<=== <i>L4L3mALERTING</i> (B1)	Now that B-channel #2 is set up, continue with normal call setup on B1, sending alerting and connect.
CONNECT	<=== <i>L4L3mCONNECT</i> (B1)	

The call is transferred by the board, but it must maintain active call setup on both B-channels for the duration of the call.

RLT Call Transfer

SETUP	==> <i>L3L4mSETUP_IND</i> (B1)	The host receives an incoming call on B-channel #1. From the <i>IISDN_CALLED_PARTY</i> info, the host application determines this call needs to be rerouted to an alternate branch office.
SETUP	<=== <i>L4L3mCALL_REQUEST</i> (B2) rtl_service = 1	The host initiates outbound call on B2, setting the rtl_service flag to '1'.
ALERTING	==> <i>L3L4mALERTING</i> (B2) includes 0x1c (FACILITY) IE with call_id.	The network sends an ALERTING message with the FACILITY info element containing the call ID for the second link.
CONNECT	==> <i>L3L4mCONNECT</i> (B2)	
FACILITY	<=== <i>L4L3mFACILITY_REQUEST</i>	The host initiates the transfer and release process by sending a FACILITY message. The call_ref and l4_ref parameters of the initial (B1) call sends a <i>L4L3mFACILITY_REQUEST</i> message, and the application populates the <i>IISDN_CALL_ID</i> field with the call_id information retrieved from the ALERTING message of the second call (B2).

Once the network establishes a direct connection between the originator of the first call and the final destination of the second call, it sends DISCONNECT messages for both B1 and B2. The calls are released on the board (and the board does not need to keep two B-channels established), but the switch maintains the actual connections between the originator and the reroute destination.

DISCONNECT=====>L3L4mDISCONNECT (B1)

DISCONNECT=====>L3L4mDISCONNECT (B2)

For more information about the messages, see *Volume 5*, in the *Dialogic® Brooktrout® Bfv APIs Reference Manual*.

Sample Application

The following code fragments show the relevant BSMI control messages used in RLT transfer.

```
/* **** */
/* **** send_call_req **** */
/* **** */
void send_call_req(int spyder_chan,int bchan,unsigned char rlt)
{
    L4_to_L3_struct      *L4L3cntlp;
    L4_to_L3_struct      msg;
    L4L3cntlp = &msg;
    /* Populate the header stuff here */
    zero_msg(L4L3cntlp);
    L4L3cntlp->lapdid = spyder_chan;
    L4L3cntlp->msgtype = L4L3mCALL_REQUEST;
    L4L3cntlp->L4_ref = 1;
    L4L3cntlp->call_ref = 0;

    /* Populate the call request stuff here */
    L4L3cntlp->data.call_req_data.bchannel = bchan;
    L4L3cntlp->data.call_req_data.interface = 0xff;
    L4L3cntlp->data.call_req_data.call_type = PRICalltyp64K;
    L4L3cntlp->data.call_req_data.rlt_service = rlt;

    L4L3cntlp->data.call_req_data.called_party.num_digits = 6;
    L4L3cntlp->data.call_req_data.called_party.num_type = PRInumtUNKNOWN;
    L4L3cntlp->data.call_req_data.called_party.num_plan = PRInumpUNKNOWN;
    L4L3cntlp->data.call_req_data.called_party.digits[0] = '6';
    L4L3cntlp->data.call_req_data.called_party.digits[1] = '0';
    L4L3cntlp->data.call_req_data.called_party.digits[2] = '3';
    L4L3cntlp->data.call_req_data.called_party.digits[3] = '8';
    L4L3cntlp->data.call_req_data.called_party.digits[4] = '9';
    L4L3cntlp->data.call_req_data.called_party.digits[5] = '8';
    L4L3cntlp->data.call_req_data.redirect_num.num_digits = 0;
    pridrv_tx_cntl_buf( L4L3cntlp);
    printf("%d: Snd L4L3mCALL_REQUEST \n", spyder_chan);
} /* end of send_call_req */
```

```

/*****
/***** Send alerting *****/
/*****
void send_alert(int spy_chan, int call_ref)
{
L4_to_L3_struct      *L4L3cntlp;
L4_to_L3_struct      msg;

    L4L3cntlp = &msg;

    zero_msg(L4L3cntlp);
    L4L3cntlp->lapdid = spy_chan;
    L4L3cntlp->L4_ref = 0;
    L4L3cntlp->call_ref = call_ref;

    printf("%d: Snd L4L3mA ALERTING REQUEST \n", spy_chan);
    L4L3cntlp->msgtype = L4L3mA ALERTING REQUEST;
if (spy_chan == NETWORK) /* we know the net is sending a facility */
    {
        L4L3cntlp->data.al_con_data.interface= 0xff;
        L4L3cntlp->data.al_con_data.bchannel= g_bchan;
        L4L3cntlp->data.al_con_data.ie_count = 1;
        L4L3cntlp->data.al_con_data.ie.ie_id = 0x1c; /* facility */
        L4L3cntlp->data.al_con_data.ie.ie_length = 0x0e;
        L4L3cntlp->data.al_con_data.ie.ie_data[0] = 0x91; /*rose*/
        L4L3cntlp->data.al_con_data.ie.ie_data[1] = 0xbe; /*rlt*/
        L4L3cntlp->data.al_con_data.ie.ie_data[2] = 0xa2; /*RR*/
        L4L3cntlp->data.al_con_data.ie.ie_data[3] = 0x09; /*length*/
        L4L3cntlp->data.al_con_data.ie.ie_data[4] = 0x02; /*id tag*/
        L4L3cntlp->data.al_con_data.ie.ie_data[5] = 0x01; /*id len*/
        L4L3cntlp->data.al_con_data.ie.ie_data[6] = 0x01; /*inv id*/
        L4L3cntlp->data.al_con_data.ie.ie_data[7] = 0x02; /*seq tag*/
        L4L3cntlp->data.al_con_data.ie.ie_data[8] = 0x01; /*seq len*/
        L4L3cntlp->data.al_con_data.ie.ie_data[9] = 0x01; /*op tag*/
        L4L3cntlp->data.al_con_data.ie.ie_data[11] = 0x01; /*op len*/
        L4L3cntlp->data.al_con_data.ie.ie_data[12] = 0x80; /*op val*/
        L4L3cntlp->data.al_con_data.ie.ie_data[13] = 0; /*callid*/
        L4L3cntlp->data.al_con_data.ie.ie_data[14] = 0;
    }
    prdrv_tx_cntl_buf( L4L3cntlp);
}

```

```

/*****
/***** Send facility *****/
/*****
/* Send L4L3mFACILITY */
void send_facility(int spy_chan, int call_ref)
{
L4_to_L3_struct      *L4L3cntlp;
L4_to_L3_struct      msg;
unsigned char*ieptr;

    L4L3cntlp = &msg;

    zero_msg(L4L3cntlp);
    L4L3cntlp->lapdid = spy_chan;
    L4L3cntlp->L4_ref = 0;
    L4L3cntlp->call_ref = call_ref;
    L4L3cntlp->msgtype = L4L3mFACILITY_REQUEST;
    printf("%d: Snd L4L3mFACILITY_REQUEST\n", spy_chan);
    L4L3cntlp->data.facility_data.call_id.len = 1;
    L4L3cntlp->data.facility_data.call_id.call_id[0] = 2;
    prdrv_tx_cntl_buf( L4L3cntlp);
}

```

Placing Calls on Hold Using BSMI

When you invoke hold functions in BSMI, the host sends an *L4L3m UNIVERSAL* message to the board with the `data.universal.msg_id` field set to a value in [Table 8](#), also detailed in Example 1. The board accepts messages from either the host (using an *L4L3m UNIVERSAL* message) and passes them to the network, or takes network messages and passes them to the host (in an *L3L4m UNIVERSAL* message).

Additional Informational Elements (IE) are added normally to the application. If sending an `MT_DL_HOLD_REJ` or an `MT_DL_RETRIEVE_REJ` message, add a CAUSE IE (see Example 2).

L3L4m UNIVERSAL messages are received with the `data.universal.msg_id` field set to a value from [Table 8](#). If IEs are contained within the message (`l34msg->data.universal.ie_count > 0`), messages are handled normally.

Table 8. Call Hold Values for L4L3mUNIVERSAL messages

<code>#define MT_DL_HOLD</code>	0x24 //ITU-T Q.932 subclass 8.1
<code>#define MT_DL_HOLD_ACK</code>	0x28 //ITU-T Q.932 subclass 8.1
<code>#define MT_DL_HOLD_REJ</code>	0x30 //ITU-T Q.932 subclass 8.1
<code>#define MT_DL_RETRIEVE</code>	0x31 //ITU-T Q.932 subclass 8.1
<code>#define MT_DL_RETRIEVE_ACK</code>	0x33 //ITU-T Q.932 subclass 8.1, must contain a CAUSE IE
<code>#define MT_DL_RETRIEVE_REJ</code>	0x37 //ITU-T Q.932 subclass 8.1, must contain a CAUSE IE
<code>#define MT_DL_STATUS</code>	0x7D //ITU-T Q.931 subclass 84.4, not used in L4L3mUNIVERSAL messages

Example 1

```
memset(&l43msg,0,sizeof(L4_to_L3_struct));
l43msg->msgtype = L4L3mUNIVERSAL;
l43msg->data.universal.msg_id      = MT_DL_RETRIEVE;
```

Example 2

```
#define IEID_CAUSE      0x08 //CAUSE IE ID code
#define EXTENSION_BIT  0x80 //extension bit for an octet
memset(&l43msg,0,sizeof(L4_to_L3_struct));
l43msg->msgtype = L4L3mUNIVERSAL;
l43msg->data.universal.msg_id      = MT_DL_RETRIEVE_REJ;
l43msg->data.universal.ie_count     = 1; //must equal all IEs
l43msg->data.universal.ie.ie_id     = IEID_CAUSE;
l43msg->data.universal.ie.ie_length = 2;
l43msg->data.universal.ie.ie_data[0] = EXTENSION_BIT | Coding_standard |
Location; //where Coding_standard and Location are from ITU-T Q.850
// subclause 2.2.2 and 2.2.3 respectively
l43msg->data.universal.ie.ie_data[1] = EXTENSION_BIT | Cause ; //where Cause
is from ITU-T Q.850 subclause 2.2.5
```

For more information on *L4L3mUNIVERSAL* and *L3L4mUNIVERSAL*, see *Volume 5*, in the *Dialogic® Brooktrout® Bfv APIs Reference Manual*.

6 - Managing Fax and Voice over IP Sessions

This chapter describes how to develop applications that use the internet for fax and voice media.

To establish Internet Protocol (IP) sessions, Dialogic uses the Session Initiation Protocol (SIP) and the H.323 Protocol.

To manage fax and voice media, Dialogic uses T.38 and RTP protocols. This chapter has the following sections:

- [*Managing Calls Using IP Telephony on page 208*](#)
- [*Failover Based on Telephony Cause Codes on page 228*](#)
- [*Configuring T.38, RTP and IP Call Control Activities on page 239*](#)
- [*Troubleshooting on page 240*](#)
- [*Understanding the SIP Protocol on page 241*](#)
- [*Using Third Party IP Stacks on page 254*](#)
- [*SR140 Internet Aware Fax \(IAF\) Support over UDP on page 262*](#)

Managing Calls Using IP Telephony

The Bfv API supports fax functionality over IP networks using the Session Initiation Protocol (SIP) and the H.323 protocol as well as the PSTN network (using ISDN, RBS, R2 MFC, and analog loop start call control protocols) through a common and consistent programming interface. This flexibility helps you to develop fax applications that can place and receive calls over traditional PSTN and IP transports using the TR1034-N or Brooktrout SR140.

Note: Be aware of the following regarding IP calls:

- H.323 (fast or slow start) or SIP is used to establish calls over the IP network.
- The TR1034-N and Brooktrout SR140 support T.38, G.711 fax pass-through, and audio calls.
- The TR1034 board module supports T.38, G.711 fax pass-through, and audio calls on some models
- Brooktrout SR140 and TR1034 both use RTP sessions before the T.38 transmission is established. The RTP contains the CED/ANSam or CNG tones used to establish a fax call.

All channels on a given module must be configured for IP call control or for PSTN call control. Dialogic does not support combined modes of call control on a given module. Multiple modules within a system can be configured to support several modes of call control, each module being configured to support only one mode.

For proper operation, the Brooktrout SR140 or the TR1034 Ethernet interface (the one used for IP call control) should be wired to a common hub or switch. [Figure 19 on page 209](#) shows a typical network wiring configuration.

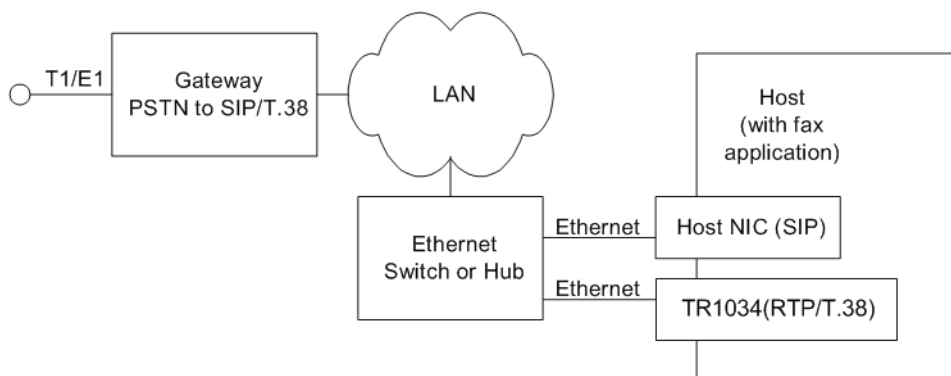


Figure 19. SIP Configuration Model

With dual-purpose modules in place, end-users can choose between PSTN or IP mode. Your applications can determine whether the module supports the IP or PSTN mode either at runtime or installation.

Adding IP Call Control using the Bfv API

Because the Dialogic® Brooktrout® Bfv API uniformly supports a wide set of transports including IP, you can use the same applications to manage calls over IP as when using the PSTN. You can configure IP call control using the Dialogic® Brooktrout® Configuration Tool (a GUI product) or by setting values in call control configuration files.

Outgoing IP Calls

Your application manages outgoing calls if it performs the following:

- Uses the function calls from the table below
- Does not do syntax validation of the dial string
- Relies on the return status from the *BfvLineOriginateCall* function to determine call completion success.

In this case the application is unaware of the mode of transport (PSTN or IP):

BfvCallDisconnect	Starts the process of terminating a telephone call.
BfvCallSetup	Starts the process of dialing an outgoing telephone call.
BfvCallStatus	Retrieves the channel's current call state.
BfvLineDialString	Places the line in an OFF_HOOK state, dials the digits specified, and returns after dialing the last digit.
BfvLineOrigCallIDB	Checks the specified dialing database for the specified telephone number, returns the amount of time to wait before dialing, and then places the call on an outgoing line and updates the dialing database.
BfvLineOriginateCall	Dials an outgoing call (equivalent to the BfvCallSetup and BfvCallWaitForComplete lower level function calls.)
BfvLineTerminateCall	Terminates the current call (equivalent to the BfvCallDisconnect and BfvCallWaitForRelease lower level function calls.)
BfvLineTransfer CapabilityQuery	Indicates the transfer capability of a channel. Also provides an application with information to determine whether two particular lines are paired to perform a two B-channel call transfer. (Not supported on IP).
BfvWaitForComplete	Waits for the outgoing telephone call to finish.
BfvWaitForRelease	Waits for the termination of a telephone call to finish.

See [Sample INVITE Request on page 215](#) for more information about using Bfv function calls.

Incoming IP Calls

Your application can receive incoming IP calls if it uses function calls from the table below

In this case the application is unaware of the mode of transport (PSTN or IP):

BfvCallAccept	Starts answering an incoming telephone call.
BfvCallDisconnect	Starts the process of terminating a telephone call.
BfvCallReject	Rejects an incoming telephone call.
BfvCallRingDetect	Enables or disables the detection of incoming phone calls.
BfvCallStatus	Retrieves the channel's current call state.
BfvCallWaitForAccept	Finishes the process of answering an incoming telephone call.
BfvCallWaitForSetup	Waits for an incoming call and returns all available information about the call to the application.
BfvLineAnswer	Answers incoming call (equivalent to the BfvCallAccept lower level function call).
BfvLineTerminateCall	Terminates the current call (Equivalent to the lower level BfvCallDisconnect and BfvCallWaitForRelease lower level function calls).
BfvLineTransferCapability Query	Indicates the transfer capability of a channel. Also provides an application with information to determine whether two particular lines are paired to perform a two B-channel call transfer. (Not supported on IP.)
BfvLineWaitForCall	Waits for incoming call (equivalent to the BfvCallWaitForSetup lower level function call).
BfvWaitForRelease	Waits for the termination of a telephone call to finish.

Channels are either PSTN or IP, but not both. You can not have PSTN and IP on the same channel. If the first module is IP, then the first set of channels is IP. If the first module is PSTN, then the first set of channels is PSTN. The first module defined by lowest module number, for example, Module 2 gets channels assigned first, then Module 3 gets channels assigned next, and so forth.

Understanding SIP Functionality

The following section provides information about SIP functionality and processes. For a detailed introduction to the SIP protocol, see [Introduction to the SIP Protocol on page 241](#).

Using a SIP Proxy Server

To make an outgoing connection using IP, your application must know the IP address of a local proxy server which is responsible for forwarding the SIP call towards its final destination. In a typical environment, the proxy server is the local IP to PSTN gateway. For more information about using proxy servers, see [Understanding the SIP Protocol on page 241](#).

To receive incoming calls using the internet, you must register your location so that proxy servers can locate you.

For more information about configuring proxy servers, see the installation and configuration guide that came with your software and the *Dialogic® Brooktrout® Dialogic® Brooktrout® Bfv API Reference Manual, Volume 6, Appendix A*.

Verifying Dialed Strings

If your application either does syntax verification/modification of the dialed string or it is desirable to do number translation or lookup without modifying the application, then the application needs to perform number translation.

Channel numbers enable your application to differentiate between types of call in the context of the Call Control Bfv API. SIP and other IP protocols use a Uniform Resource Identifier (URI). The software detects and recognizes the format of the dial string and sends the call to the appropriate channel. In Brooktrout SDK 4.0 and beyond, the dial strings follow the URI syntax, so tel: and fax formats are not supported and must be reformatted by your application.

Prefixes to dial strings are provided as a means for your application to behave intelligently by supporting channel selection in hybrid systems. Prefixes are optional.

Using prefixes, dial strings can also be pre-qualified by your application as in the following example:

SIP/T38://xxxxx@brooktrout.com

SIP/T38://+01-781-555-1212

ISDN://+01-781-449-9009

When the current *fax.c* sample program is compiled, it takes command line parameters to define the number dialed. With no changes to the source code, you can replace the number with a URI to allow it to take advantage of T.38. The sample program is effectively IP-enabled purely by virtue of the dial string it uses.

In a SIP environment, the following command:

```
fax -u 0 -s xxxxx@brooktrout.com foo.pkt
```

initiates the following sequence of events:

1. Call control processes the URI dial string.
2. Sends it to the SIP protocol stack.
3. The SIP protocol stack sends it to the SIP redirect server.
4. The SIP redirect server tells the SIP stack where calls to brooktrout.com should go.

Note: Redirect and proxy servers are optional. You can send an INVITE directly to a user client.

5. SIP stack places the call there.
6. Remote SIP proxy optionally redirects call to where it ought to go.

If the phone number string contains a prefix or characters followed by a colon(:) and the prefix is not SIP, the software sends a return status indicating that the number the application dialed was invalid.

The following are valid dial string examples for a SIP channel.
Comments are shown in *italics*.

Joe Smith<sip:Joe@somewhere.com>	<i>SIP endpoint address.</i>
sip:800-555-1212@somewhere.com	<i>Endpoint gateway specified.</i>
sip:800-555-1212@myproxy.com	<i>Proxy explicitly specified.</i>
800-555-1212@Somewhere.com	<i>Endpoint gateway specified.</i>
800-555-1212@myproxy.com	<i>Proxy explicitly specified.</i>
800-555-1212	<i>Will use default proxy server.</i>
+1(800)555-1212	<i>Will use default proxy server.</i>
192.168.1.45	<i>Valid, but not recommended.</i>
Joe@192.168.1.1	<i>Valid, but not recommended.</i>
sip:somewhere.com	<i>User part (left side of '@') is implied.</i>
sip:joe@somewhere.com:9876	<i>An explicit port specification.</i>
sip:011442871234@somewhere.com;user=phone	<i>Specifies that user part is a phone number.</i>

Using prefixed dial strings would have the same effect, as in:

```
fax -u 0 -s sip:xxxxx@brooktrout.com foo.pkt
```

Using an alternative URI scheme:

```
fax -u 0 -s 781-555-1212@cisco-gw.brooktrout.com foo.pkt
```

➤ **initiates the following sequence of events:**

1. Call Control processes the URI dial string, sends it to the SIP protocol stack.
 2. SIP stack places the SIP call to the Cisco XXXX Gateway.
 3. The Cisco Gateway places the call on the PSTN, sending call progress information to the SIP stack and connect the call with a RTP stream.
 4. When the Cisco Gateway identifies the call as a fax call it will then send a T.38 Re-INVITE
 5. A 200 OK is sent to connect the T.38 stream.
 6. The fax is then sent and the call is torn down.
- No SIP proxy or redirect server is required if not configured or needed for point to point calls.

Again, using prefixed dial strings would have the same effect as in:

```
-u 0 -s sip:781-433-9454@cisco-gw.brooktrout.com foo.pkt
```

Sample INVITE Request

The following sample uses the Bfv API to send the INVITE request.

CALL CONTROL CONFIGURATION FILE

```
1314_trace=none
  1413_trace=none
  api_trace=none
  internal_trace=none
  host_module_trace=none
  ip_stack_trace=none
  trace_file=c:\Brooktrout\Boston\config\ecc.log
  max_trace_files=1
  max_trace_file_size=10
[module.6]
  channels=60
  set_api=bfv
[module.6/clock_config]
  clock_mode=master
  clock_source=internal
[module.6/ethernet.1]
  dhcp=disabled
  ip_address=10.50.50.50
  ip_netmask=255.255.255.0
  ip_gateway=10.50.50.1
  ip_broadcast=0.0.0.0
  media_port_min=56000
  media_port_max=56999
  ethernet_speed=auto
  ethernet_duplex=half
  ethernet_flow_control=auto
  ip_arp_timeout=10
[module.6/port.1]
  port_config=inactive
[module.6/port.2]
  port_config=inactive
[module.6/host_cc.1]
  host_module=1
  number_of_channels=60
[host_module.1]
  module_library=brktsip.dll
  enabled=true
[host_module.1/parameters]
  sip_max_sessions=256
  sip_Max-Forwards=70
  sip_From=Anonymous <sip:no_from_info@anonymous.invalid>
  sip_Contact=0.0.0.0:0
  sip_ContactV6=
  sip_username=user@brooktrout.com
  sip_session_name=session_name
```

```
sip_session_description=session_description
sip_description_URI=user@brooktrout.com
sip_email=email@brooktrout.com
sip_phone=+1-4085551212
sip_user_agent=Brktsip/6.4.0B2 (Dialogic)
[host_module.1/t38parameters]
t38_fax_rate_management=transferredTCF
fax_transport_protocol=t38_only
t38_fax_udp_ec=t38UDPRedundancy
rtp_ced_enable=true
t38_max_bit_rate=14400
t38_fax_version=0
media_passthrough_timeout_inbound=1000
media_passthrough_timeout_outbound=4000
media_renegotiate_delay_inbound=1000
media_renegotiate_delay_outbound=-1
t38_fax_fill_bit_removal=false
t38_fax_transcoding_jbig=false
t38_fax_transcoding_mmr=false
t38_t30_fastnotify=false
t38_type_of_service=0
t38_UDPTL_redundancy_depth_control=5
t38_UDPTL_redundancy_depth_image=2
[host_module.1/rtp]
rtp_frame_duration=20
rtp_jitter_buffer_depth=100
rtp_codec=pcmu pcma
rtp_silence_control=inband
t38_offer_as_ced=true
rtp_type_of_service=0
rtp_voice_frame_replacement=0
```

BFV APPLICATION

```
BT_ZERO(args);
args.phonenum = "201021000@10.50.50.21"
args.call_protocol_code = CALL_PROTOCOL_FAX
BfvLineOriginateCall(lp, &args);
```

SIP INVITE

```
1  INVITE sip:200021000@10.50.50.21 SIP/2.0
2  From: <sip:user@brooktrout.com>;tag=87d05c0-0-13c4-55013-6b38-24b4c0c9-6b38
3  To: <sip:200021000@10.50.50.21>
4  Call-ID: 8b9eb30-0-13c4-55013-6b38-5b0d867b-6b38
5  CSeq: 1 INVITE
6  Via: SIP/2.0/UDP 10.50.50.100:5060;branch=z9hG4bK-6b38-1a2d3d2-790c92e7
7  Supported: timer
8  Max-Forwards: 70
9  User-Agent: Brktsip/6.4.0B2 (Dialogic)
10 Contact: <sip:10.50.50.100>
11 Content-Type: application/sdp
12 Content-Length: 295
13 v=0
14 o=user@brooktrout.com 2209016249 0542990000 IN IP4 10.50.50.100
15 s=session_name
16 i=session_description
17 e=email@brooktrout.com
18 p=+1-4085551212
19 t=0 0
20 m=audio 56000 RTP/AVP 0
21 c=IN IP4 10.50.50.50
22 a=rtpmap:0 pcmu/8000
23 m=audio 56000 RTP/AVP 8
24 c=IN IP4 10.50.50.50
25 a=rtpmap:8 pcma/8000
INVITE sip:10.50.50.100:5060 SIP/2.0
Via: SIP/2.0/UDP 10.50.50.21:5060;branch=z9hG4bK026C8
Remote-Party-ID:
    <sip:201021000@10.50.50.21>;party=calling;screen=no;privacy=off
From: <sip:201021000@10.50.50.21>;tag=913F0-1145
To: <sip:user@brooktrout.com>;tag=87d05c0-0-13c4-55013-6b38-24b4c0c9-6b38
Date: Tue, 26 Apr 2011 23:36:18 GMT
Call-ID: 8b9eb30-0-13c4-55013-6b38-5b0d867b-6b38
Supported: 100rel,timer,resource-priority,replaces,sdp-anat
Min-SE: 1800
Cisco-Guid: 3452201248-1871974880-2147975719-3150639659
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY,
    INFO, REGISTER
CSeq: 101 INVITE
Max-Forwards: 70
Timestamp: 1303860978
Contact: <sip:201021000@10.50.50.21:5060>
Expires: 180
Allow-Events: telephone-event
```

```
Supported: precondition
Content-Type: application/sdp
Content-Length: 459
04/26 16:27:49.48 ECC Stack: INFO    -  TRANSPORT    -
v=0
o=CiscoSystemsSIP-GW-UserAgent 6809 6043 IN IP4 10.50.50.21
s=SIP Call
c=IN IP4 10.50.50.21
t=0 0
m=image 17414 udptl t38
c=IN IP4 10.50.50.21
a=T38FaxVersion:0
a=T38MaxBitRate:14400
a=T38FaxFillBitRemoval:0
a=T38FaxTranscodingMMR:0
a=T38FaxTranscodingJBIG:0
a=T38FaxRateManagement:transferredTCF
a=T38FaxMaxBuffer:200
a=T38FaxMaxDatagram:180
a=T38FaxUdpEC:t38UDPRedundancy
m=audio 0 RTP/AVP 8
c=IN IP4 10.50.50.21
a=rtpmap:8 PCMA/8000

26 SIP/2.0 200 OK
27 From: <sip:201021000@10.50.50.21>;tag=913F0-1145
28 To: <sip:user@brooktrout.com>;tag=87d05c0-0-13c4-55013-6b38-24b4c0c9-6b38
29 Call-ID: 8b9eb30-0-13c4-55013-6b38-5b0d867b-6b38
30 CSeq: 101 INVITE
31 Via: SIP/2.0/UDP 10.50.50.21:5060;branch=z9hG4bK026C8
32 Supported: timer
33 User-Agent: Brktsip/6.4.0B2 (Dialogic)
34 Contact: <sip:10.50.50.100>
35 Content-Type: application/sdp
36 Content-Length: 407
37 v=0
38 o=user@brooktrout.com 2209016255 0989137000 IN IP4 10.50.50.100
39 s=session_name
40 i=session_description
41 e=email@brooktrout.com
42 p=+1-4085551212
43 t=0 0
44 m=image 56004 udptl t38
45 c=IN IP4 10.50.50.50
46 a=T38FaxVersion:0
47 a=T38MaxBitRate:14400
48 a=T38FaxRateManagement:transferredTCF
```

```
49 a=T38FaxMaxBuffer:200
50 a=T38FaxMaxDatagram:72
51 a=T38FaxUdpEC:t38UDPRedundancy
52 m=audio 0 RTP/AVP 8
53 c=IN IP4 10.50.50.50
```

SIP INVITE DESCRIPTION

- 1 Defined by args.phonenum field passed to BfvLineOriginateCall().
- 2 Defined by the sip_From setting in the call control configuration file.
- 3 Defined by args.phonenum field passed to BfvLineOriginateCall().
- 4 Defined by the current domain of the UAC.
- 5 Generated internally.
- 6 Defined by the IP of the primary NIC (network interface controller).
- 7 Generated internally.
- 8 Defined by the sip_Max-Forwards setting in the call control configuration file.
- 9 Defined by the sip_user_agent setting in the call control configuration file.
- 10 Defined by the sip_Contact setting in the call control configuration file (default is the IP address of the primary NIC).
- 11 Generated internally.
- 12 Generated internally.
- 13 Generated internally.
- 14 Defined by the sip_username setting in the call control configuration file and IP address of primary NIC.
- 15 Defined by the sip_session_name setting in the call control configuration file.
- 16 Defined by the sip_session_description setting in the call control configuration file.
- 17 Defined by the sip_email setting in the call control configuration file.
- 18 Defined by the sip_phone setting in the call control configuration file.
- 19 Generated internally.
- 20 Defined by the media_port_min and media_port_max settings in the call control configuration file.
- 21 Defined by the ip_address setting in the call control configuration file.
- 22 Generated internally.
- 23 Defined by the media_port_min and media_port_max settings in the call control configuration file.
- 24 Defined by the ip_address setting in the call control configuration file.
- 25 Generated internally.
- 26 Generated internally in response to the SIP INVITE.
- 27 Generated from the received SIP INVITE From header
- 28 Generated from the received SIP INVITE To header
- 29 Generated from the received SIP INVITE Call-ID header
- 30 Generated from the received SIP INVITE CSeq header
- 31 Generated from the received SIP INVITE CSeq header
- 32 Generated internally.
- 33 Defined by the sip_user_agent setting in the call control configuration file.
- 34 Defined by the sip_Contact setting in the call control configuration file (default is the IP address of the primary NIC).
- 35 Generated internally.
- 36 Generated internally.
- 37 Generated internally.
- 38 Defined by the sip_username setting in the call control configuration file

-
- and IP address of primary NIC.
- 39 Defined by the sip_session_name setting in the call control configuration file.
 - 40 Defined by the sip_session_description setting in the call control configuration file.
 - 41 Defined by the sip_email setting in the call control configuration file.
 - 42 Defined by the sip_phone setting in the call control configuration file.
 - 43 Generated internally.
 - 44 Defined by the media_port_min and media_port_max settings in the call control configuration file.
 - 45 Defined by the ip_address setting in the call control configuration file.
 - 46 Defined by the common value in the INVITE and by the t38_fax_version settings in the call control configuration file.
 - 47 Defined by the common value in the INVITE and by the t38_max_bit_rate settings in the call control configuration file.
 - 48 Defined by the common value in the INVITE and by the t38_fax_rate_management settings in the call control configuration file.
 - 49 Defined by the common value in the INVITE and by the t38_fax_max_buffer settings in the call control configuration file.
 - 50 Defined by the common value in the INVITE and by the t38_fax_max_datagram_rcv settings in the call control configuration file.
 - 51 Defined by the common value in the INVITE and by the t38_fax_udp_ec settings in the call control configuration file.
 - 52 Defined by the INVITE and port set to 0 to disable
 - 53 Defined by the ip_address setting in the call control configuration file.

Call Progress Values

Brooktrout Bfv API has mapped all possible IP call INVITE responses to Bfv call progress values. See [Table 9](#) for values for functions that provide a final call progress value:

Table 9. Mapping of SIP responses to Bfv FCP values

SIP response code	Bfv final call	Progress code
486	Busy here	FCP_BUSY1
600	Busy everywhere	FCP_ROBUSY
503	Service Unavailable	FCP_SITINTC
180	Ringing	FCP_RING1
200	OK (when in response to an INVITE when SDP media type and subtypes are image and T.38 respectively.)	FCP_ANSWER_TONE_DETECT

SIP Options

SIP User Agents such as SIP Gateways or Session Border Controllers (SBC) are sometimes configured to transmit OPTIONS requests to other SIP endpoints. This is done in an attempt to query the operational status of SIP entities that the Gateway or SBC is configured to communicate with.

If a SIP endpoint fails to respond to the OPTIONS request or if the SIP endpoint returns a failure response, the Gateway or SBC can designate the SIP endpoint as unreachable or out-of-service and re-route SIP calls as appropriate.

The Dialogic Brooktrout SIP stack responds to incoming OPTIONS requests automatically with a 200 (OK) response. The 200 (OK) response transmitted by the Dialogic Brooktrout SIP stack contains the following **Allow** header:

Allow: INVITE, ACK, CANCEL, BYE OPTIONS

In addition, the 200 (OK) response also contains the following **Accept** header:

Accept: application/sdp

The ability to transmit SIP OPTIONS requests is not supported by the Dialogic Brooktrout SIP stack implementation.

Understanding H.323 Functionality

This Brooktrout SDK supports the H.323 protocol (version 4 and Annex D), providing end point functionality only. Our current H.323 implementation does not operate as a gateway or H.323 Gatekeeper, but you can configure your application to communicate with a gateway or H.323 Gatekeeper.

The H.323 implementation supports both a primary and alternate H.323 Gatekeeper. When an application is configured to communicate with a Gatekeeper, all the RAS messages exchanged with the Gatekeeper are done inside of the H.323 implementation without requiring the application to get involved. The H.323 implementation registers the end points transport address and alias information with the Gatekeeper and resolves destination alias addresses with the Gatekeeper. If the application turns on alternate Gatekeeper support and the primary Gatekeeper becomes unavailable, the H.323 implementation automatically falls back to an alternate Gatekeeper without involving the application.

Multiple Protocols

SIP and H.323 can co-exist in the same system. You can configure a single hardware module or Brooktrout SR140 module to support either SIP or H.323, selecting which protocol stack to use on a per-module basis. The protocols can operate simultaneously on different modules.

Supporting Media

The H.323 implementation supports T.38 (Fax Media) and RTP media over UDP (H.323 Annex D).

The H.323 implementation supports the ability to negotiate a pseudo G.711 media session with the intent of renegotiating to a T.38 media session even when RTP prompt playback/record capability is not desired.

This functionality is required to interoperate with certain supported equipment. Media renegotiation to T.38 is also supported after a call is setup.

The H.323 implementation supports basic call control functionality, specifically outbound call setup and teardown and inbound call detection, answering, and teardown. See the sections on [Outgoing IP Calls on page 210](#) and [Incoming IP Calls on page 211](#) for information about the functions that support these activities.

Using H.323 Address Forms

To allow applications to work for both SIP and H.323, the dialstring field for *BfvLineOriginateCall* and *BfvCallSetup* accepts a format of `phone_number@ip_address` (a current SIP format).

The phone number is an optional field as is the port in the `ip_address`. The existing H.323 dialstrings remain unchanged and are still supported as described below. This option provides a unified way of placing a non-gatekeeper call that works for both SIP and H.323.

Examples of supported dialstrings for H.323:

```
1234@208.242.99.10
1234@208.242.99.10:1720
```

Dialogic products accept the following H.323 Called Party Address forms:

- A transport address
- An E.164 alias
- A H.323 ID alias

Unless noted, the rules below also apply to an H.323 Calling Party Address.

Transport Address (IP Address)

This Called Party Address must start with the identifier “TA:” followed by the transport address.

A transport address consists of an IP address followed by the “:” character followed by a port number or another H.323 address.

A port number is not required and, if it is not specified, then the software uses the default H.323 port of 1720.

IP address followed by a port number example:

```
TA:198.133.219.25:1721
```

IPv6 addresses are also supported and need to be in hexadecimal notation, inside brackets. For example;

[HHHH:HHHH:HHHH:HHHH:HHHH:HHHH:HHHH:HHHH]

E.164 Alias (Phone Number)

This Called Party Address can start with the optional identifier “TEL:”, followed by one or more E.164 address destinations. Turn on H.323 Gatekeeper support when using E.164 aliases.

An E.164 alias is a phone number that is up to 128 characters long and includes the characters 0 – 9, *, #. You can set the terminal's E.164 alias using the *h323_e164alias* parameter in the call control configuration file. You can specify multiple aliases, each starting on a new line using the same parameter name.

E.164 alias example:

TEL:7814494100

or

7814494100

Extension and Subaddress information is related to E.164 addresses. Extension and Subaddress are not supported for H.323 Calling Party Addresses.

An Extension starts with the identifier “EXT:” followed by an E.164 address or the identifier “EXTID:” followed by an H.323 ID. H.323 IDs are explained in the next section.

A Subaddress starts with the identifier “SUB:” followed by an E.164 address.

H.323 ID Alias (Name)

This Called Party Address must start with the identifier “NAME:” followed by an H.323 ID. Turn on H.323 Gatekeeper support when using H.323 ID aliases.

An H.323 ID is a text string that is up to 256 Unicode characters long. You can set the terminal's H.323 ID alias using the *h323_h323IDalias* parameter in the call control configuration file. You can specify multiple aliases, each starting on a new line using the same parameter name.

H.323 alias example:

NAME:JohnSmith

Different types of H.323 Called Party Addresses are joined together to create a hybrid address by using the delimiting character. The delimiting character is the comma. Example of reaching the destination 7814494100 by going through an H.323 gateway located at IP address 198.133.219.25:

TA:198.133.219.25,TEL:7814494100

or

TA:198.133.219.25,7814494100

Failover Based on Telephony Cause Codes

Overview

Identifying call failures within an IP telephony network allows an application to re-route calls depending on the failure. In many cases, the failover to other network devices occurs seamlessly when the network has the following:

- H.323 Gatekeepers
- SIP Registrar Servers or SIP Redirect Servers

In cases where the network does not have these components, the failover responsibility falls on the application.

Typically, in the absence of gatekeepers or other call routing devices, the telephony endpoint, under the control of the application, is in direct contact with the gateway. In this network configuration, the application must be able to identify the various types of failures in order to effectively manage the call routes.

Common Failures

The following are some common failures:

- Unreachable IP address
- Reachable IP address with no SIP/H.323 response
- Gateway SIP/H.323 channels exceeded
- Gateway PSTN channels exceeded
- Gateway with responding SIP/H.323 and PSTN down (no PSTN cable connected)
- Gateway with responding SIP/H.323 and PSTN down due to an alarm
- Gateway with responding SIP/H.323 and PSTN up but not waiting for calls
- Gateway failure during an active call by disconnecting the network cable on Brooktrout SR140
- Gateway failure during an active call by disconnecting the network cable on the gateway

-
- Gateway failure during an active call by disconnecting the PSTN cable.

Applications needing specific information about a call failure can use the cause codes reported by *BfvLineTerminateCall*. However, using cause codes to determine failover scenarios is complicated. In most cases, the cause codes supplied by the gateway are manufacturer-specific and depend on the protocol being used.

Therefore, if you use any particular code in determining a failover scenario (where re-routing should occur), you should consider the context including the protocol and the gateway manufacturer.

Failover Scenarios

Refer to [Table 10](#) to determine failover scenarios. The table shows cause codes that have been found uniformly consistent with conditions requiring re-routing.

Note that cause codes 18 and 1000 are for cases where the gateway is unreachable or inoperative.

The cause codes in this table are suggested based on data collected from various Gateways (shown in [Table 11](#)). Entries with “-” specify scenarios for which no data was collected.

Table 10. Failover Cause Codes

Description	Cause Code
No user responding	18
Call Rejected	21
Destination out of order	27
Network out of order	38
Temporary failure	41
Requested circuit channel not avail	44
Service/optio not avail; unavail; unspecified	63
No dial tone	1000

Known Failures From Various Gateways

H.323 and SIP

In [Table 11](#), the entries are divided between H.323 and SIP. In the case of H.323, cause codes are sent by the gateway as values corresponding to Q.931 error codes. These are returned unaltered by *BfvLineTerminateCall*. In the case of SIP, the gateway sends SIP error codes which are then translated by *BfvLineTerminateCall* into Q.931 error codes before being returned.

Using cause codes to determine failover scenarios is complicated. In most cases, the cause codes supplied by the gateway are manufacturer-specific and depend on the protocol being used. Applications needing specific information about a call failure can use the cause codes reported by *BfvLineTerminateCall* or *BfvCallWaitForComplete()*. Note when the Bfv API function returns `BT_STATUS_TIMEOUT`, the cause code will not be valid.

Table 11. Known Failover Cause Code Data

Scenario	SR140 Protocol	GnuGK	Alcatel	Avaya	SIP Control	Cisco 2821	CCM6.01	CCM6.1	Quintum	IM1010
<i>H.323</i>										
Unreachable IP Address	H.323	1000	1000	1000	1000	1000	1000	1000	1000	1000
Reachable IP address with no SIP/H.323 responds	H.323	1000	1000	1000	1000	1000	1000	1000	1000	1000
Gateway failure during an active call by disconnecting the network cable on the Brooktrout SR140	H.323	0, 16	0, 16	0, 16	0, 16	0, 16	0, 16	0, 16	0, 16	0, 16
Gateway failure during an active call by disconnecting the network cable on the gateway	H.323	0	0	0	0	0	0	0	0	0
Gateway with responding SIP/H323 and PSTN down (no PSTN cable connected)	H.323	1000	27	34	-	1000	1	1	17	-
Gateway with responding SIP/H.323 and PSTN down due to an alarm	H.323	17	27	34	-	-	-	-	0	-
Gateway SIP/H.323 channels exceeded	H.323	1000	34	1000	-	1000	63, 41	44	1000	-
Gateway PSTN channels exceeded	H.323	17	-	-	-	-	-	-	-	-
Gateway failure during an active call by disconnecting the PSTN cable	H.323	0	0	38	-	34	34	34	16	-
Gateway with responding SIP/H.323 and PSTN up but not waiting for calls	H.323	17	27	34	-	44	44	44	41	-

Scenario	SR140 Protocol	GnuGK	Alcatel	Avaya	SIP Control	Cisco 2821	CCM6.01	CCM6.1	Quintum	IM1010
<i>SIP</i>										
Unreachable IP address	SIP	18	18	18	18	18	18	18	18	18
Reachable IP address with no SIP/H.323 responds	SIP	41	41	41	41	41	41	41	41	41
Gateway failure during an active call by disconnecting the network cable on Brooktrout SR140	SIP	41	41	41	41	41	41	41	41	41
Gateway failure during an active call by disconnecting the network cable on the gateway	SIP	18	18	18	18	18	18	18	18	18
Gateway with responding SIP/H.323 and PSTN down (no PSTN cable connected)	SIP	-	41	-	41	1	1	1	17	41
Gateway with responding SIP/H.323 and PSTN down due to an alarm	SIP	-	41	-	-	-	-	-	-	41
Gateway SIP/H.323 channels exceeded	SIP	-	21	-	41	41	41	41	17	-
Gateway PSTN channels exceeded	SIP	-	-	-	-	-	-	-	0	41
Gateway failure during an active call by disconnecting the PSTN cable	SIP	-	0	-	16	16	16	16	16	16
Gateway with responding SIP/H.323 and PSTN up but not waiting for calls	SIP	-	41	-	41	41	41	41	41	41

SIP to Q.931 Conversion

Table 12 shows the translation performed to create Q.931 codes from SIP error codes.

Table 12. SIP to Q.931 Conversion

SIP Cause Code	Description	Q.931 Cause Code
200	OK	16
400	Bad Request	41
401	Unauthorized	21
402	Payment required	21
403	Forbidden	21
404	Not found	1
405	Method not allowed	63
406	Not acceptable	79
407	Proxy authentication required	21
408	Request timeout	102
409	Conflict	41
410	Gone	22
413	Request entity too long	127
414	Request-URI too long	127
415	Unsupported media type	79
416	Unsupported URI Scheme	127
420	Extension required	127
423	Interval too brief	127
480	Temporarily unavailable	18
481	Call leg/transaction does not exist	41

SIP Cause Code	Description	Q.931 Cause Code
482	Loop detected	25
484	Address incomplete	28
485	Ambiguous	1
486	Busy here	17
487	Request Cancelled	21
488	Not acceptable here	31
500	Internal server error	41
501	Not implemented	79
502	Bad gateway	38
503	Service unavailable	41
504	Server time-out	102
505	Version not supported	127
513	Message Too Large	127
600	Busy everywhere	17
603	Decline	21
604	Does not exist anywhere	1
606	Not acceptable	31

Processing Media Using the T.38 Protocol

The Brooktrout SDK supports real-time sending and receiving faxes over IP following the T.38 protocol for exchanging messages and data through IP fax gateways or IAF devices over an IP network.

Because Brooktrout's Bfv Call Control API uniformly supports a wide set of transports including IP, you can use the same fax functions for fax over IP as when sending faxes over the PSTN. You can configure your system using the Brooktrout Configuration Tool (a GUI product) or by editing user-defined files.

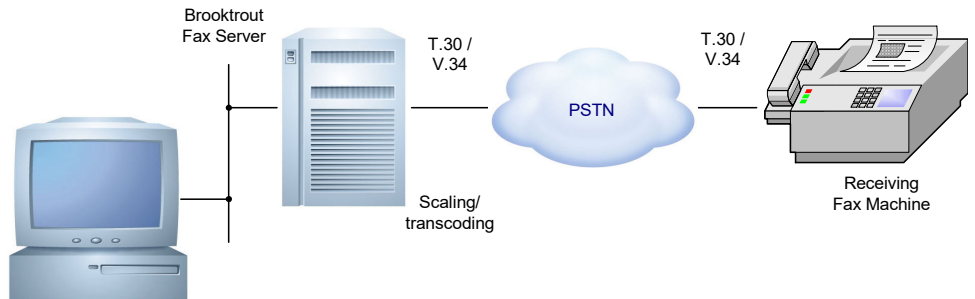
Using standard fax machines over the PSTN, the following information is presented.

- The sender of a fax gets immediate notification of a fax being sent successfully.
- The receiver gets information on the sender's telephone number and the time the fax was received.

Bfv's IP fax support provides this information as well.

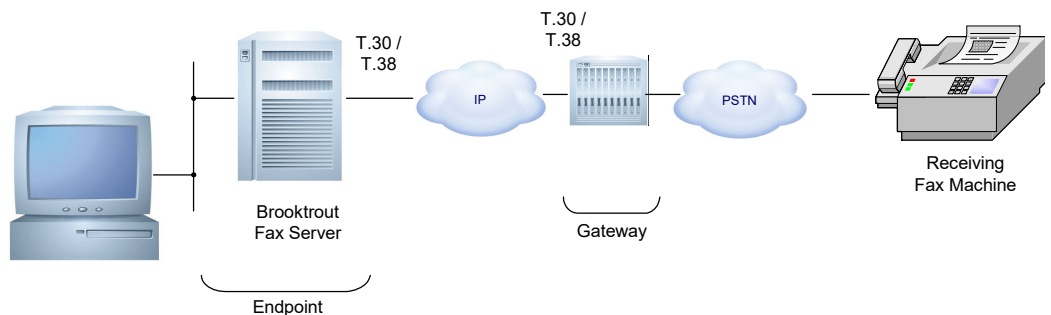
Traditionally, faxing has been done over the TDM telephone network, as defined in the ITU specification T.30. In a traditional PSTN based T.30-only fax transmission, sending a fax requires three fax components:

T.30 Protocol Engine	While T.30 is a mature technology, an effective T.30 implementation is complex due in large part to the challenge of connecting with the installed base of 120 million fax machines whose own compliance to the standard varies considerably.
Image Conversion Engine	The sending device must adapt (scale and transcode) the image to the capabilities of the receiver. The receiver must check the received file for errors and try to correct ones that occur.
Modem	This is the vehicle to carry the protocol and image data across the PSTN network.



While the method of transport is different, IP environments support the functionality of these elements.

There are two types of devices used to implement T.38: an endpoint and a T.38-aware gateway. TR1034-based applications form endpoints. The following diagram shows how they support a likely IP fax scenario:



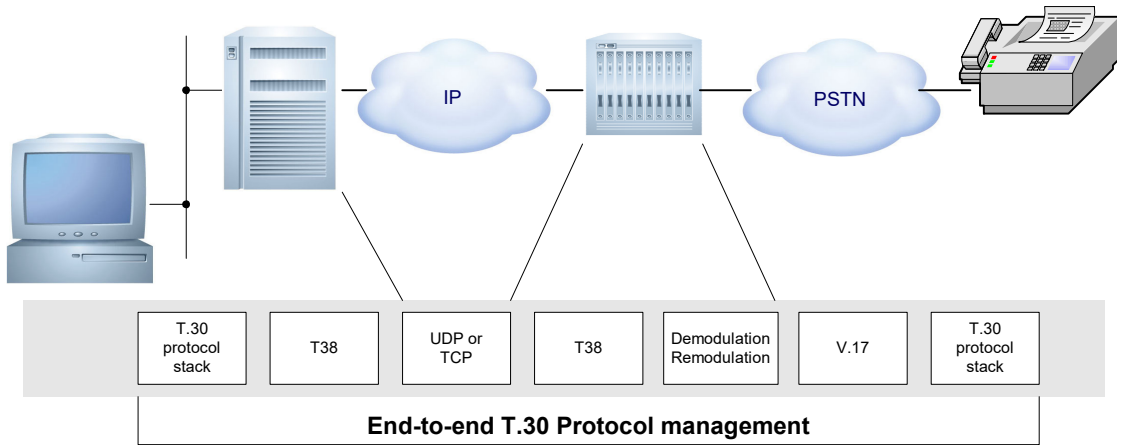
The endpoint uses the T.30 protocol to negotiate the connection and performs the image conversion. The gateway simply passes the fax between a PSTN and IP connection.

In endpoint facsimile devices, such as a network fax server, the T.38 protocol provides the equivalent to the modem in traditional faxing. In a gateway, the T.38 protocol is used to translate T.30 protocol and image data from the modems in the gateway to and from the IP endpoint connection, using the following procedure.

1. With T.38 in an endpoint, the application connects a T.38 fax server to an IP network and transmits the T.30 protocol and fax image data to the receiving gateway using T.38 packets over the IP network.
2. The receiving T.38 gateway, in turn, translates the T.38 packets and repackages them into T.30 protocol signals and transfers them to the receiving fax machine using modem modulation.
3. The receiving fax machine has a T.30 protocol engine that communicates with the T.30 protocol engine in the fax server through the gateway.
4. With T.38 in a gateway, the sending fax machine sends a fax using modem modulation to transport T.30 protocol and image data to a gateway via the PSTN.
5. The gateway demodulates the incoming T.30 fax signals and image data and repackages them into T.38 packets.
6. The gateway then sends the T.38 packets to a T.38 endpoint, which then delivers the packets in T.30 protocol so the endpoint can receive the fax.

Gateway-to-gateway scenarios are also possible where two fax machines communicate using two gateways. In that case, the T.30 protocol engines in the two fax machines are transported across the packet network using T.38.

This diagram shows how the protocols work together during the call:



In all cases the application must establish the call first using the IP call control protocol. The call control protocol is responsible for the initial call set up and tear down.

Sending and Receiving Faxes

Your application can perform transport independent fax transmission and reception using the Bfv API fax functions. The application is unaware of the mode of transport (PSTN or SIP). Dialogic supports all fax functions for Fax over IP. See *Volume 4, Fax Processing, in the Dialogic® Brooktrout® Bfv API Reference Manual* for more information about Fax functions.

Configuring T.38, RTP and IP Call Control Activities

If your application runs on Windows systems, you can use the Brooktrout Configuration Tool to configure call control. Use the IP Call Control Module configuration window to modify values for:

- General information
- IP parameters for both SIP and H.323
- T.38 parameters
- RTP parameters

You can also use the following files to configure call control:

- The user-defined configuration file
The user-defined configuration file (*btcall.cfg*) contains configuration parameters for the Bfv API and driver.
- The call control configuration file
The call control configuration file (*callctrl.cfg*) contains configuration parameters that define how the user wants the Bfv API to configure the modules for call control.

See the *Dialogic® Brooktrout® Fax Products SDK Installation and Configuration Guide* that came with your software for information about using the Configuration Tool or Appendix A of the *Dialogic® Brooktrout® Bfv APIs Reference Manual, Volume 6* for file configuration to configure your system and applications to support fax over IP.

Troubleshooting

You can use any existing Bfv problem solving tools to troubleshoot your T.38 application. Use the log files created by these tools to understand what is happening and modify your application. See [Debugging on page 90](#) for more information about these tools.

Dialogic has provided a *debug_control* mode for T.38 problem solving (use under guidance of Dialogic Technical Services and Support). See *Volume 1* of the *Dialogic® Brooktrout® Bfv API Reference Manual* for more information.

Understanding the SIP Protocol

You can use the Session Initiation Protocol (SIP), an application-layer control (signaling) protocol, to create, modify, and terminate sessions with one or more participants.

These sessions include internet telephone calls, multimedia distribution, and multimedia conferences. SIP invitations used to create sessions carry session descriptions that allow participants to agree on a set of compatible media types. SIP makes use of elements called proxy servers to help do the following:

- Route requests to the user's current location
- Authenticate and authorize users for services
- Implement provider call-routing policies
- Provide features to users.

SIP also provides a registration function that allows users to upload their current locations for use by proxy servers.

SIP runs on top of several different transport protocols.

This section has the following sections:

- [Introduction to the SIP Protocol on page 241](#)
- [Overview of SIP Functionality on page 242](#)
- [SIP works with both IPv4 and IPv6. on page 243](#)

Introduction to the SIP Protocol

There are many applications of the internet that require the creation and management of a session (a session is considered to be an exchange of data between an association of participants). The implementation of these applications is complicated by the practices of participants: users might move between endpoints, they might be addressable by multiple names, and they might communicate in several different media - sometimes simultaneously. (For copyright information on this section, see [page 253](#).)

Numerous protocols carry various forms of real-time multimedia session data such as voice, video, or text messages. The Session Initiation Protocol (SIP) works in concert with these protocols by enabling internet endpoints (called user agents) to discover one another and to agree on a characterization of a session they would like to share.

For locating prospective session participants, and for other functions, SIP enables the creation of an infrastructure of network hosts (called proxy servers) to which user agents can send registrations, invitations to sessions, and other requests. SIP is an agile, general-purpose tool for creating, modifying, and terminating sessions that works independently of underlying transport protocols, regardless of the type of session being established.

Overview of SIP Functionality

SIP is an application-layer control protocol that can establish, modify, and terminate multimedia sessions (conferences) such as internet telephony calls. SIP can also invite participants to already existing sessions, such as multicast conferences. Media is added to (and removed from) an existing session. SIP transparently supports name mapping and redirection services, supporting personal mobility - users can maintain a single externally visible identifier regardless of their network location.

SIP supports five facets of establishing and terminating multimedia communications:

- User location: determination of the end system to be used for communication;
- User availability: determination of the willingness of the called party to engage in communications;
- User capabilities: determination of the media and media parameters to be used;
- Session setup: “ringing”, establishment of session parameters at both called and calling party;
- Session management: including transfer and termination of sessions, modifying session parameters, and invoking services.

SIP is not a vertically integrated communications system. SIP is rather a component that is used with other IETF protocols to build a complete multimedia architecture. Typically, these architectures include protocols such as the Real-time Transport Protocol (RTP) (RFC 1889 [28]) for transporting real-time data and providing QoS feedback, the Real-Time streaming protocol (RTSP) (RFC 2326 [29]) for controlling delivery of streaming media, the Media Gateway Control Protocol (MGACO) (RFC 3015 [30]) for controlling gateways to the Public Switched Telephone Network (PSTN), and the Session Description Protocol (SDP) (RFC 2327 [1]) for describing multimedia sessions. Therefore, SIP should be used in conjunction with other protocols in order to provide complete services to the users. However, the basic functionality and operation of SIP does not depend on any of these protocols.

SIP does not provide services. Rather, SIP provides primitives that are used to implement different services. For example, SIP can locate a user and deliver an opaque object to the current location. If this primitive is used to deliver a session description written in SDP, for instance, the endpoints can agree on the parameters of a session. If the same primitive is used to deliver a photo of the caller as well as the session description, a “caller ID” service is easily implemented. As this example shows, a single primitive is typically used to provide several different services.

SIP does not offer conference control services such as floor control or voting and does not prescribe how a conference is to be managed. SIP is used to initiate a session that uses some other conference control protocol. Since SIP messages and the sessions they establish can pass through entirely different networks, SIP cannot, and does not, provide any kind of network resource reservation capabilities.

The nature of the services provided make security particularly important. To that end, SIP provides a suite of security services, which include denial-of-service prevention, authentication (both user to user and proxy to user), integrity protection, and encryption and privacy services.

SIP works with both IPv4 and IPv6.

Overview of Operation

This section introduces the basic operations of SIP using simple examples. This section is tutorial in nature and does not contain any normative statements.

The first example shows the basic functions of SIP:

- Locating an end point
- Signaling a desire to communicate
- Negotiating session parameters to establish the session
- Tearing down the established session

Figure 20 on page 246 shows a typical example of a SIP message exchange between two users, Caller A and Caller B. (Each message is labeled with the letter “F” and a number for reference by the text.) In this example, Caller A uses a SIP application on a PC (referred to as a softphone) to call Caller B on the SIP phone over the internet. Also shown are two SIP proxy servers that act on behalf of Caller A and Caller B to facilitate the session establishment. This typical arrangement is often referred to as the “SIP trapezoid” as shown by the geometric shape of the dotted lines in *Figure 20 on page 246*.

Caller A “calls” Caller B using Caller B’s SIP identity, a type of Uniform Resource Identifier (URI) called a SIP URI. It has a similar form to an email address, typically containing a user name and a host name. In this case, it is sip:callerb@biloxi.com, where biloxi.com is the domain of Caller B’s SIP service provider. Caller A has a SIP URI of sip:callera@atlanta.com. Caller A might have typed in Caller B’s URI or perhaps clicked on a hyperlink or an entry in an address book. SIP also provides a secure URI, called a SIPS URI. An example would be sips:callerb@biloxi.com. A call made to a SIPS URI guarantees that secure, encrypted transport (namely TLS) is used to carry all SIP messages from the caller to the domain of the callee. From there, the request is sent securely to the callee, but with security mechanisms that depend on the policy of the domain of the callee.

SIP is based on an HTTP-like request/response transaction model. Each transaction consists of a request that invokes a particular method, or function, on the server and at least one response. In this example, the transaction begins with Caller A’s softphone sending an INVITE request addressed to Caller B’s SIP URI. INVITE is an example of a SIP method that specifies the action that the requester (Caller A) wants the server (Caller B) to take. The INVITE request

contains a number of header fields. Header fields are named attributes that provide additional information about a message. The ones present in an INVITE include the following:

- Unique identifier for the call
- Destination address
- Caller A's address
- Information about the type of session that Caller A wishes to establish with Caller B

The INVITE (message F1 in *Figure 20*) might look like this:

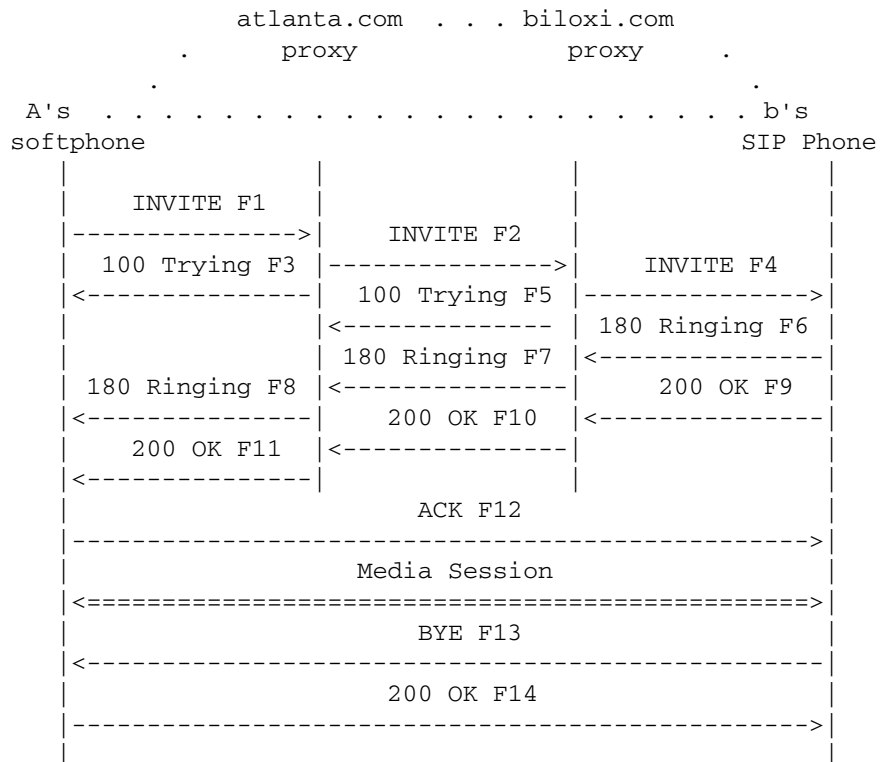


Figure 20. SIP Session Setup Example With Sip Trapezoid

```

INVITE sip:b@biloxi.com SIP/2.0
Via: SIP/2.0/UDP pc33.atlanta.com;branch=z9hG4bK776asdhds
Max-Forwards: 70
To: b <sip:b@biloxi.com>
From: A <sip:a@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:a@pc33.atlanta.com>
Content-Type: application/sdp
Content-Length: 142

```

(A's SDP not shown)

INVITE Message

Note: Caller A's SDP not shown

	The first line of the text-encoded message contains the method name (INVITE). The lines that follow are a list of header fields. This example contains a minimum required set. The header fields are briefly described below:
Via	Contains the address (pc33.atlanta.com) at which Caller A is expecting to receive responses to this request. It also contains a branch parameter that identifies this transaction.
To	Contains a display name (Caller B) and a SIP or SIPS URI (sip:callerb@biloxi.com) towards which the request was originally directed. Display names are described in RFC 2822 [3].
From	Also contains a display name (Caller A) and a SIP or SIPS URI (sip:callera@atlanta.com) that indicate the originator of the request. This header field also has a tag parameter containing a random string (1928301774) that was added to the URI by the softphone. It is used for identification purposes.
Call-ID	Contains a globally unique identifier for this call, generated by the combination of a random string and the softphone's host name or IP address. The combination of the To tag, From tag, and Call-ID completely defines a peer-to-peer SIP relationship between Caller A and Caller B and is referred to as a dialog.
CSeq	(or Command Sequence) contains an integer and a method name. The CSeq number is incremented for each new request within a dialog and is a traditional sequence number.
Contact	Contains a SIP or SIPS URI that represents a direct route to contact Caller A, usually composed of a username at a fully qualified domain name (FQDN). While an FQDN is preferred, many end systems do not have registered domain names, so IP addresses are permitted. While the Via header field tells other elements where to send the response, the Contact header field tells other elements where to send future requests.
Max-Forwards	Serves to limit the number of hops a request can make on the way to its destination. It consists of an integer that is decremented by one at each hop.
Content-Type	Contains a description of the message body (not shown).

Content-Length

Contains an octet (byte) count of the message body.

The details of the session, such as the type of media, codec, or sampling rate, are not described using SIP. Rather, the body of a SIP message contains a description of the session, encoded in some other protocol format. One such format is the Session Description Protocol (SDP) (RFC 2327 [1]). This SDP message (not shown in the example) is carried by the SIP message in a way that is analogous to a document attachment being carried by an email message, or a web page being carried in an HTTP message.

Since the softphone does not know the location of Caller B or the SIP server in the biloxi.com domain, the softphone sends the INVITE to the SIP server that serves Caller A's domain, atlanta.com. The address of the atlanta.com SIP server could have been configured in Caller A's softphone, or it could have been discovered by the Dynamic Host Configuration Protocol (DHCP), for example.

The atlanta.com SIP server is a type of SIP server known as a proxy server. A proxy server receives SIP requests and forwards them on behalf of the requestor. In this example, the proxy server receives the INVITE request and sends a 100 (Trying) response back to Caller A's softphone. The 100 (Trying) response indicates that the INVITE has been received and that the proxy is working on Caller A's behalf to route the INVITE to the destination. Responses in SIP use a three-digit code followed by a descriptive phrase. This response contains the same To, From, Call-ID, CSeq and branch parameter in the Via as the INVITE, which allows Caller A's softphone to correlate this response to the sent INVITE. The atlanta.com proxy server locates the proxy server at biloxi.com, possibly by performing a particular type of DNS (Domain Name Service) lookup to find the SIP server that serves the biloxi.com domain. This is described in [4]. As a result, it obtains the IP address of the biloxi.com proxy server and forwards, or proxies, the INVITE request there. Before forwarding the request, the atlanta.com proxy server adds an additional Via header field value that contains its own address (the INVITE already contains Caller A's address in the first Via). The biloxi.com proxy server receives the INVITE and responds with a 100 (Trying) response back to the atlanta.com proxy server to indicate that it has received the INVITE and is processing the request. The proxy server consults a database, generically called a location service, that contains the current IP address of Caller B. The biloxi.com proxy server adds another Via header field value with its own address to the INVITE and proxies it to Caller B's SIP phone.

Caller B's SIP phone receives the INVITE and alerts Caller B to the incoming call from Caller A so that Caller B can decide whether to answer the call, that is, Caller B's phone rings. Caller B's SIP phone indicates this in a 180 (Ringing) response, which is routed back through the two proxies in the reverse direction. Each proxy uses the Via header field to determine where to send the response and removes its own address from the top. As a result, although DNS and location service lookups were required to route the initial INVITE, the 180 (Ringing) response is returned to the caller without lookups or without state being maintained in the proxies. This also has the desirable property that each proxy that sees the INVITE also sees all responses to the INVITE.

When Caller A's softphone receives the 180 (Ringing) response, it passes this information to Caller A, perhaps using an audio ringback tone or by displaying a message on Caller A's screen.

In this example, Caller B decides to answer the call. When he picks up the handset, the SIP phone sends a 200 (OK) response to indicate that the call has been answered. The 200 (OK) contains a message body with the SDP media description of the type of session that Caller B is willing to establish with Caller A. As a result, there is a two-phase exchange of SDP messages: Caller A sent one to Caller B, and Caller B sent one back to Caller A. This two-phase exchange provides basic negotiation capabilities and is based on a simple offer/answer model of SDP exchange. If Caller B did not wish to answer the call or was busy on another call, an error response would have been sent instead of the 200 (OK), which would have resulted in no media session being established. The 200 (OK) (message F9 in [Figure 20 on page 246](#)) might look like this as Caller B sends it out:

```
SIP/2.0 200 OK
  Via: SIP/2.0/UDP server10.biloxi.com; branch=z9hG4bKnashds8;
    received=192.0.2.3
Via: SIP/2.0/UDP bigbox3.site3.atlanta.com; branch=z9hG4bK77ef4c2312983.1;
  received=192.0.2.2
Via: SIP/2.0/UDP pc33.atlanta.com; branch=z9hG4bK776asdhs;
  received=192.0.2.1
To: b <sip:b@biloxi.com>;tag=a6c85cf
From: A <sip:a@atlanta.com>;tag=1928301774
Call-ID: a84b4c76e66710@pc33.atlanta.com
CSeq: 314159 INVITE
Contact: <sip:b@192.0.2.4>
Content-Type: application/sdp
Content-Length: 131
```

(Caller B's SDP not shown)

The first line of the response contains the response code (200) and the reason phrase (OK). The remaining lines contain header fields. The Via, To, From, Call-ID, and CSeq header fields are copied from the INVITE request. (There are three Via header field values - one added by Caller A's SIP phone, one added by the atlanta.com proxy, and one added by the biloxi.com proxy.) Caller B's SIP phone has added a tag parameter to the To header field. This tag is incorporated by both endpoints into the dialog and is included in all future requests and responses in this call. The Contact header field contains a URI at which Caller B can be directly reached at a SIP phone. The Content-Type and Content-Length refer to the message body (not shown) that contains Caller B's SDP media information.

In addition to DNS and location service lookups shown in this example, proxy servers can make flexible “routing decisions” to decide where to send a request. For example, if Caller B's SIP phone returned a 486 (Busy Here) response, the biloxi.com proxy server could proxy the INVITE to Caller B's voicemail server. A proxy server can also send an INVITE to a number of locations at the same time. This type of parallel search is known as forking (see below).

In this case, the 200 (OK) is routed back through the two proxies and is received by Caller A's softphone, which then stops the ringback tone and indicates that the call has been answered. Finally, Caller A's softphone sends an acknowledgement message, ACK, to Caller B's SIP phone to confirm the reception of the final response (200 (OK)). In this example, the ACK is sent directly from Caller A's softphone to Caller B's SIP phone, bypassing the two proxies. This occurs because the endpoints have learned each other's address from the Contact header fields through the INVITE/200 (OK) exchange, which was not known when the initial INVITE was sent. The lookups performed by the two proxies are no longer needed, so the proxies drop out of the call flow. This completes the INVITE/200/ACK three-way handshake used to establish SIP sessions.

Caller A and Caller B's media session has now begun, and they send media packets using the format to which they agreed in the exchange of SDP. In general, the end-to-end media packets take a different path from the SIP signaling messages.

During the session, either Caller A or Caller B may decide to change the characteristics of the media session. This is accomplished by sending a re-INVITE containing a new media description. This re-INVITE references the existing dialog so that the other party knows that it is to modify an existing session instead of establishing a new session. The other party sends a 200 (OK) to accept the change. The requestor responds to the 200 (OK) with an ACK. If the other party does not accept the change, he sends an error response such as 488 (Not Acceptable Here), which also receives an ACK. However, the failure of the re-INVITE does not cause the existing call to fail - the session continues using the previously negotiated characteristics.

At the end of the call, Caller B disconnects (hangs up) first and generates a BYE message. This BYE is routed directly to Caller A's softphone, again bypassing the proxies. Caller A confirms receipt of the BYE with a 200 (OK) response, which terminates the session and the BYE transaction. No ACK is sent - an ACK is only sent in response to an INVITE request. The reasons for this special handling for INVITE relate to the reliability mechanisms in SIP, the length of time it can take for a ringing phone to be answered, and forking. For this reason, request handling in SIP is often classified as either INVITE or non- INVITE, referring to all other methods besides INVITE.

In some cases, it may be useful for proxies in the SIP signaling path to see all the messaging between the endpoints for the duration of the session. For example, if the biloxi.com proxy server wished to remain in the SIP messaging path beyond the initial INVITE, it would add to the INVITE a required routing header field known as Record- Route that contained a URI resolving to the hostname or IP address of the proxy. This information would be received by both Caller B's SIP phone and (due to the Record-Route header field being passed back in the 200 (OK)) Caller A's softphone and stored for the duration of the dialog. The biloxi.com proxy server would then receive and proxy the ACK, BYE, and 200 (OK) to the BYE. Each proxy can independently decide to receive subsequent messages, and those messages pass through all proxies that elect to receive it. This capability is frequently used for proxies that are providing mid-call features.

Registration is another common operation in SIP. Registration is one way that the biloxi.com server can learn the current location of Caller B. Upon initialization, and at periodic intervals, Caller B's SIP phone sends REGISTER messages to a server in the biloxi.com domain known as a SIP registrar. The REGISTER messages associate Caller B's SIP or SIPS URI (sip:callerb@biloxi.com) with the machine into which he is currently logged (conveyed as a SIP or SIPS URI in the Contact header field). The registrar writes this association, also called a binding, to a database, called the location service, where it is used by the proxy in the biloxi.com domain. Often, a registrar server for a domain is co-located with the proxy for that domain. It is an important concept that the distinction between types of SIP servers is logical, not physical.

Caller B is not limited to registering from a single device. For example, both the SIP phone at home and the one in the office could send registrations. This information is stored together in the location service and allows a proxy to perform various types of searches to locate Caller B. Similarly, more than one user is registered on a single device at the same time.

The location service is just an abstract concept. It generally contains information that allows a proxy to input a URI and receive a set of zero or more URIs that tell the proxy where to send the request. Registrations are one way to create this information, but not the only way. Arbitrary mapping functions are configured at the discretion of the administrator.

Finally, it is important to note that in SIP, registration is used for routing incoming SIP requests and has no role in authorizing outgoing requests. Authorization and authentication are handled in SIP either on a request-by-request basis with a challenge/response mechanism, or by using a lower layer scheme.

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Using Third Party IP Stacks

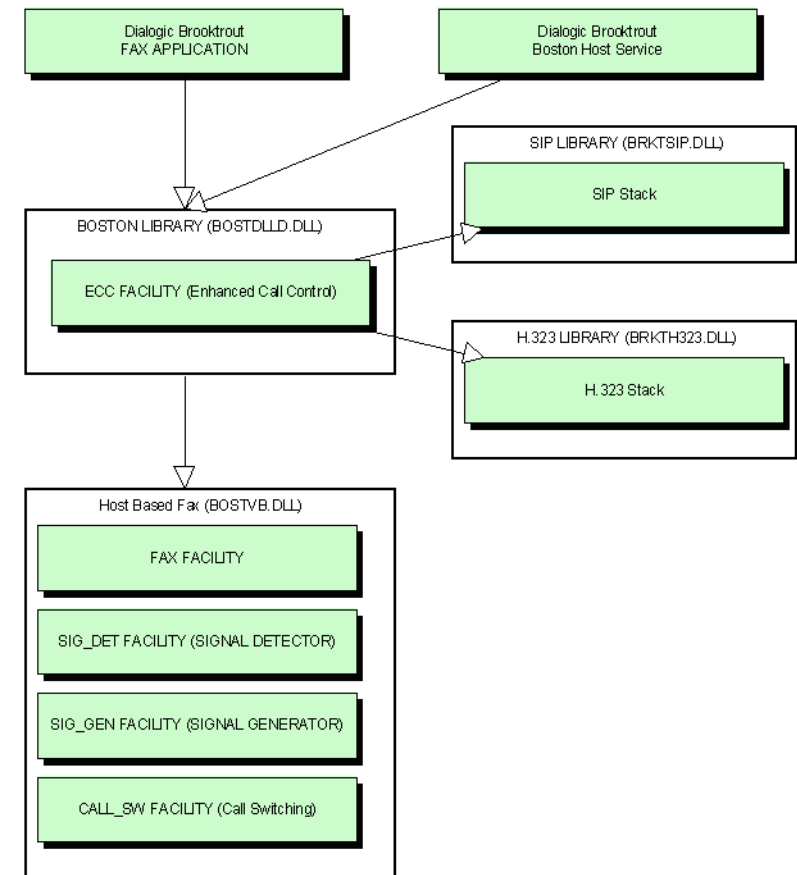
The Brooktrout Bfv Fax can be integrated with systems that have their own IP call control stacks. The primary IP stacks are SIP and H.323. These stacks negotiate an RTP and a T.38 port to perform fax. With the use of the Brooktrout SR140 software, a pure fax software solution can be integrated on systems that have VoIP features.

Systems that have available board slots can take advantage of the TR1034-N board to unload CPU cycles from the system to the board for fax processing. With this integration, it will be up to the third party IP stack to negotiate RTP and T.38 ports.

Integrating Bfv IP Fax

Figure 21 shows the individual components that an application interacts with via the Bfv API. Although other facilities exist, only the fax-related facilities are shown below.

Figure 21. Components Interacting with Application



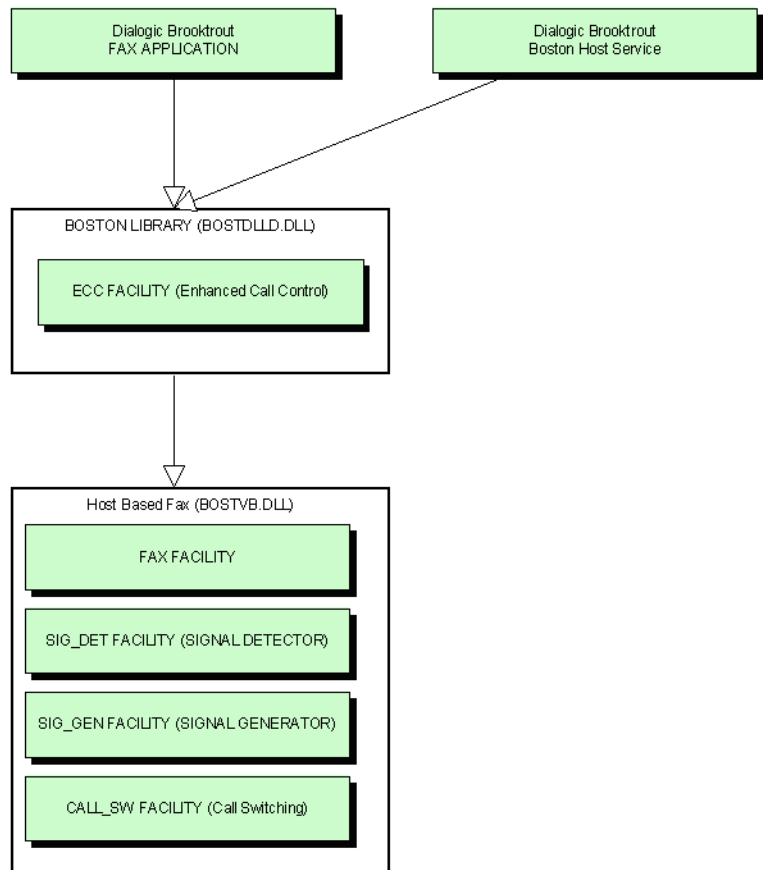
Components

The following figure shows the components included in this configuration. Note that you must disable the functionality of the following:

- ECC Component
- H.323 Stack
- SIP Stack

Refer to [Disable ECC Component on page 257](#) which disables the stacks too.

Figure 22. Components in Configuration



Configuration

You can configure the Bfv with the Brooktrout Configuration Tool on Windows or by creating the *callctrl.cfg* files manually.

The Brooktrout Configuration Tool is a full-solution configuration because the call control is included in the configuration along with the IP Stack which has been integrated with BFV.

It can be used to create the initial *callctrl.cfg* and *btcall.cfg*. However, for this setup, you must edit the *callctrl.cfg* manually.

Dialogic does not recommend that you use Brooktrout Configuration Tool beyond the initial creation of the configuration files. Refer to the Call Control Configuration File section in the *Dialogic® Brooktrout® Bfv APIs Reference Manual* for more information on the parameters available in *callctrl.cfg*.

Disable ECC Component

To disable the ECC component and allow a third party to manage call control, remove the following sections in *callctrl.cfg*:

```
[module.X/host_cc.1]
[host_module.1]
[host_module.1/t38parameters]
[host_module.1/parameters]
```

The following *callctrl.cfg* examples are for Linux with paths to the default installation directory.

Brooktrout SR140 Software-Based Integration - Linux

The following *callctrl.cfg* configuration is an example for a Brooktrout SR140 “pure” software based integration for Linux in the default installation directory.

```
# callctrl.cfg
#
# Sample Call Control configuration file for Boston Bfv
# API with SR140.
#
[module.41]
    vb_firm=/usr/sys/Brooktrout/Boston/fw/bostvb.so
# This parameter should be set to the number of channels
# licensed for the SR140 product
    channels=30
[module.41/ethernet.1]
    ip_interface=eth0
```

TR1034 Board-Based Integration - Linux

The following *callctrl.cfg* configuration is an example for a TR1034 board based integration for Linux in the default installation directory.

```
# callctrl.cfg
#
# Sample Call Control configuration file for Boston Bfv
# API with TR1034.
#
[module.2]
    cc_type=1
    channels=30
    set_api=bfv
    pcm_law=mulaw
[module.2/clock_config]
    clock_mode=master
    clock_source=internal
    clock_compatibility=none
    bus_speed=2
    master_ref_fallback=disabled
    master_drive=clock_a
[module.2/ethernet.1]
```

```
# Specify a static IP address such as 192.168.0.2
ip_address=0.0.0.0
# Specify an IP mask such as 255.255.255.0
ip_netmask=0.0.0.0
# Specify an IP address such as 192.168.0.1
ip_gateway=0.0.0.0
ip_broadcast=0.0.0.0
ethernet_speed=auto
ip_arp_timeout=10
```

Call Negotiation

Inbound Call

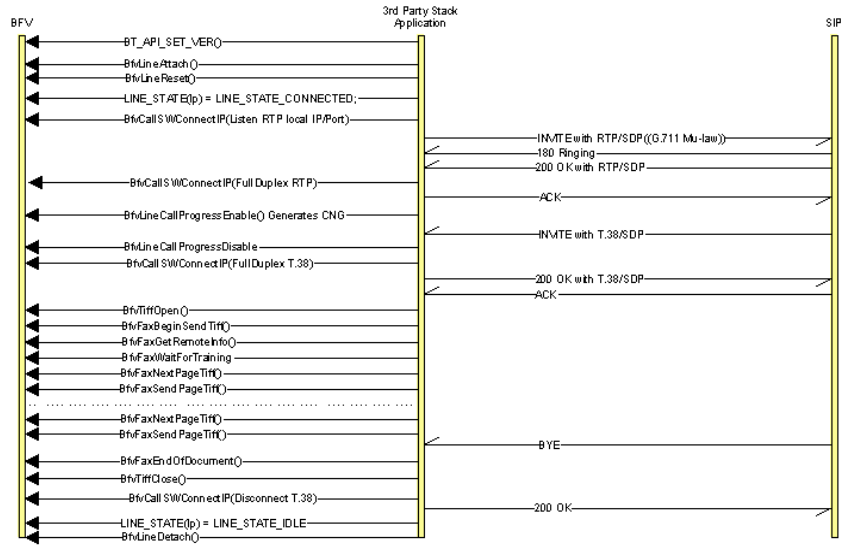
The following section uses SIP to demonstrate the interaction between the third party stack in the application and the Bfv API for negotiating RTP and T.38 for an inbound call.

Applications that perform their own RTP can generate the CED/ANSam tone and avoid setting the RTP stream on the Bfv API.

If this is the case, applications can skip calling `BfvCallSWConnectIP(FullDuplex RTP)` and `BfvTonePlayBeep()` CED.

If both RTP and T.38 are done with the Bfv API, some ICMP errors may appear as the port is switched from RTP to T.38. To avoid the ICMP errors use unique ports for RTP and T.38.

Figure 24. Outbound SIP Sequence



SR140 Internet Aware Fax (IAF) Support over UDP

The T.38 IAF mode allows fax endpoints that are directly interconnected by an IP network, to send the fax at much higher speeds than normal, since they are not constrained by modem or G.711 limitations. This mode is defined in the ITU-T T.38 Recommendation defining a real-time method for sending faxes directly between two fax-over-IP (FoIP) endpoints over an all-IP network. Real-time and standards based; continues to support existing business compliance.

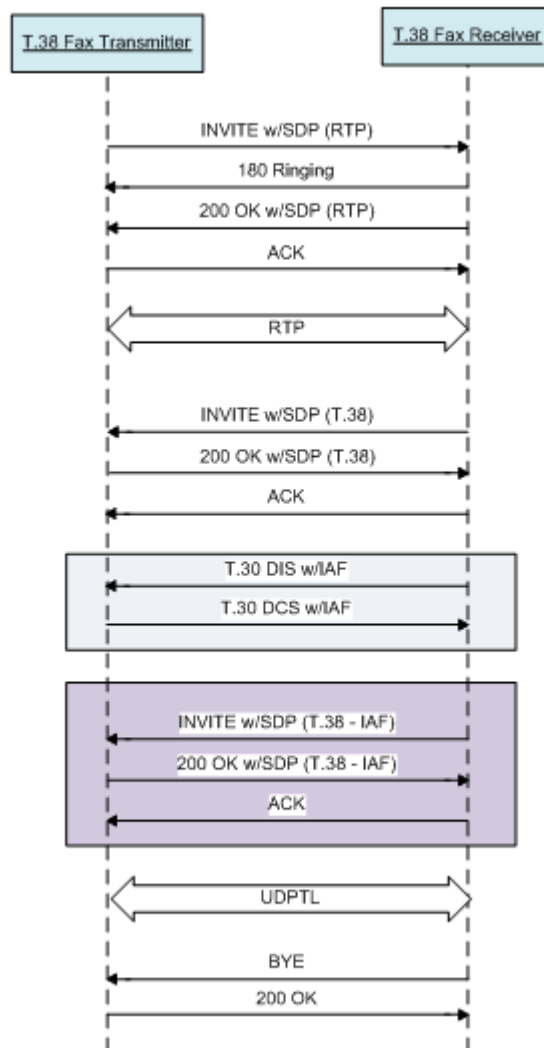
This feature takes advantage of direct IP-IP interconnection, such as Enterprise IP Networks or SIP Trunks with Peering. This will allow the SR140 to support T.38 speeds of up to 150kbps between two IAF devices.

This feature requires an optional, additional Dialogic® Brooktrout® SR140 Fax Software license for speeds up to 150 kbps. The IAF feature license shall belong to the full SR140 family and requires a fax channel license for use. This will not co-exist with SR140-L or SR140-LL. IAF is supported for 60 simultaneous channels at image speeds of 150kbps.

SR140 IAF requires SIP call control, T.38 version 1 or higher and ECM to be enabled.

The following is an example of the SIP call flow between two IAF enabled endpoints that negotiate into IAF mode:

Figure 25. SIP Call Flow



A device will advertise that it supports an IAF transaction by setting bit 123 in its T.30 DIS capabilities message. When both T.38 endpoints support IAF mode, and all the feature requirements have been met, a T.38 re-invite will be sent requesting to perform a T.38 transaction as an IAF. When this request is accepted, the fax data rate will be transmitted at the new IAF negotiated rate and the transmitter will respond with its T.30 DCS capabilities indicating IAF. If an endpoint does not support IAF or does not respond to an IAF re-invite, the fax will continue as a T.38 or G.711 fax.

To support the configuration of IAF, new User-Defined Configuration (*btcall.cfg*) parameters have been made available:

'iaf_enable', decimal units, to enable or disable IAF support.
Default 0, disabled.

'iaf_max_bit_rate', unsigned int, bps units, Default 0 means max supported by license.

The following Bfv API structure changes will be made to support T.38 IAF functionality. Please see the *Dialogic® Brooktrout® Bfv API Reference Manual* for more information.

A new field has been added to the *args_fax_t30_params* structure defined in *mill_api.h* to allow Bfv applications to specify the IAF maximum bitrate on a per-call basis using *BfvFaxT30Params()*. The new field is named *iaf_max_bitrate* and is defined as follows:

```
struct args_fax_t30_params {
    RES res;
    unsigned bit_rate;
    unsigned scan_time;
    unsigned modulation_type;
    unsigned line_compression;
    MILL_LINE *async_lp;
    unsigned iaf_bit_rate;
    char xmt_level;
    char rcv_level;
};
```

When passing the *args_fax_t30_params* structure to the *BfvFaxT30Params()* function in order to specify an IAF maximum bitrate, the *modulation_type* field of the *args_fax_t30_params* structure should be set to *MODULATION_IAF*.

Two new fields have been added to the *PAGE_RES* structure to return information to the Bfv application notifying whether the fax transmission occurred in IAF mode and, if so, what the data rate was for the page image. The new fields are named *iaf_initial_speed* and *iaf_final_speed* and are defined as follows:

```
typedef struct page_res {
    int page_complete_type;
    int continue_breaks;
    unsigned ascii_bytes;
    ...
    unsigned char iaf_initial_speed;
    unsigned long iaf_final_speed;
} PAGE_RES;
```

7 - Robbed Bit Signaling

This chapter describes robbed bit signaling as used with BSMI-level call control.

The chapter has the following sections:

- [*General Information on page 266*](#)
- [*Wink Start & Delay Dial Signaling on page 277*](#)
- [*Wink Start with Feature Group B & D on page 282*](#)
- [*Immediate Start/Fixed Pause Signaling on page 286*](#)
- [*Ground Start Signaling on page 290*](#)
- [*Loop Start Signaling on page 302*](#)
- [*FXO Loop Start on page 303*](#)

General Information

This chapter covers protocols Dialogic supports for interfacing with the Local Exchange Carrier (LEC) Network (that is, the North American Telephone Network).

Note: This chapter applies only to BSMI (low-level call control) users. Bfv API call control users do not need to be concerned with this information.

These protocols apply either to digital (T1 or E1) or analog lines. The different types of lines simply provide a different mechanism for conveying the signal-electric signals (loop current and ring voltage) on analog lines and bits on digital lines.

On digital lines, these protocols are sometimes referred to as Channel Associated Signaling (CAS) or Robbed-Bit Signaling (RBS) due to the nature of the signaling. Each channel's state is represented as a set of 2 signaling bits, and these bits are transmitted on the line at constant intervals. The LEC protocols are normally associated with T1 lines, where they were originally used, but nowadays it is possible to find these protocols used in PBX's using E1 lines (which provide 4 signaling bits per channel).

A different technique is to reserve a full channel on a T1 or E1 line to carry information about all channels, and to use the available bits as a continuous stream carrying information “packets” instead of repeating a number of signaling bits over time. This technique is called Common-Channel Signaling (CCS), and is used on ISDN lines among others.

In T1 lines, audio and line state information is grouped in frames, each frame consists of 8 bits of data for each of the 24 channels, plus a framing bit, adding to 193 bits/frame (8 bits/byte * 1 byte/channel * 24 channels/frame + 1bit). The sampling rate is 8000 Hz, so the bit rate is $8000 \times 193 = 1,544,000$, or 1.544 Mb/s.

The technique used for carrying the signaling bits on T1 is to use (“rob,” hence the term Robbed-Bit Signaling) some of the bits normally intended to represent data (voice/fax/data) on a channel for the purposes of call control. Research has shown that robbing the least significant bit of each channel's sample every 6th frame causes a virtually imperceptible (for humans) level of distortion for voice. However, when raw data is being sent, this loss becomes unacceptable, so for simplicity only 7 of the 8 bits are used for data applications. This technique allows for all 24 channels to be used for calls, as opposed to 23 channels used for calls and one for call control (23B+D), as with ISDN.

In E1 lines (32 timeslots per line), one of the timeslots (0) in each frame is reserved for framing and synchronization data, and another (timeslot 16) is used for the signaling bits — in the first frame it carries information about audio timeslots 1 and 17, in the second frame for timeslots 2 and 18 and so on until all audio channels are covered, and then the process starts over.

The protocols describe the value and meaning of the signaling bits and the timing between changes in their value.

Modules support the following LEC protocols:

- E&M Wink Start
- E&M Immediate Start
- E&M Delay Dial
- FXO Loop Start
- FXS Loop Start
- FXO Ground Start
- FXS Ground Start

All signaling modes support pulsed (10 pulses per second), DTMF and MF dialing. However, detection of digits must be handled by the host.

ITU's R2 protocol, normally used in E1 lines is also supported, and it is described in [Using the BSMI R2 Signaling Capability](#) on [page 335](#).

Wink Start, Immediate Start and Delay Dial protocols are typically used to connect equivalent devices (for example, Central Offices). The same protocol must be loaded on both channel involved in a call. Loop Start and Ground Start protocols connect different types of devices (a Central Office or PBX to a Station). In this case the side acting as Central Office must load the FXO (foreign exchange office) version of the protocol, while the Station side must load the FXS (foreign exchange station) version.

The messages and data structures used for the host application and the module to communicate form the BSMI interface, described in *Volume 5 of the Dialogic® Brooktrout® Bfv API Reference Manual*. The same Bfv API is used also for ISDN and R2 — the same messages are used to control/notify similar events across all protocols.

Structure IISDN_BCHANNEL_ID contains fields common to all CAS protocols plus data structures containing LEC- and R2-specific configuration (structures IISDN_ROBBED_BIT_DATA and IISDN_E1_CAS_R2_DATA respectively). Some of the message fields have names that reflect their ISDN roots, but their definition is extended for use with all CAS protocols (LEC protocols as well as R2). The most important ones are *lapdid*, which in CAS protocols identifies the trunk number in the module (0-based) and *call_ref*, a 16-bit value in which the most-significant byte must always contain the trunk number (same value as *lapdid*) and the least significant byte

the B-channel number on that trunk (also 0-based). Some of the data structures associated with the message will have a specific field for specifying the B-channel, which must also be set.

Enabling a channel and selecting the protocol to run is done through message L4L3mENABLE_CAS, while message L4L3mDISABLE_CAS is used to disable a channel. It is possible to have different channels on the same T1 trunk running different protocols. Once a DS0 channel is configured to run a LEC protocol most processing specific to that protocol, is handled by the module. Channels are numbered 0-(N-1) for all line types (T1, E1 and analog) and protocols.

Dialogic supports the most common protocols. Contact Dialogic to make requests for ones not known to be officially supported. See [Getting Technical Support](#) on [page 19](#) for contact information.

Timer Definitions

The LEC structure (named IISDN_ROBBED_BIT_DATA for historical reasons) contains a series of 16-bit values used to configure the various timing parameters associated with the protocols. [Table 13](#) lists the timers in alphabetical order and contains each timer's unit, meaning, and default value. The unit indicates by how much a timer value must be multiplied to find its actual duration in milliseconds, while the default (the number in parentheses following its define value) indicates the number of ticks.

Proper timing with LEC protocols is very important. There are over 30 unique timings that are set, with a significant subset of these applicable to several protocols. You should match their entries in the table to the appropriate timing diagrams.

Table 13. Robbed Bit Signaling Timers

Timer	Granularity	Definition	Default Value
answer_timer	50 ms	Timed Answer Supervision. Timer started after the last digit has been dialed. Used to report distant end answer when the protocol doesn't provide a specific signal to indicate the called party has answered the call.	IISDN56_ANSW_TIMER_DEFAULT (60) 3 seconds

Table 13. Robbed Bit Signaling Timers (Continued)

Timer	Granularity	Definition	Default Value
critical_timer	50 ms	Critical Interdigit Timer. While racking incoming distant digits, the maximum time to wait between digits before giving up and returning an incomplete rack.	IISDN56_CRIT_TIMER_DEFAULT (20) 1 second
delayed_on_hook_timer	50 ms	Delayed on hook timer. If set to a value greater than 0, the amount of time it waits for the host to set the line onhook (message L4L3mCLEAR_REQUEST) after the protocol has detected termination of the call. If this timer expires, the protocol sets the line onhook automatically. If this value is set to 0 or infinity (0xFFFF), the protocol waits forever for the host to set the line onhook. The timer is only used if field delayed_onhook_mode is set to TRUE.	IISDN56_DELAYED_ON_HOOKTIMER_DEFAULT (20) 1 second
delayed_wink_timer	20 ms	Delayed Wink timer. If set to a value greater than 0, the amount of time it waits for the host to send the wink ((message L4L3mTX_WINK) after the protocol has detected an incoming call. If this timer expires, the protocol sends the wink automatically. If this value is set to 0 or infinity (0xFFFF), the protocol waits forever for the host to send the wink. The timer is only used if field delayed_wink_mode is set to TRUE.	IISDN_DELAYED_WINK_TIMER_DEFAULT (20) 1 second
dptimer_break	10 ms	Dial Pulse Break. On-hook (break) time for outdialing a dial pulse digit.	IISDN56_DPBREAK_TIMER_DEFAULT (4) 40 ms
dptimer_end_dial_pause	10 ms	Dial Pulse Pause After Dialing. Time after outpulsing the last digit before Instant ISDN sends any supervision signals or receives answer supervision.	IISDN56_DPEND_DIAL_TIMER_DEFAULT (5) 50 ms

Table 13. Robbed Bit Signaling Timers (Continued)

Timer	Granularity	Definition	Default Value
dptimer_immed_delay	10 ms	Dial Pulse Immediate Delay. Time after an incoming seizure before Instant ISDN will accept incoming dial pulse digits.	IISDN56_IMMED_DELAY_TIMER_DEFAULT (10) 100 ms
dptimer_interdigit	10 ms	Dial Pulse Interdigit. Time between digits when performing digit outpulsing.	IISDN56_DPINTERDGT_TIMER_DEFAULT (30) 300 ms
dptimer_make	10 ms	Dial Pulse Make. Off-hook (make) time for outdialing a dial pulse digit.	IISDN56_DPMAKE_TIMER_DEFAULT (6) 60 ms
dptimer_post_wink	10 ms	Amount of time the protocol will keep the line in idle state after transmission of a wink signal before reporting to the host the wink has been sent. This ensures a minimum period for the idle signal to be present on the line, even if the host sends the line offhook immediately after being notified of an incoming call or end of a transmitted wink.	IISDN56_DPPOSTWINK_TIMER_DEFAULT (5) 50 ms
dptimer_pre_wink	10 ms	Dial Pulse Prewink. Time Instant ISDN pauses before sending a wink signal.	IISDN56_DPPREWINK_TIMER_DEFAULT (5) 50 ms
dptimer_wink	10 ms	Duration of a transmitted wink.	IISDN56_DPWINK_TIMER_DEFAULT (20) 200 ms
first_indigit_timer	50 ms	Normal First Interdigit Timer. While racking incoming digits, the maximum time to wait for the first digit before giving up and returning an incomplete rack (valid during normal interdigit timing only).	IISDN56_FIRST_DGT_TIMER_DEFAULT (60) 3 seconds
fixed_pause_timer	50 ms	Fixed Pause Wait Timer. Time to wait before allowing outgoing dialing to commence (applies when outgoing start dial type is fixed pause only).	IISDN56_FIXP_TIMER_DEFAULT (2) 100 ms

Table 13. Robbed Bit Signaling Timers (Continued)

Timer	Granularity	Definition	Default Value
glare_ detection_ timer	10 ms	Glare Detection. Timer started at the beginning of outseizure that determines if a wink start or delay dial was returned too quickly to be an acknowledgment of the outseizure, but should be treated as glare.	IISDN56_GLARE_TIMER_DEFAULT (5) 50 ms
guard_ interval_timer	50 ms	Guard Interval Timer. Time after the circuit becomes idle before it is used for another outgoing call.	IISDN56_GUARD_TIMER_DEFAULT (20) 1 second
hookflash_ timer	10 ms	Hookflash Timer. Duration of a transmitted hook flash signal.	IISDN56_HOOKFLASH_TIMER_DEFAULT(80) 800 ms
hooktimer_ onhook_rls	10 ms	Normal Outgoing Release Filter. Filter on receive signal bit detector or the amount of time it must be idle before distant end releasing is assumed. This filter is in effect some time after answer (specified by modified outgoing release time) until completion of call.	IISDN56_ONHK_TIMER_DEFAULT (50) 500 ms
hooktimer_ onhk_mod_in_ rls	10 ms	Modified Incoming Release Filter. Filter on receive signal bit detector or the amount of time it must be idle before distant end releasing is assumed. This filter is in effect from inseizure until sometime after answer (specified by modified incoming release time).	IISDN56_MOD_ONHK_IN_TIMER_DEFAULT (80) 800 ms
hooktimer_ onhk_mod_ out_rls	10 ms	Modified Outgoing Release Filter. Filter on receive signal bit detector or the amount of time it must be idle before distant end releasing is assumed. This filter is in effect from answer until some time after answer (specified by modified outgoing release time).	IISDN56_MOD_ONHK_OUT_TIMER_DEFAULT (80) 800 ms

Table 13. Robbed Bit Signaling Timers (Continued)

Timer	Granularity	Definition	Default Value
hooktimer_ offhook_inseize	10 ms	Inseizure Filter. Time that the receive signal bit must be in-use to be considered an inseizure from the connected equipment.	IISDN56_OFHK_INSZ_TIMER_DEFAULT (5) 50 ms
hooktimer_ offhook_ answer	10 ms	Answer Filter. Filter used on the receive signal bit during the interval during which Instant ISDN is looking for an answer from the connected equipment.	IISDN56_OFHK_ANS_TIMER_DEFAULT (10) 100 ms
hooktimer_ min_wink	10 ms	Wink Filter. Minimum off-hook that will be interpreted as a valid start dial signal when outgoing start dial is selected as wink start or delay dial.	IISDN56_MIN_WINK_TIMER_DEFAULT (10) 100 ms
hooktimer_ maxmake	10 ms	Longest make duration that is counted as part of a dial pulse. A longer make signal will cause digit racking to be terminated.	IISDN56_MAXMAKE_TIMER_DEFAULT (8) 8 ms
hooktimer_ maxbreak	10 ms	Longest break duration that is counted as part of a dial pulse. A longer break signal will be considered the end of a pulse digit. If the duration eventually exceeds the value specified in field critical timer, it is interpreted as a disconnection signal.	IISDN56_MAXBREAK_TIMER_DEFAULT (8) 8 ms
hooktimer_ minmake	10 ms	Minimum Dial Pulse Make Time. Shortest make region that is counted as a dial pulse make region. Anything shorter should be ignored.	IISDN56_MINMAKE_TIMER_DEFAULT (1) 1 ms

Table 13. Robbed Bit Signaling Timers (Continued)

Timer	Granularity	Definition	Default Value
hooktimer_minbreak	10 ms	Minimum Dial Pulse Break Time. Shortest break region that is counted as a dial pulse break region. Anything shorter should be ignored.	IISDN56_MINBREAK_TIMER_DEFAULT (1) 1 ms
ign_insz_post_rls_timer	10 ms	Ignore Inseize After Release Timer. Time after the interface has idled before Instant ISDN will allow an inseizure from the connected equipment.	IISDN56_IGNINSZ_TIMER_DEFAULT (5) 50 ms
ign_rls_post_ans_timer	10 ms	Ignore Release After Answer Timer. Time after the call has been answered before Instant ISDN will allow the connected equipment to try to tear down the call.	IISDN56_IGNRLS_TIMER_DEFAULT (5) 50 ms
modified_in_rls_timer	50 ms	Modified Incoming Release Timer. Timer started at answer that determines when modified incoming release filter timing ends and normal incoming release filter timing begins.	IISDN56_MOD_INRLS_TIMER_DEFAULT (10) 500 ms
modified_out_rls_timer	50 ms	Modified Outgoing Release Timer. Timer started at answer that determines when modified outgoing release filter timed ends, and normal outgoing release filter timing begins.	IISDN56_MOD_OUTRLS_TIMER_DEFAULT (10) 500 ms
no_ringing_timer	50 ms	Interval between rings during the ring cycle on outbound calls (FXS Loop Start and FXS Ground Start calls only).	IISDN56_NO_RINGING_TIMER_DEFAULT (80) 4 seconds

Table 13. Robbed Bit Signaling Timers (Continued)

Timer	Granularity	Definition	Default Value
outseize_ack_timer	50 ms	CO Outseize Acknowledge Timer. Time to continue looking for the outgoing start dial signal from the connected equipment on outseizure before declaring an outseizure failure.	IISDN56_OUTS_ACK_TIMER_DEFAULT (60) 3 seconds
ringing_timer	50 ms	Duration of ring signal during the ring cycle on outbound calls (FXS Loop Start and FXS Ground Start calls only).	IISDN56_RINGING_TIMER_DEFAULT (40) 2 seconds

Timing Diagrams

To aid in the development process, timing diagrams that illustrate call setup and call teardown signaling in the various supported protocols are provided in the sections that follow. The diagrams consist of four parts:

- **IISDN SMI Messages:** Indicates the Bfv API messages sent to and received from the card during the call scenario.
- **IISDN Timers:** The timers possibly in use during the call scenario are listed, and active only during the periods on the diagram where the graph blocks are shaded.
- **Receive and Transmit Signaling Bits:** The thick black lines represent the signaling bits used to represent the call states in each protocol. Wink Start, Immediate Start/Fixed Pause and Delay protocols use only one bit (A-bit) to carry information, so only one line is shown for each direction. In these protocols, the value of the A-bit is seen as the “hook switch state” (“on hook” is low or zero, “off hook” is high or one). FXO/FXS Loop Start and Ground Start protocols use 2 signaling bits, A and B, to carry information, so two lines are shown for each direction. In these protocols, the bit values represent current feed and ring signal (FXO protocols), and hook state/ring current and ring ground (FXS protocols).
- **Arrows and Vertical Dashed Lines:** The arrows and vertical dashed lines indicate that there is a causal relationship between an event that occurs to the start of other events. For instance, the expiration of a timer can result in a change in the hook switch state, or a change in hook switch state can result in the start of a timer and issuance of an L3L4 message.

Note: The diagrams are designed to give the reader a general understanding of the sequence of events for various robbed bit protocols over time. The timers and spacing on the graphs are not proportional to the actual events that occur.

Wink Start & Delay Dial Signaling

In the Wink Start protocol, the device seizing the line expects a wink signal (a short-duration - 140 to 290ms - offhook signal) to be sent back as acknowledgment before sending the address digits.

In the Delay Dial, the device seizing the line expects an off-hook response from the far end for at least 140ms, and waits for the far end to return to on-hook state before sending the address digits.

Although the protocols are different, the sequence of signals sent by both Wink Start and Delay Dial protocols is virtually identical, so these two protocols are often handled together.

Only two signaling bit patterns are used for these protocols. Pattern 00 (A and B bits zero) indicates the line is idle, while AB = 11 indicates a line seizure on the initiating end and the off-hook signal on the receiving end. Since the two bits must always have the same value, usually only the A bit is verified and the B bit ignored, thus eliminating the need to handle invalid bit patterns.

Wink start and delay dial trunks use almost identical signaling. To configure a module for standard wink start signaling, the host must issue an L4L3mENABLE_CAS with an IISDN_ROBBED_BIT_DATA structure that contains the following settings:

- `send_glare_err_event` = 0 (not used for wink start)
- `in_trunk_type` = `IISDNttWINK_START` (the default value)
- `out_trunk_type` = `IISDNttWINK_START` (the default value)
- `fgb_fgd_mode` = 0 (refer to [Wink Start with Feature Group B & D](#) on [page 282](#) for wink start with Feature Group B and D signaling)

To configure a module for delay dial signaling, the host must issue an L4L3mENABLE_CAS with an IISDN_ROBBED_BIT_DATA structure that contains the following settings:

- `send_glare_err_event` = 0 (not used for wink start)
- `in_trunk_type` = `IISDNttDELAY_DIAL`
- `out_trunk_type` = `IISDNttDELAY_DIAL`
- `fgb_fgd_mode` = 0

The remaining settings are described in the context of incoming and outgoing calls in the subsections that follow. Modules process calls in the same manner on both wink start and delay dial trunks.

Incoming Call Processing

During an incoming call, the host receives an L3L4mPRE_SEIZE message if `send_preseize_event = 1` in the IISDN_ROBBED_BIT_DATA structure. This message is sent when the module detects an incoming seizure (off hook). The module automatically responds to the seizure by sending a wink signal, indicating it is ready to receive digits. The maximum number of digits the module expects to receive must be specified in the `max_incoming_digit_count` field. The digits are reported to the host in an L3L4mSETUP_IND message. To answer the call, the host must issue an L4L3mCONNECT_REQUEST message.

Figure 26 shows how a module processes an incoming call received over a wink start or delay dial trunk.

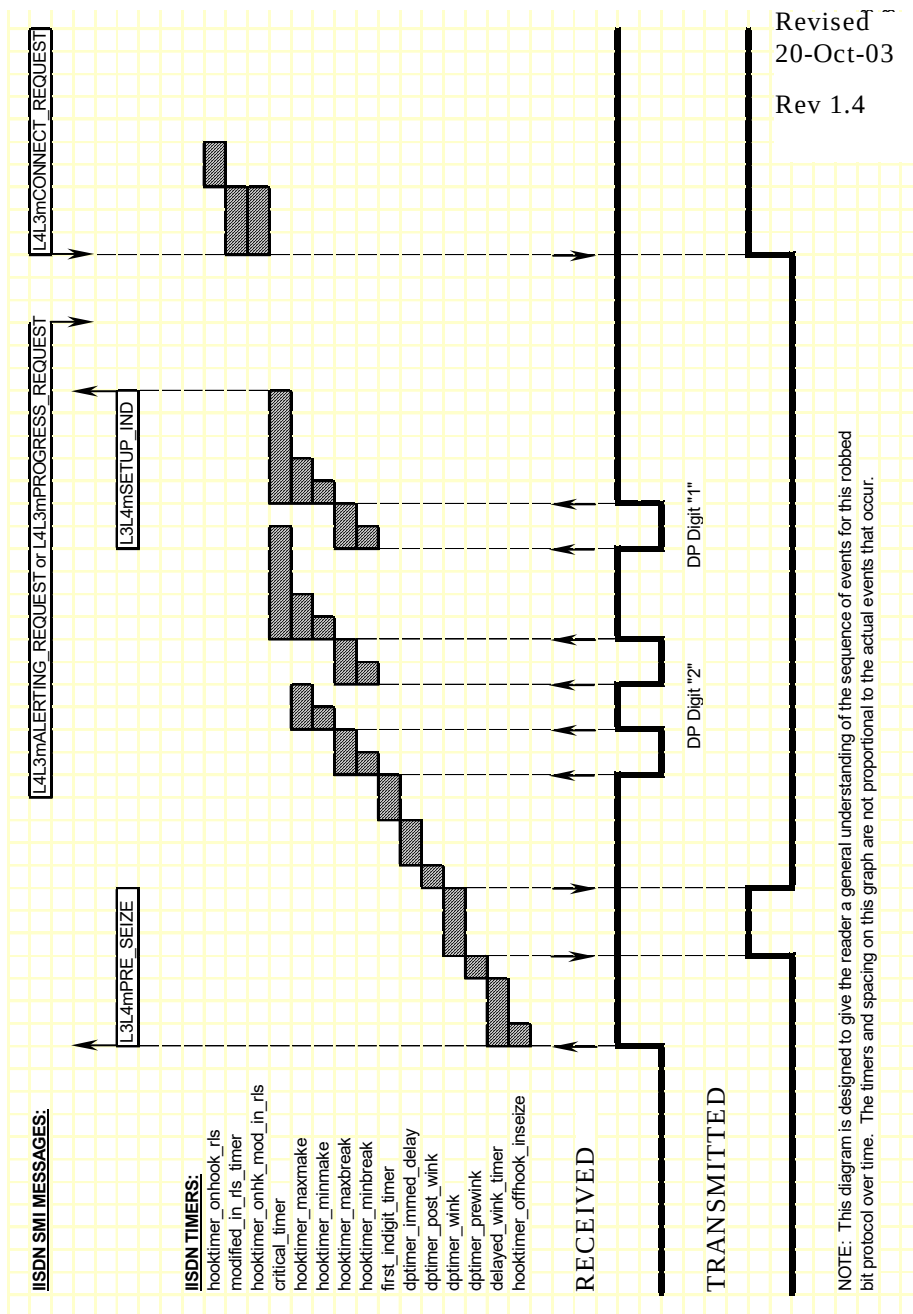


Figure 26. Wink Start and Delay Dial – Incoming Call Timing

Outgoing Call Processing

During an outgoing call, the host receives an L3L4mPROGRESS in response to the L4L3mCALL_REQUEST to start the call. The host then receives an L3L4mSEIZE_COMP message if `send_seize_comp_event = 1` in the IISDN_ROBBED_BIT_DATA structure. This message is sent when the module detects a wink from the network, indicating the network is ready to receive digits. After the digits are transmitted, the module waits for answer supervision from the network (indicating the far end has gone off hook) or for an answer timer to expire (if `timed_answer_supervision = 1`). When answer supervision is received or the timer expires, the host receives an L3L4mCONNECT indicating the call is connected.

Figure 27 shows how the module makes an outgoing call over a wink start or delay dial trunk.

Call Teardown Processing

For an incoming call teardown, refer to *Figure 47* on [page 314](#) at the end of this section; for an outgoing call teardown, refer to *Figure 48* on [page 315](#) at the end of this section.

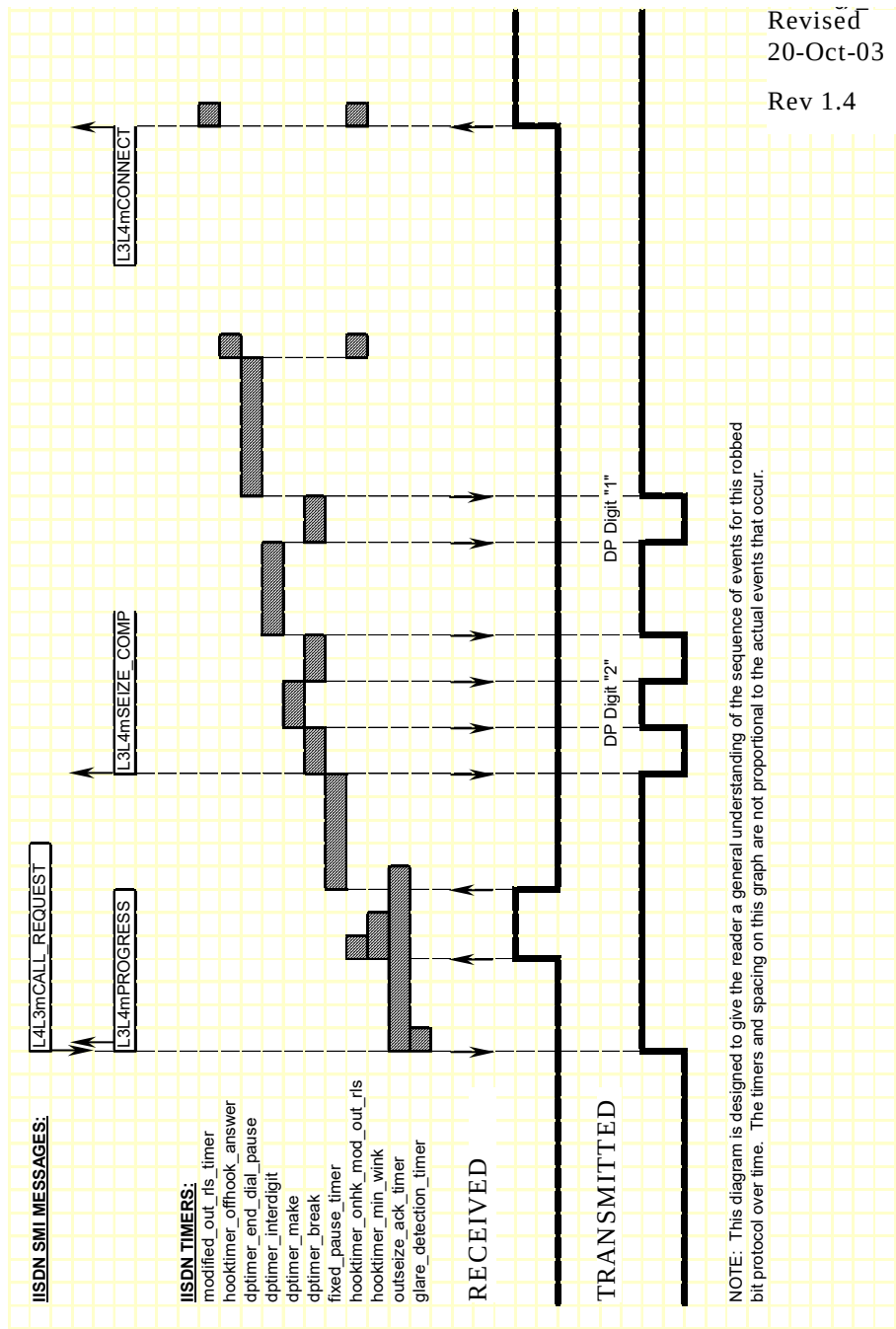


Figure 27. Wink Start and Delay Dial – Outgoing Call Timing

Wink Start with Feature Group B & D

To configure a module for wink start with Feature Group B and D signaling, the host must issue an L4L3mENABLE_CAS with an IISDN_ROBBED_BIT_DATA structure that contains the following settings:

- send_glare_err_event = 0 (not used for wink start)
- in_trunk_type = IISDNttWINK_START (the default value)
- out_trunk_type = IISDNttWINK_START (the default value)
- fgb_fgd_mode = 1 (enabling Feature Group B and D)

The remaining settings are described in the context of incoming and outgoing calls in the subsections that follow.

Incoming Call Processing

During an incoming call, the host receives an L3L4mPRE_SEIZE message if send_preseize_event = 1 in the IISDN_ROBBED_BIT_DATA structure. This message is sent when the module detects an incoming seizure (off hook). The module automatically responds to the seizure by sending a wink, indicating it is ready to receive digits. The maximum number of digits the module expects to receive must be specified in the max_incoming_digit_count field. The digits are reported to the host in an L3L4mSETUP_IND message.

To acknowledge the receipt of digits, the host must issue an L4L3mTX_WINK message to send another wink to the network. To answer the call, the host must issue an L4L3mCONNECT_REQUEST message.

Figure 28 shows how a module processes an incoming Feature Group B and D call received over a wink start trunk.

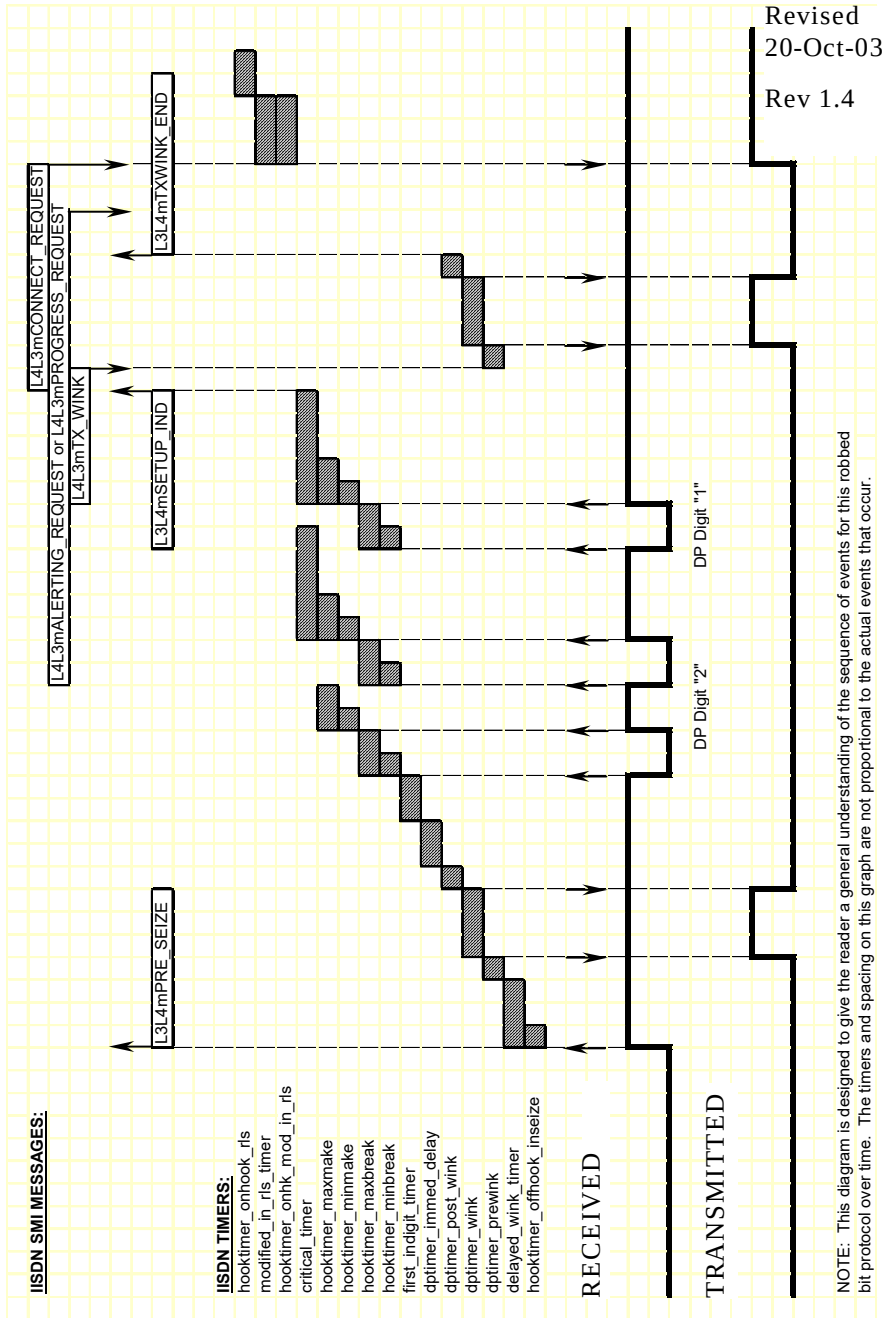


Figure 28. Wink Start with Feature Group B&D - Incoming Call Timing Diagram

Outgoing Call Processing

The host receives an L3L4mPROGRESS in response to the L4L3mCALL_REQUEST to start the call. The host then receives an L3L4mRX_WINK message when the module detects a wink from the network. The module waits a period of time (determined by the fixed pause timer) before sending digits. When the timer expires, the host receives an L3L4mSEIZE_COMP message if send_seize_comp_event = 1 in the IISDN_ROBBED_BIT_DATA structure.

Call Teardown Processing

For an incoming call teardown, refer to [Figure 47](#) at the end of this section; for an outgoing call teardown, refer to [Figure 48](#) at the end of this section.

Wink start with Feature B and D is designed to support transmitting digits other than dial pulse digits. If called_party_digits = 0 in the L4L3mCALL_REQUEST that started the call, the module waits in outdialing state indefinitely. During that time, another device connected to the module over the TDM bus (such as a digital tone generator) is used to output digits to the network. When that device is finished transmitting, the host must issue an L4L3mEND_DIAL message to make the module continue call processing.

After the digits are transmitted, the network acknowledges their receipt by sending a wink to the module. The host receives an L3L4mRX_WINK when this wink is detected. The module then waits for answer supervision from the network (indicating the far end has gone off hook) or for an answer timer to expire (if timed_answer_supervision = 1). When answer supervision is received or the timer expires, the host receives an L3L4mCONNECT indicating the call is connected.

[Figure 29](#) shows how the module makes an outgoing Feature Group B/D call over a wink start trunk.

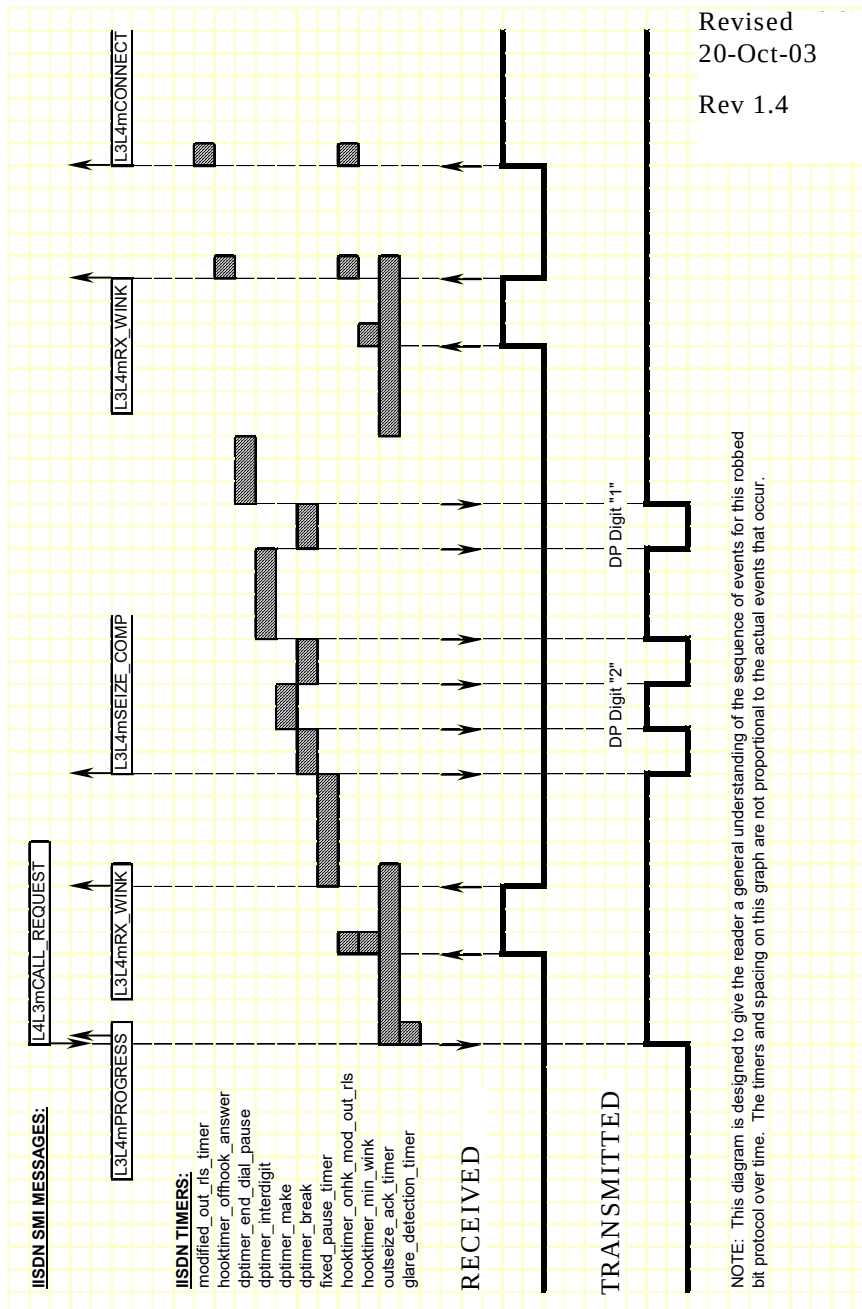


Figure 29. Wink Start with Feature Group B&D - Outgoing Call Timing Diagram

Immediate Start/Fixed Pause Signaling

Immediate Start signaling applies to incoming calls only; Fixed Pause signaling applies to outgoing calls only. Since these signaling types functionally complement each other, they are handled together by modules. The exact protocol to be used for a call is automatically selected depending on whether an incoming or outgoing call is being handled. To avoid confusion, the symbolic constants used to select these two protocols, `IISDNttIMMEDIATE_DIAL` and `IISDNttFIXED_PAUSE`, are set to the same value.

To configure a module for immediate start and fixed pause mode signaling, the host must issue an `L4L3mENABLE_CAS` with an `IISDN_ROBBED_BIT_DATA` structure that contains the following settings:

- `send_glare_err_event` = 0 (not used)
- `in_trunk_type` = `IISDNttIMMEDIATE_DIAL` or `IISDNttFIXED_PAUSE`
- `out_trunk_type` = `IISDNttIMMEDIATE_DIAL` or `IISDNttFIXED_PAUSE`
- `fgb_fgd_mode` = 0 (not used)

The remaining settings are described in the context of incoming and outgoing calls in the subsections that follow.

Incoming Call Processing (Immediate Start)

During an incoming call, the host receives an `L3L4mPRE_SEIZE` message if `send_preseize_event` = 1 in the `IISDN_ROBBED_BIT_DATA` structure. This message is sent when the module detects an incoming seizure (off hook). The module then collects dial pulse digits from the network; the maximum number of digits the module expects to receive must be specified in the `max_incoming_digit_count` field. The digits are reported to the host in an `L3L4mSETUP_IND` message. To answer the call, the host must issue an `L4L3mCONNECT_REQUEST` message.

Figure 30 shows how a module processes an incoming call received over an immediate start trunk.

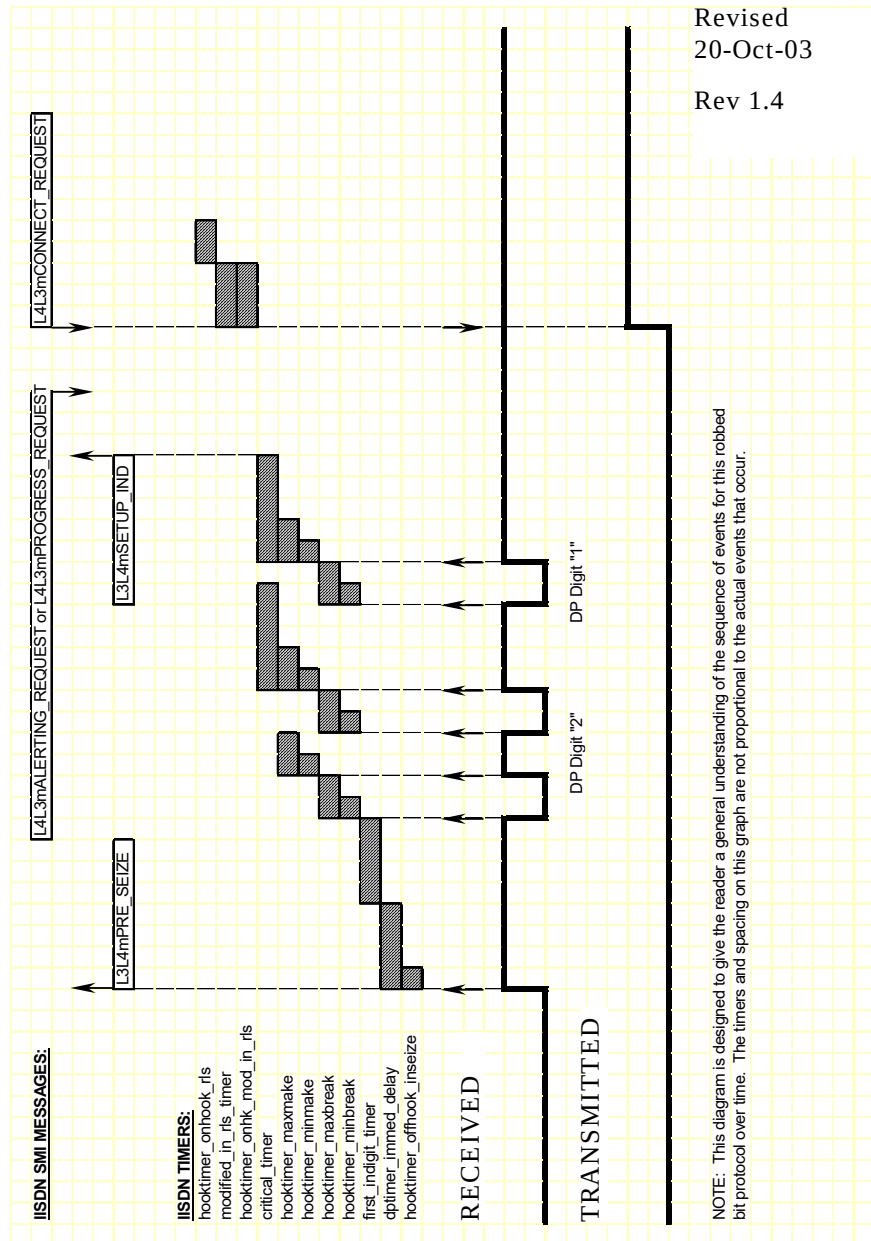


Figure 30. Immediate Start – Incoming Call Timing Diagram

Outgoing Call Processing (Fixed Pause Mode)

During an outgoing call, the host receives an L3L4mPROGRESS in response to the L4L3mCALL_REQUEST to start the call. The module waits a period of time (determined by the fixed pause timer) before sending digits. When the timer expires, the host receives an L3L4mSEIZE_COMP message if send_seize_comp_event = 1 in the IISDN_ROBBED_BIT_DATA structure. After transmitting digits, the module waits for answer supervision from the network (indicating the far end has gone off hook) or for an answer timer to expire (if timed_answer_supervision = 1). When answer supervision is received or the timer expires, the host receives an L3L4mCONNECT indicating the call is connected.

Figure 31 shows how the module makes an outgoing call over a fixed pause mode trunk.

Call Teardown Processing

For an incoming call teardown, refer to *Figure 47* at the end of this section; for an outgoing call teardown, refer to *Figure 48* at the end of this section.

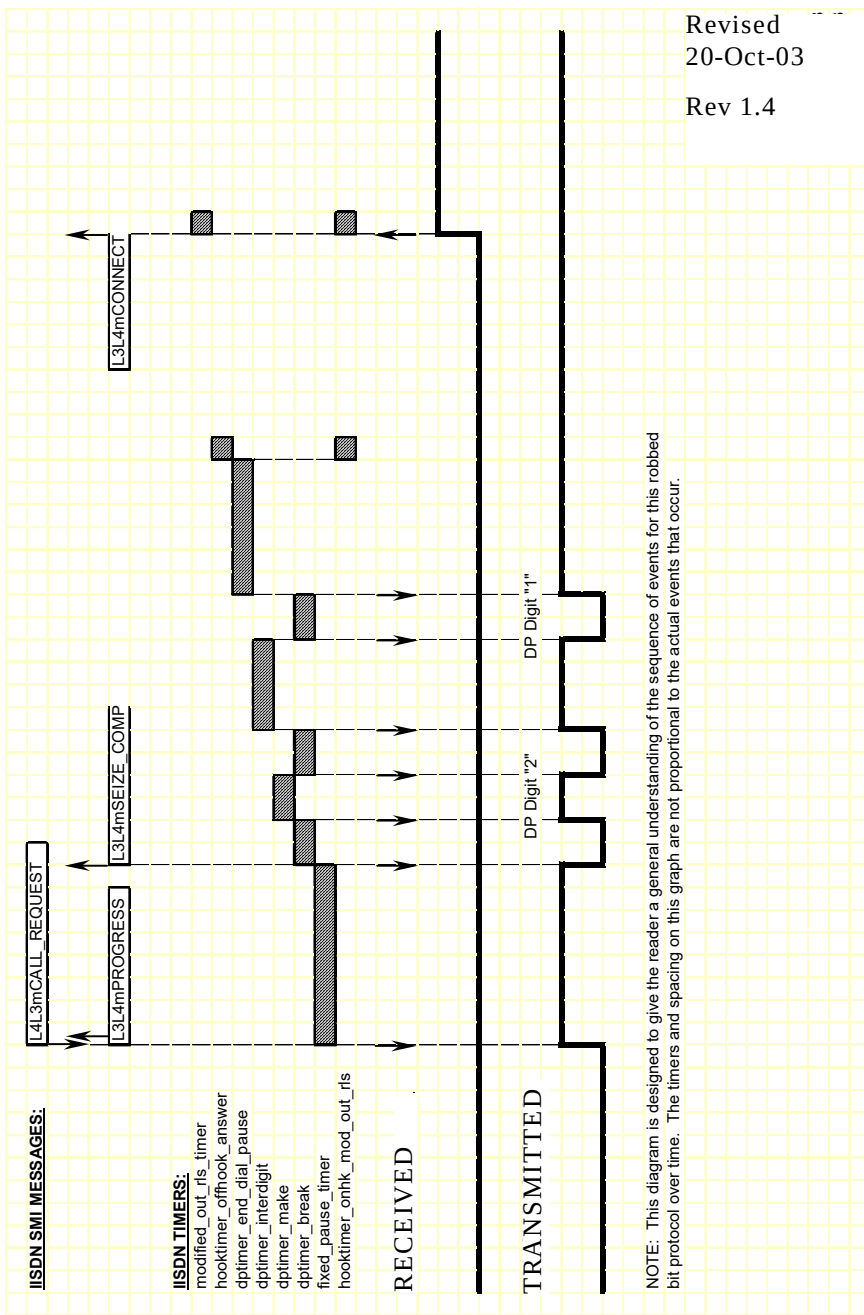


Figure 31. Fixed Pause – Outgoing Call Timing Diagram

Ground Start Signaling

Modules support two types of ground start signaling:

- Foreign Exchange - Office (FXO)
- Foreign Exchange - Subscriber (FXS)

When operating in FXO mode, the module assumes the far end of the connection is an FXS termination. When operating in FXS mode, the module assumes the far end is an FXO termination. The timing diagrams on the pages that follow use the labels “FXO” or “FXS” rather than “Rx” or “Tx” for hookswitch states to illustrate this point.

FXO Ground Start

To configure a module for FXO ground start signaling, the host must issue an L4L3mENABLE_B_CHANNEL with an IISDN_ROBBED_BIT_DATA structure that contains the following settings:

- in_trunk_type = IISDNttFXO_GDSTART
- out_trunk_type = IISDNttFXO_GDSTART
- timed_answer_supervision = 1 (timer expiration indicates far end answer)
- fgb_fgd_mode = 0 (not used)

The remaining settings are described in the context of incoming and outgoing calls in the subsections that follow.

Incoming Call Processing

During an incoming call, the host receives an L3L4mPRE_SEIZE message if send_preseize_event = 1 in the IISDN_ROBBED_BIT_DATA structure. This message is sent when the module detects ring ground at the far end (FXS end). The module then collects dial pulse digits from the network; the maximum number of digits the module expects to receive must be specified in the max_incoming_digit_count field. The digits are reported to the host in an L3L4mSETUP_IND message. To answer the call, the host must issue an L4L3mCONNECT_REQUEST message.

Figure 32 shows how a module processes an incoming call received over an FXO ground start trunk.

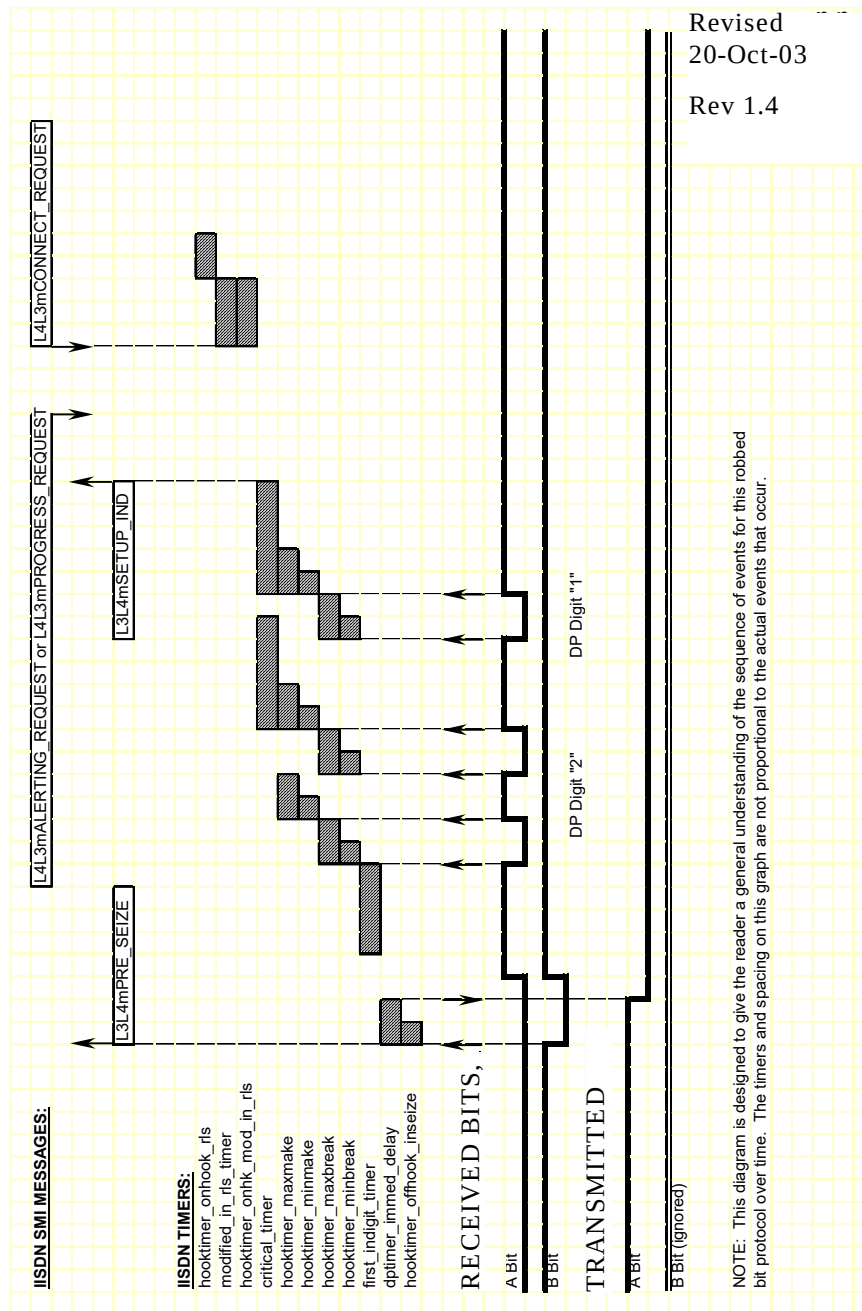


Figure 32. FXO Ground Start – Incoming Call Timing Diagram

Outgoing Call Processing

If `send_glare_err_event = 0` in the `IISDN_ROBBED_BIT_DATA` structure, the host receives an `L3L4mERROR` message containing the value `L3L4errGLARE` if the outgoing call attempt fails due to glare. Glare occurs when the module attempts to make an outgoing call on a channel at the same time an incoming call arrives on the channel. The call attempt fails because the network always wins channel contention.

During an outgoing call, the host receives an `L3L4mPROGRESS` in response to the `L4L3mCALL_REQUEST` to start the call. The host then receives an `L3L4mSEIZE_COMP` message if `send_seize_comp_event = 1` in the `IISDN_ROBBED_BIT_DATA` structure. This message is sent after the fixed pause timer expires. No digits are collected in FXO ground start mode; the module simply waits for an answer timer to expire (if `timed_answer_supervision = 1`). When the timer expires, the host receives an `L3L4mCONNECT` indicating the call is connected.

Figure 33 shows how the module makes an outgoing call over an FXO ground start trunk.

Call Teardown Processing

For an incoming call teardown, refer to *Figure 34*; for an outgoing call teardown, refer to *Figure 35*.

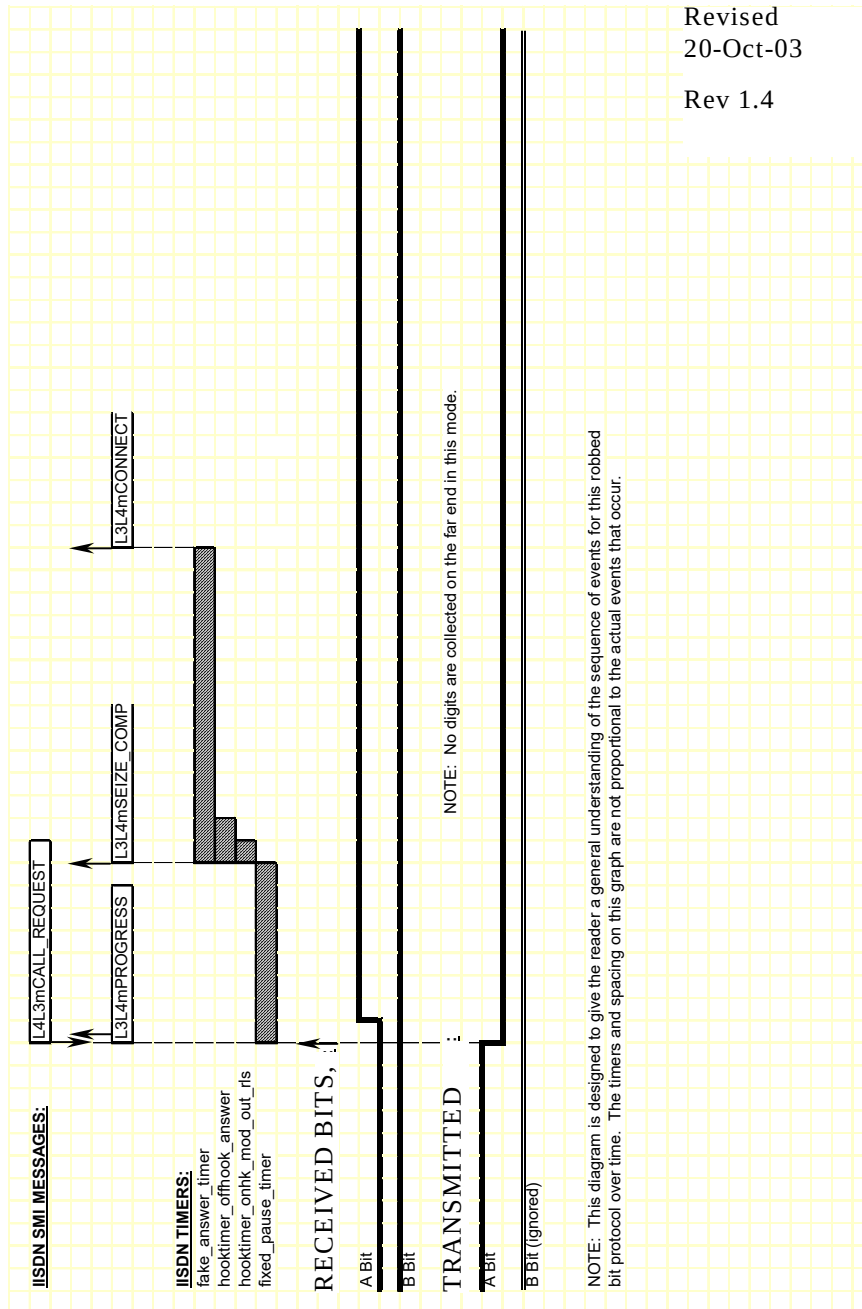


Figure 33. FXO Ground Start – Outgoing Call Timing Diagram

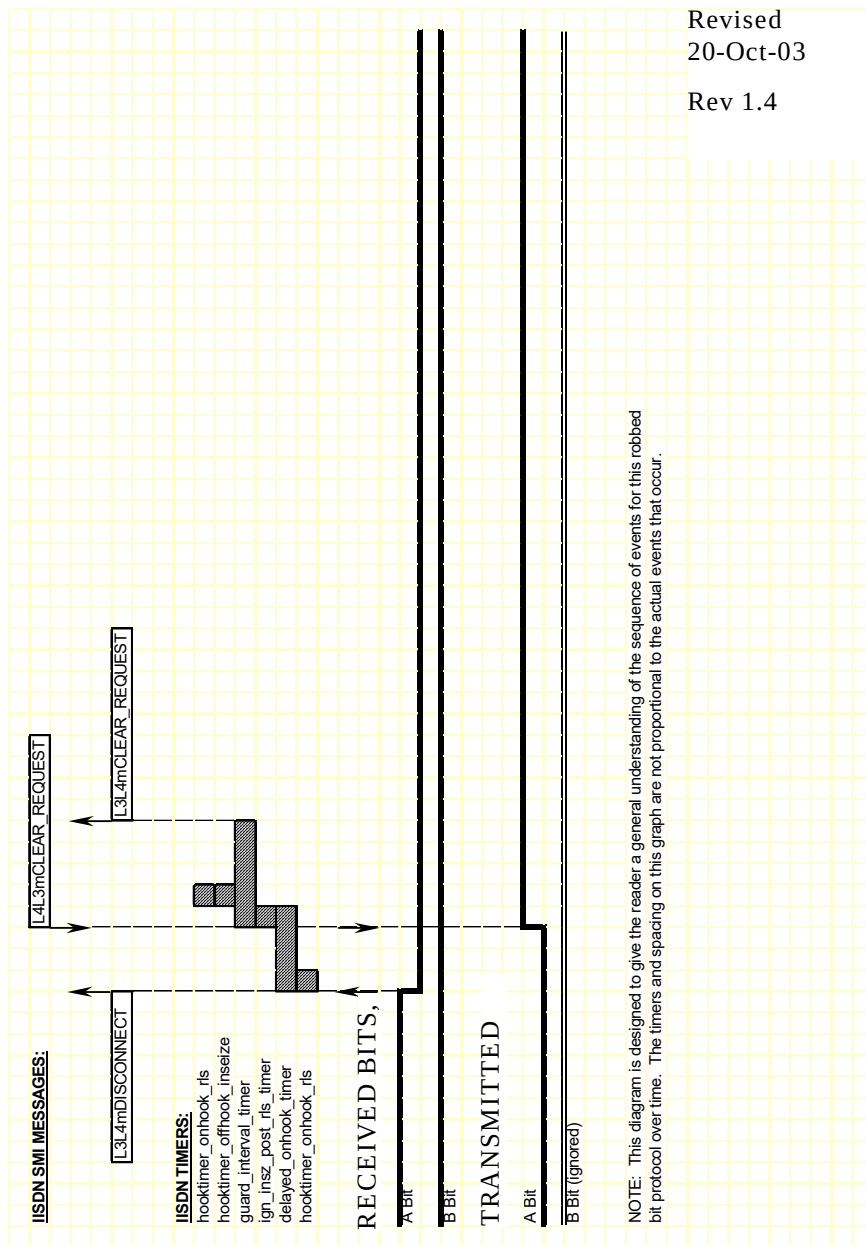


Figure 34. FXO Ground Start - Incoming Clear Timing Diagram

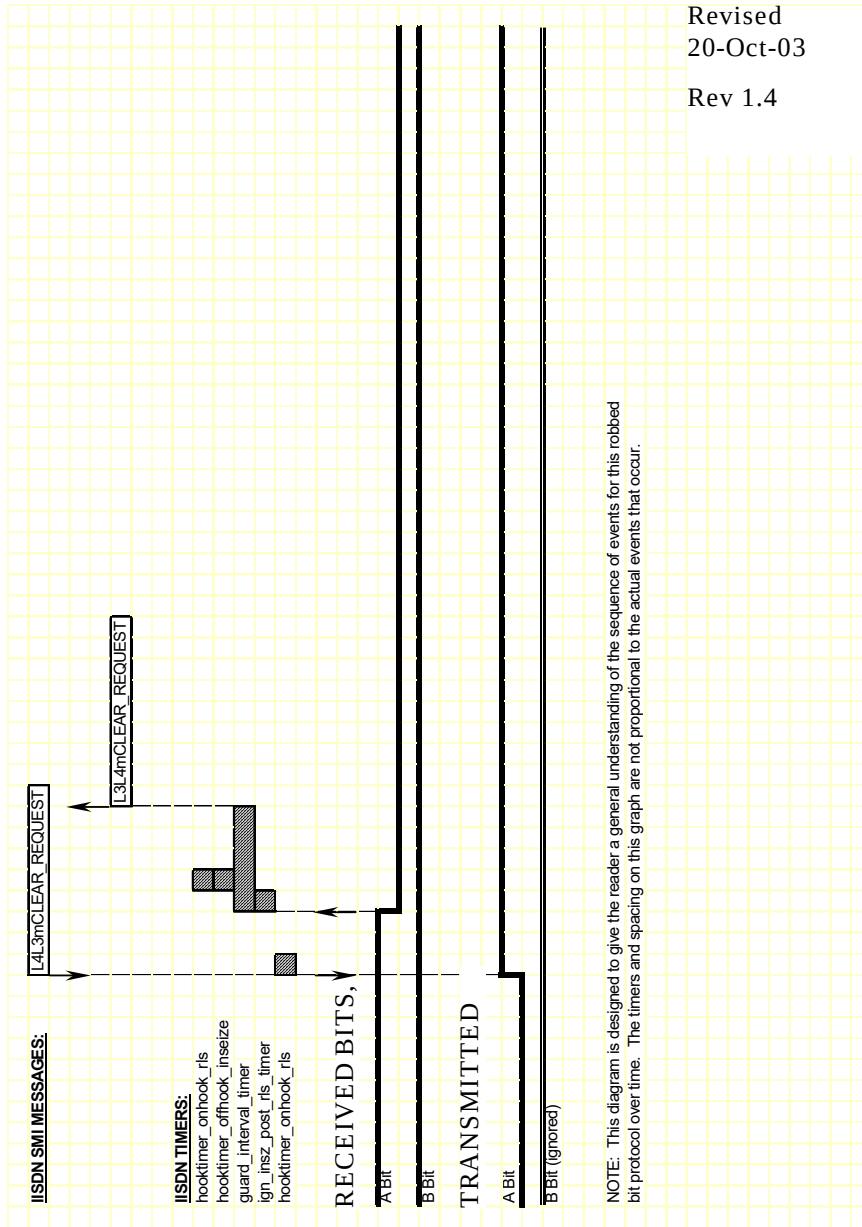


Figure 35. FXO Ground Start - Outgoing Clear Timing Diagram

FXS Ground Start

To configure a module for FXS ground start signaling, the host must issue an L4L3mENABLE_CAS with an IISDN_ROBBED_BIT_DATA structure that contains the following settings:

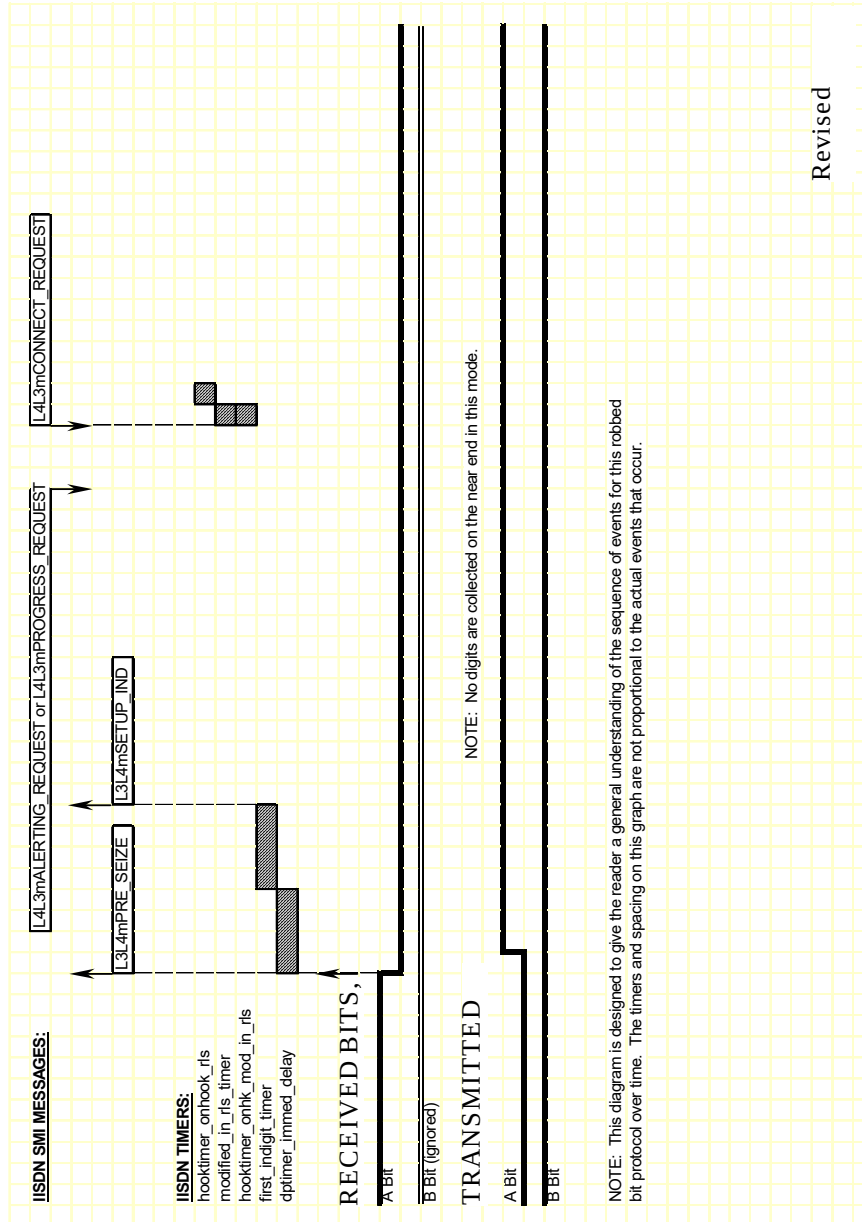
- in_trunk_type = IISDNtt5ESS_GDSTART
- out_trunk_type = IISDNtt5ESS_GDSTART
- max_incoming_digit_count = 0 (no digits collected)
- timed_answer_supervision = 1 (timer expiration indicates far end answer)
- fgb_fgd_mode = 0 (not used)

The remaining settings are described in the context of incoming and outgoing calls in the subsections that follow.

Incoming Call Processing

During an incoming call, the host receives an L3L4mPRE_SEIZE message if send_preseize_event = 1 in the IISDN_ROBBED_BIT_DATA structure. This message is sent when the module detects an incoming seizure. No digits are collected in FXS ground start mode; to answer the call, the host must issue an L4L3mCONNECT_REQUEST message.

Figure 36 shows how a module processes an incoming call received over an FXS ground start trunk.



Revised

Figure 36. FXS Ground Start – Incoming Call Timing Diagram

Outgoing Call Processing

If `send_glare_err_event = 0` in the `IISDN_ROBBED_BIT_DATA` structure, the host receives an `L3L4mERROR` message containing the value `L3L4errGLARE` if the outgoing call attempt fails due to glare. Glare occurs when the module attempts to make an outgoing call on a channel at the same time an incoming call arrives on the channel. The call attempt fails because the network always wins channel contention.

During an outgoing call, the host receives an `L3L4mPROGRESS` in response to the `L4L3mCALL_REQUEST` to start the call. The module starts the fixed pause timer when it detects tip ground at the far end (FXO end). When the timer expires, the host receives an `L3L4mSEIZE_COMP` message if `send_seize_comp_event = 1` in the `IISDN_ROBBED_BIT_DATA` structure.

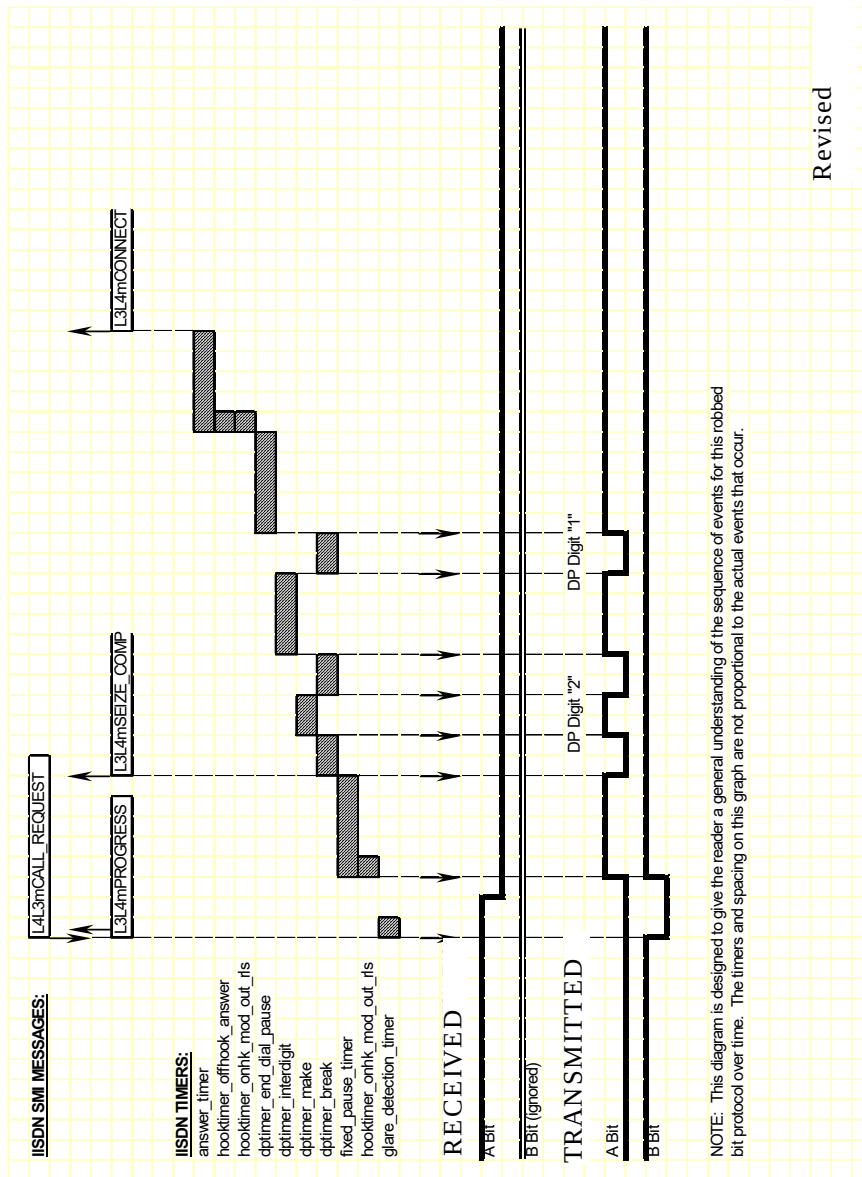
FXS ground start supports transmitting digits other than dial pulse digits. If `called_party_digits = 0` in the `L4L3mCALL_REQUEST` that started the call, the module waits in outdialing state indefinitely. During that time, another device connected to the module over the TDM bus (such as a digital tone generator) is used to outpulse digits to the network.

After the digits are transmitted, the module waits for an answer timer to expire (if `timed_answer_supervision = 1`). When the timer expires, the host receives an `L3L4mCONNECT` indicating the call is connected.

[Figure 37](#) shows how the module makes an outgoing call over an FXS ground start trunk.

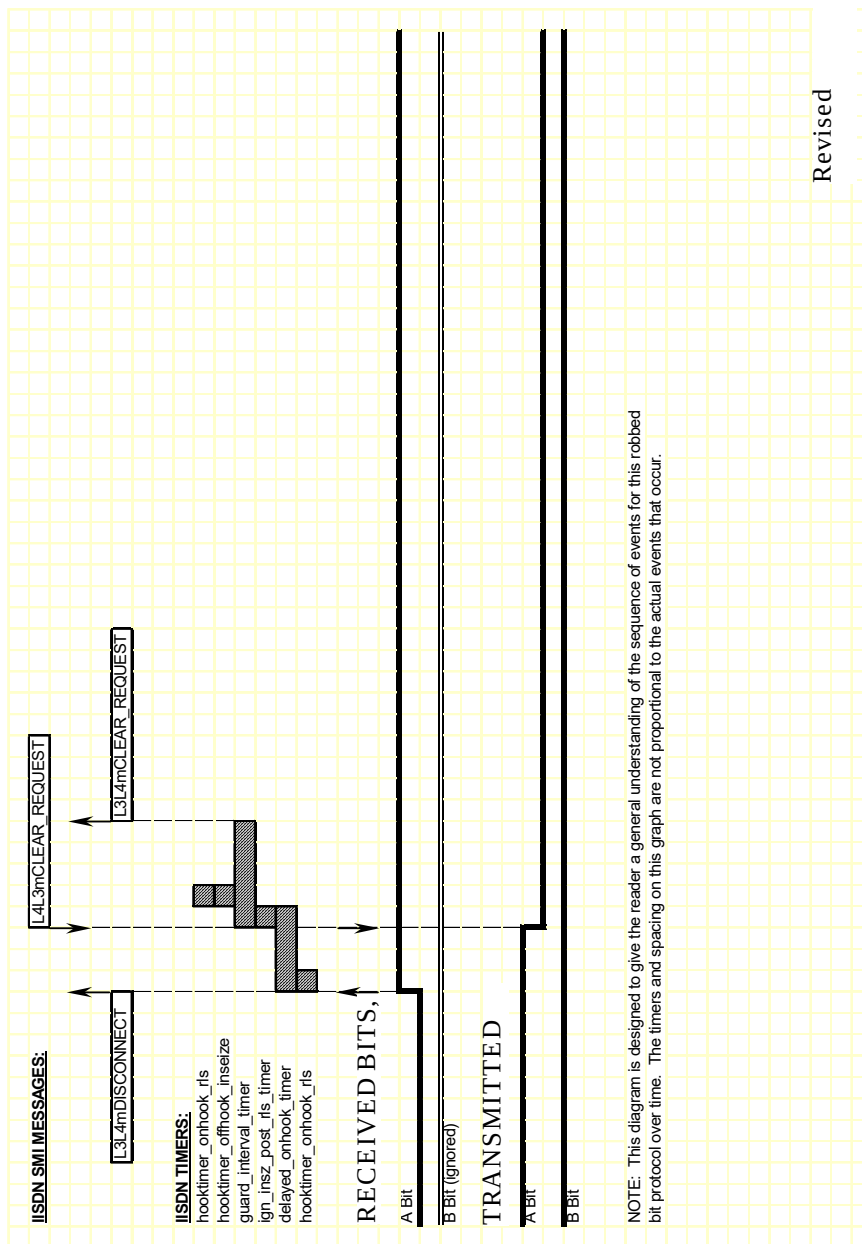
Call Teardown Processing

For an incoming call teardown, refer to [Figure 38](#); for an outgoing call teardown, refer to [Figure 39](#).



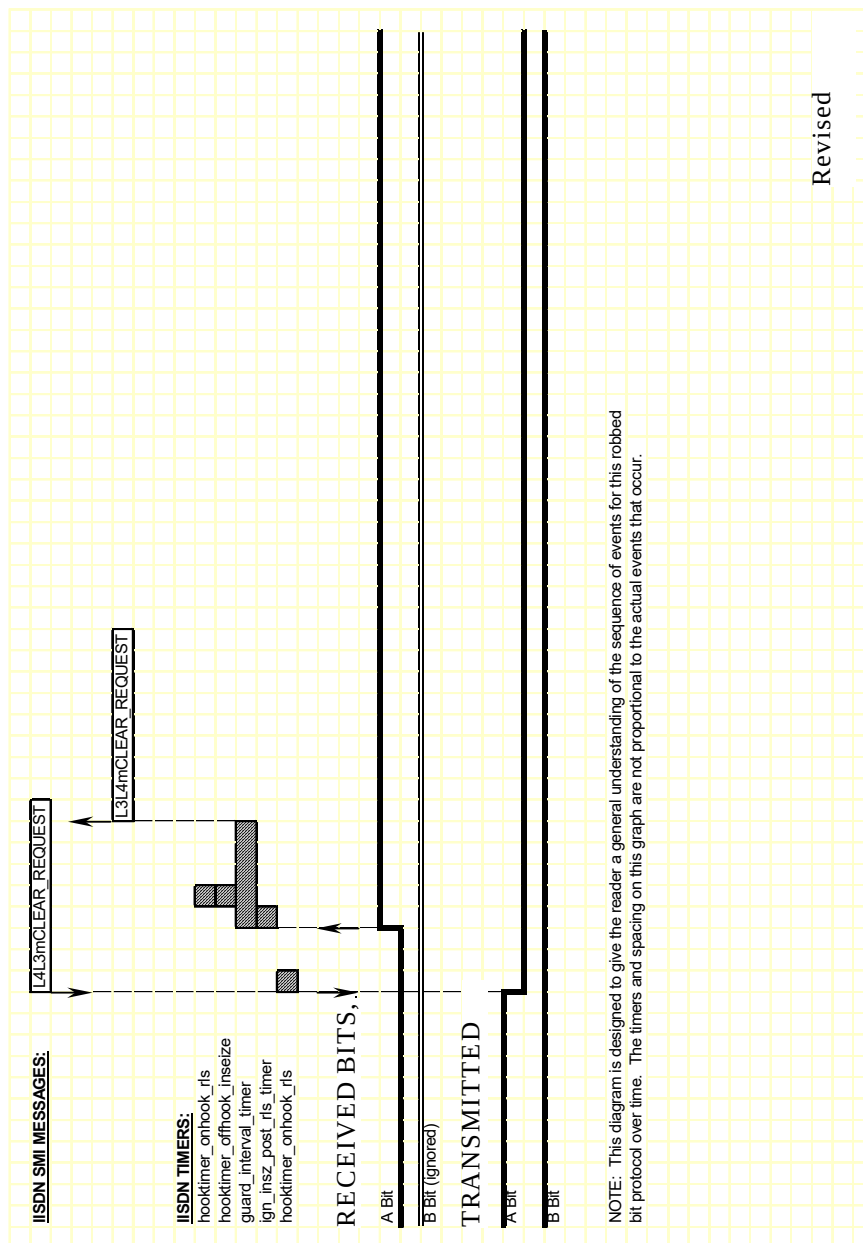
Revised

Figure 37. FXS Ground Start – Outgoing Call Timing Diagram



Revised

Figure 38. FXS Ground Start - Incoming Clear Timing Diagram



Revised

Figure 39. FXS Ground Start - Outgoing Clear Timing Diagram

Loop Start Signaling

Modules support two types of loop start signaling:

- FXO Loop Start
- FXS Loop Start

When operating in FXO mode, the module assumes the far end of the connection is an FXS termination. When operating in FXS mode, the module assumes the far end is an FXO termination.

FXS Loop Start is the protocol used for Brooktrout's analog modules. The sequence of events and timings is the same for both analog or digital line interfaces.

The timing diagrams in this section show the Receive and Transmit Signaling Bits in reference to the digital signaling mode. The analog signals are a direct mapping of the signaling bits.

In the FXS diagrams, the electric signals sent by the station are represented in bit A: loop current presence (off-hook) is represented by setting bit A high (one); absence (on-hook) by bit A low (zero). Bit B does not change. The signals received by the station, ring signal and loop current feed (battery), are represented using both A and B bits: loop current feed is represented by setting bit A low (zero); absence by bit A high (one). Presence of ring signal is represented by setting bit B low (zero); absence by bit B high (one).

The FXO diagrams are the mirror-image of the FXS diagrams. The signals sent by the module (FXO) toward the network (FXS) are the same that the station (FXS) received from the network (FXO) on the FXS diagrams.

FXO Loop Start

To configure a module for FXO loop start signaling, the host must issue an L4L3mENABLE_CAS with an IISDN_ROBBED_BIT_DATA structure that contains the following settings:

- send_glare_err_event = 0 (not used for loop start)
- in_trunk_type = IISDNttFXO_LPSTART
- out_trunk_type = IISDNttFXO_LPSTART
- timed_answer_supervision = 0 (answer supervision required)
- fgb_fgd_mode = 0 (not used)

The remaining settings are described in the context of incoming and outgoing calls in the subsections that follow.

Note: The FXO end of a loop start connection uses the same A and B bit values for both idle and connected states. Because of this, the module cannot detect when the FXO end goes on hook. To idle the channel, the host must detect the presence of a busy/reorder tone coming from the FXO end and issue an L4L3mCLEAR_REQUEST to disconnect the call. To ensure the FXO end returns to an idle state, the host must also reset the channel by issuing an L4L3mDISABLE_CAS immediately followed by an L4L3mENABLE_CAS in order to make another call.

Incoming Call Processing

During an incoming call, the host receives an L3L4mPRE_SEIZE message if `send_preseize_event = 1` in the IISDN_ROBBED_BIT_DATA structure. This message is sent when the module detects an incoming seize from the far end (FXS end). The module then collects dial pulse digits from the network; the maximum number of digits the module expects to receive must be specified in the `max_incoming_digit_count` field. The digits are reported to the host in an L3L4mSETUP_IND message. To answer the call, the host must issue an L4L3mCONNECT_REQUEST message. [Figure 40](#) shows how a module processes an incoming call received over an FXO loop start trunk.

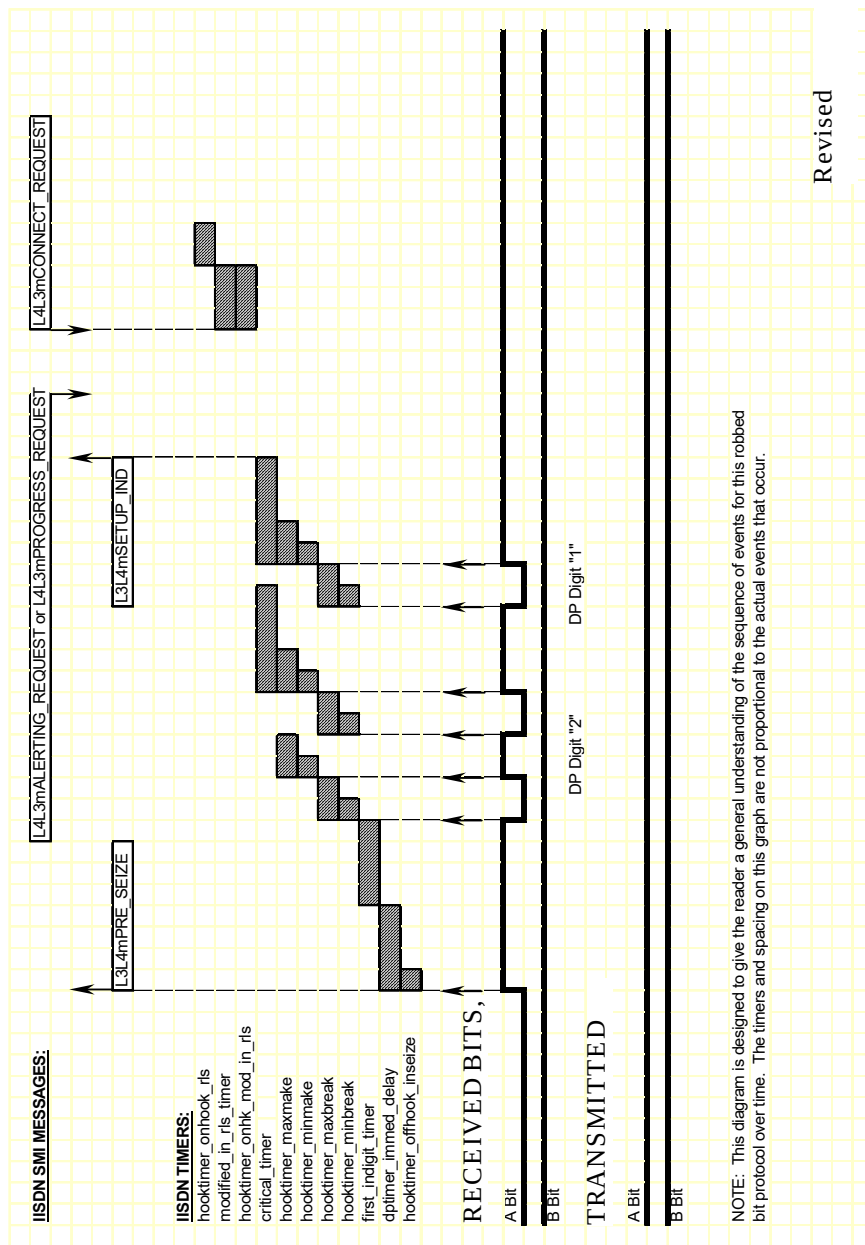
Outgoing Call Processing

During an outgoing call, the host receives an L3L4mPROGRESS in response to the L4L3mCALL_REQUEST to start the call. The module automatically transmits a ringing pattern using the B signaling bit. When the far end goes off hook, the host receives an L3L4mCONNECT indicating the call is connected.

[Figure 41](#) shows how the module makes an outgoing call over an FXO ground start trunk.

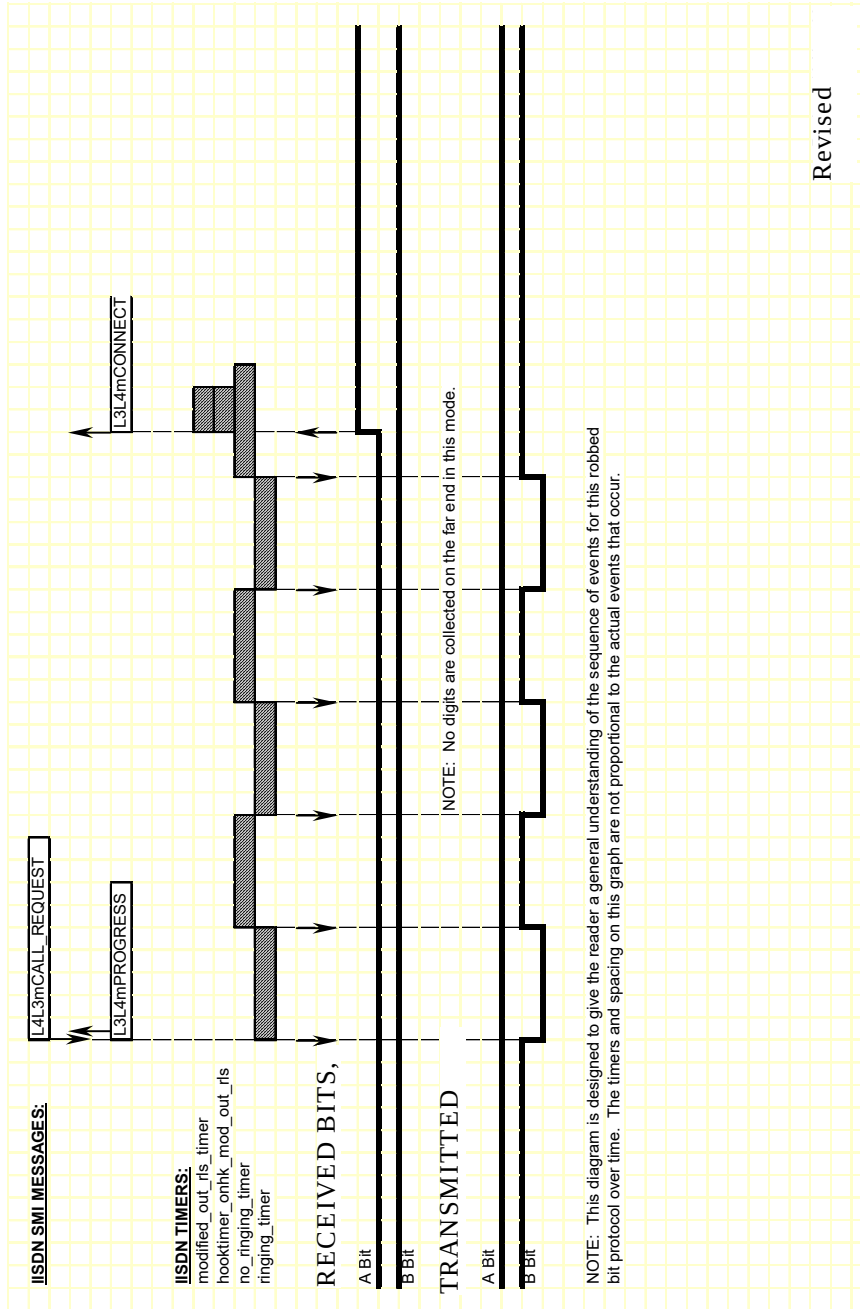
Call Teardown Processing

For an incoming call teardown, refer to [Figure 42](#); for an outgoing call teardown, refer to [Figure 43](#).



Revised

Figure 40. FXO Loop Start – Incoming Call Timing Diagram



Revised

Figure 41. FXO Loop Start – Outgoing Call Timing Diagram

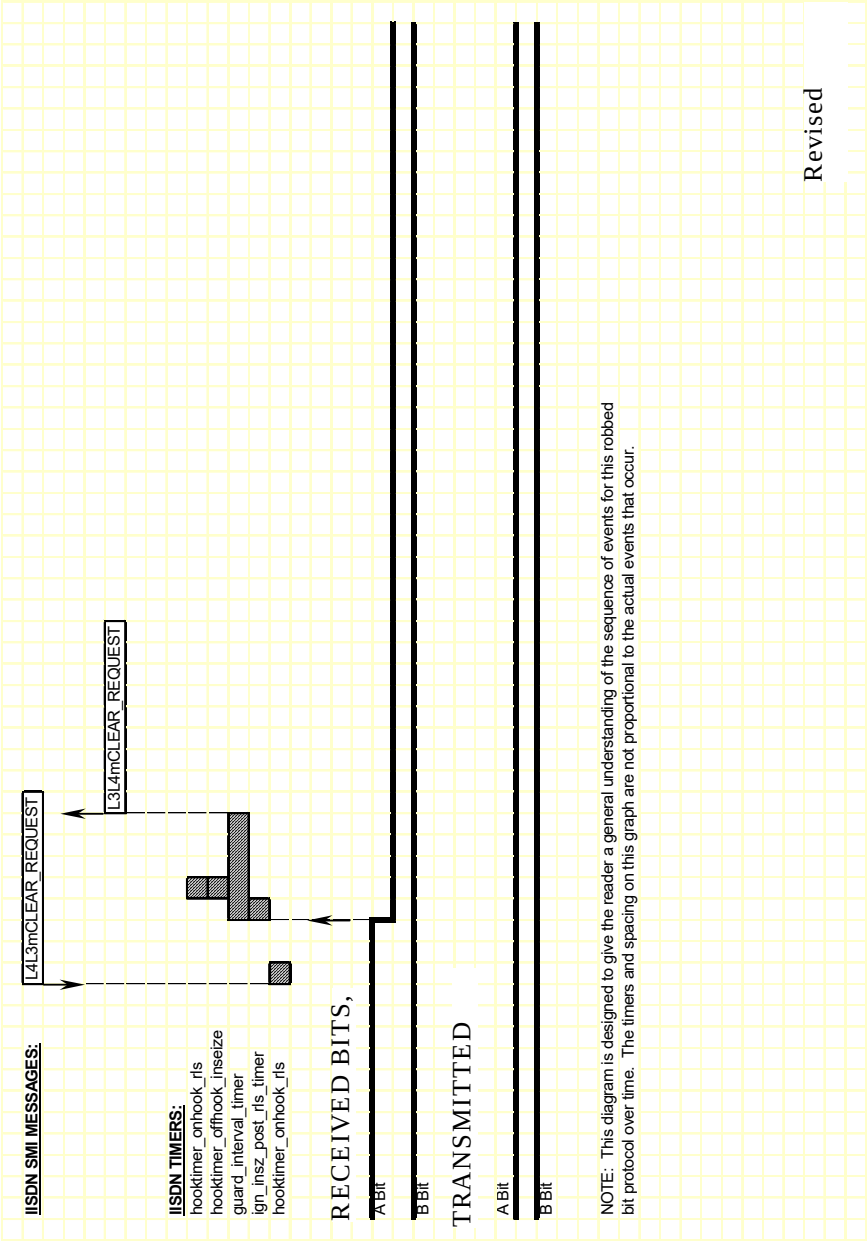


Figure 42. FXO Loop Start – Outgoing Clear Timing Diagram

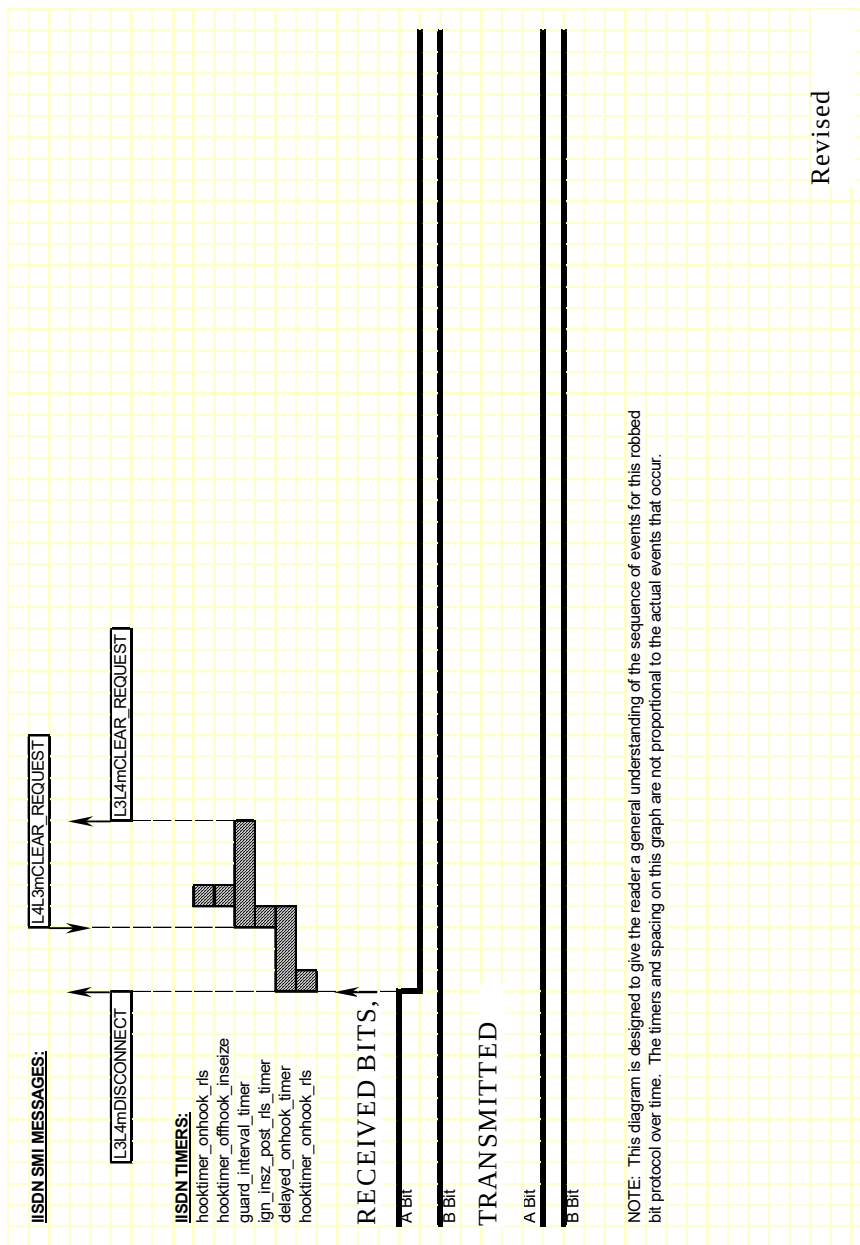


Figure 43. FXO Loop Start – Incoming Clear Timing Diagram

FXS Loop Start

To configure a module for FXS loop start signaling, the host must issue an L4L3mENABLE_CAS with an IISDN_ROBBED_BIT_DATA structure that contains the following settings:

- in_trunk_type = IISDNtt5ESS_LPSTART
- out_trunk_type = IISDNtt5ESS_LPSTART
- max_incoming_digit_count = 0 (no digits collected)
- timed_answer_supervision = 1 (timer expiration indicates far end answer)
- fgb_fgd_mode = 0 (not used)

The remaining settings are described in the context of incoming and outgoing calls in the subsections that follow.

Note: The FXO end of a loop start connection uses the same A and B bit values for both idle and connected states. Because of this, the module cannot detect when the FXO end goes on hook. To idle the channel, the host must detect the lack of data coming from the FXO end and issue an L4L3mCLEAR_REQUEST to disconnect the call. To ensure the FXO end returns to an idle state, the host must also reset the channel by issuing an L4L3mDISABLE_CAS immediately followed by an L4L3mENABLE_CAS in order to make another call.

Incoming Call Processing

During an incoming call, the host receives an L3L4mPRE_SEIZE message if send_preseize_event = 1 in the IISDN_ROBBED_BIT_DATA structure. This message is sent when the module detects ringing from the far end (FXO end). No digits are collected in FXS loop start mode; to answer the call, the host must issue an L4L3mCONNECT_REQUEST message.

Figure 44 shows how a module processes an incoming call received over an FXS loop start trunk.

Outgoing Call Processing

During an outgoing call, the host receives an L3L4mPROGRESS in response to the L4L3mCALL_REQUEST to start the call. The host then receives an L3L4mSEIZE_COMP message if send_seize_comp_

event = 1 in the IISDN_ROBBED_BIT_DATA structure. This message is sent after the fixed pause timer expires. When the timer expires, the host receives an L3L4mSEIZE_COMP message if send_seize_comp_event = 1 in the IISDN_ROBBED_BIT_DATA structure.

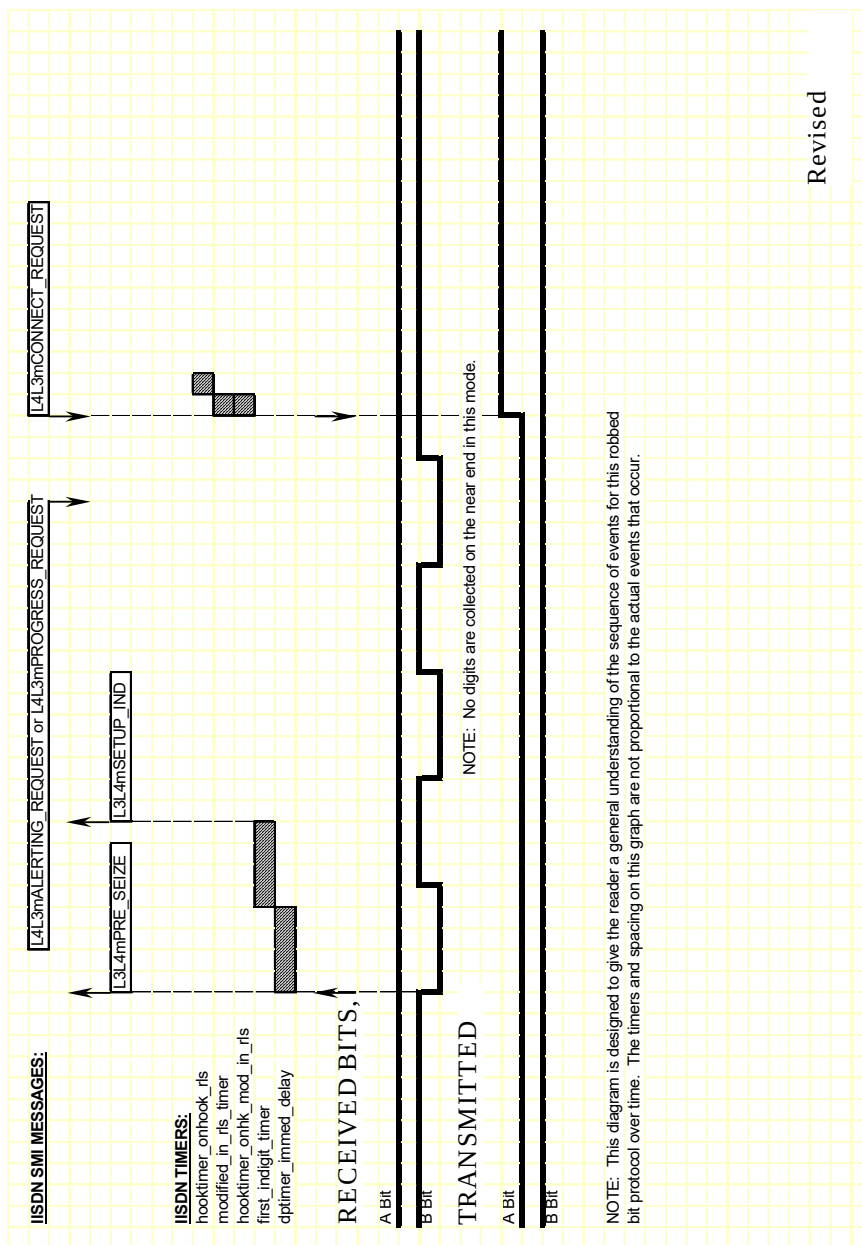
FXS ground start supports transmitting digits other than dial pulse digits. If called_party_digits = 0 in the L4L3mCALL_REQUEST that started the call, the module waits in outdialing state indefinitely. During that time, another device connected to the module over the TDM bus (such as a digital tone generator) is used to output digits to the network.

After the digits are transmitted, the module waits for an answer timer to expire (if timed_answer_supervision = 1). When the timer expires, the host receives an L3L4mCONNECT indicating the call is connected.

Figure 45 shows how the module makes an outgoing call over an FXS ground start trunk.

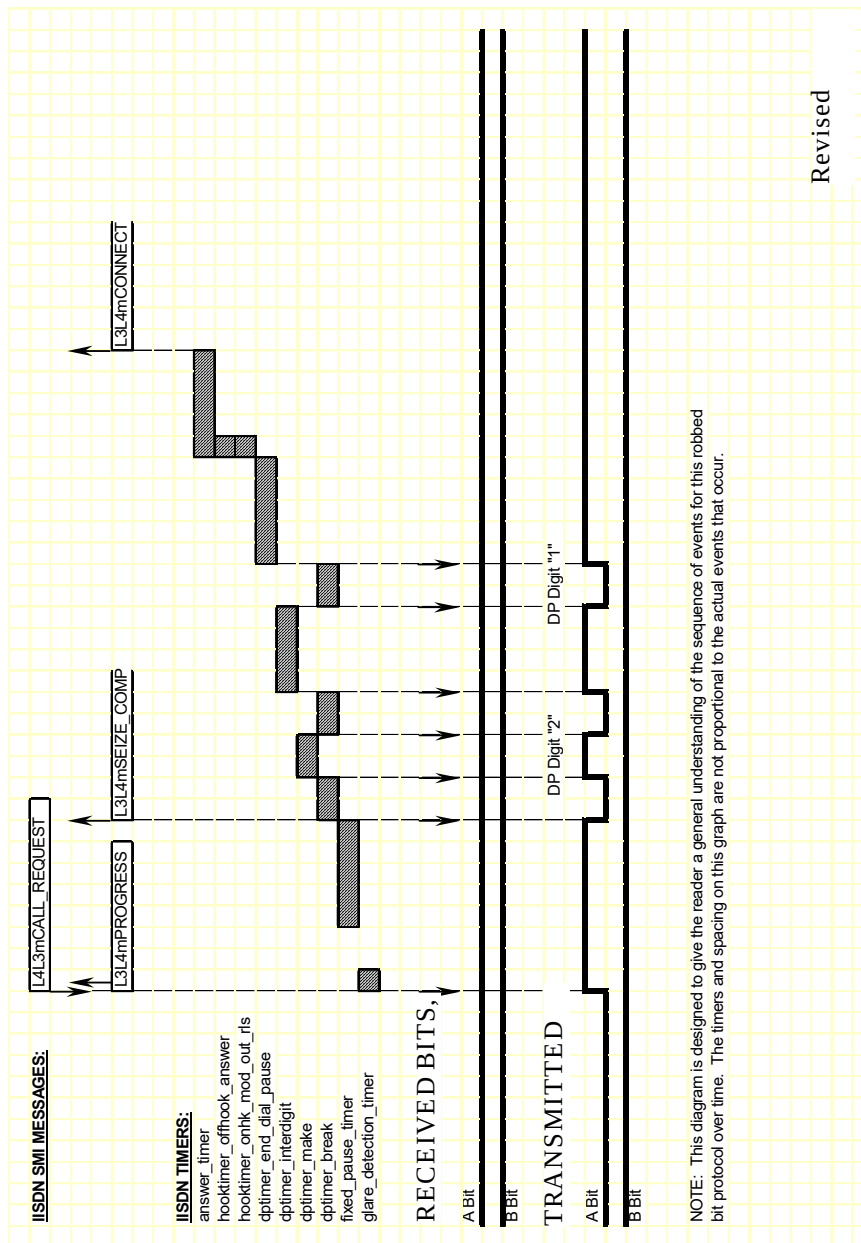
Call Teardown Processing

In this mode, an incoming call teardown cannot be detected; the host must determine when the call is inactive and initiate an outgoing call teardown. For an outgoing call teardown, refer to *Figure 46*.



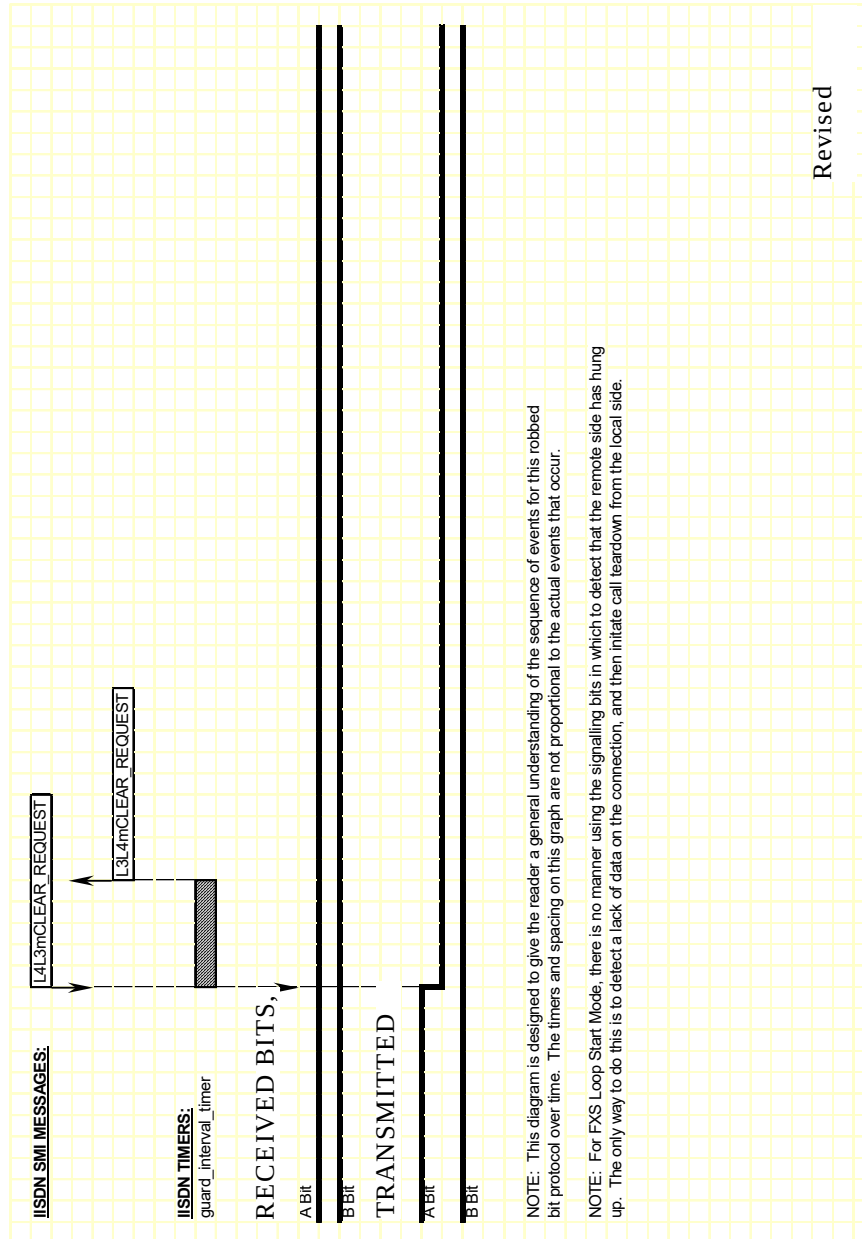
Revised

Figure 44. FXS Loop Start – Incoming Call Timing Diagram



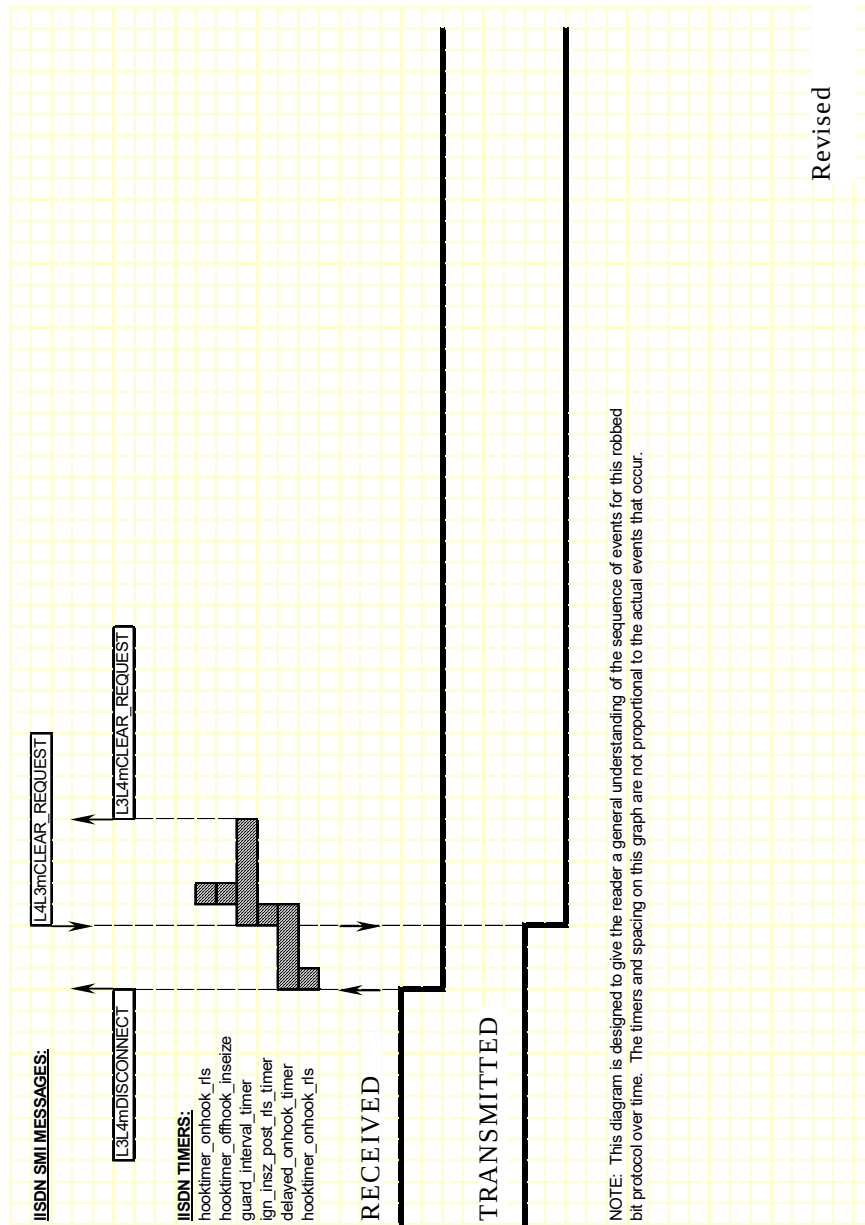
Revised

Figure 45. FXS Loop Start – Outgoing Call Timing Diagram



Revised

Figure 46. FXS Loop Start – Outgoing Clear Timing Diagram



Revised

Figure 47. Non-FXO/FXS RBS Protocols - Incoming Clear Timing Diagram

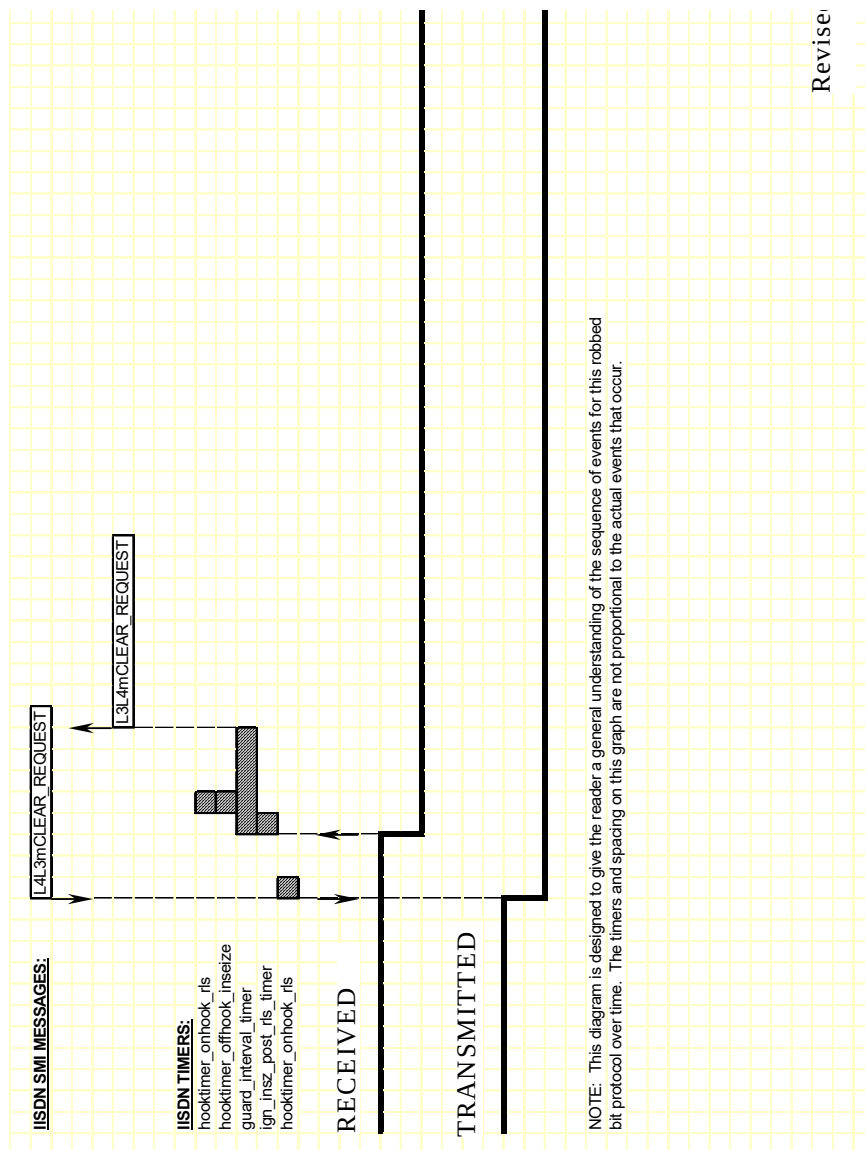


Figure 48. Non-FXO/FXS RBS Protocols - Outgoing Clear Timing Diagram

8 - ISDN Call Processing and Management

This chapter describes ISDN call processing using BSMI-level call control.

The chapter has the following sections:

- [*ISDN Call Processing Overview on page 318*](#)
- [*Translating Q.931 to Simple Message Interface on page 327*](#)
- [*Using the overlap_rcv feature of L4L3mENABLE_PROTOCOL on page 328*](#)
- [*Q.921/Q.931 Timers on page 332*](#)

Modules provide multipurpose platforms for fully integrated network access. The nature of the functions performed by this interface are determined by the host application developed to control this interface.

Note: This chapter applies only to BSMI (low-level call control) users. Bfv API call control users do not need to be concerned with this information.

Note: QSIG and IP are not supported by BSMI.

C language code fragments have been included to illustrate key points in the text. It is recommended that the information in this section be reviewed in conjunction with the BSMI section of the *Dialogic® Brooktrout® Bfv API Reference Manual, Volume 5*.

BSMI interprets undecoded Q.931 packets from the network, removing information not needed by most applications and making it available via an L3L4m message. If you need specific Q.931 information not delivered with the message but that is documented as contained in a Q.931 packet, such as call setup messages which have two IEs (Information Elements) – Lower and Higher level compatibilities, you can get access to the entire undecoded packet via an L3L4mRAW_QDATA message (see the BSMI section of the *Dialogic® Brooktrout® Bfv API Reference Manual, Volume 5*, for details). Raw data for Q.931 packets that are not even processed by BSMI in the first place and for Q.931 packets containing user-defined ISDN messages is also notified via this message.

ISDN Call Processing Overview

This subsection presents an overview of ISDN incoming and outgoing call setup and tear down and overlapped dialing. Switched 56 calls use a similar mechanism.

Making an ISDN Incoming Call

A typical incoming ISDN call is illustrated in [Figure 49](#). In this call scenario, the module is answering a call. The following message exchange is relative to the module:

1. Call arrival is indicated by receipt of a SETUP message from the network containing information about the call (calling party and called party numbers, etc.).
2. The module generates two messages upon receipt of the SETUP.
 - ♦ CALL PROCEEDING message to the network indicating the call is being processed.
 - ♦ L3L4mSETUP_IND message to the host indicating receipt of the SETUP message from the network; information about the call (calling party and called party numbers, etc.) is included in this message if received from the network.
3. If the incoming call is to be accepted, the host optionally sends an L4L3mALERTING_REQUEST message to the module.
4. Upon receipt of the L4L3mALERTING_REQUEST message, the module sends an ALERTING message to the network.
5. The host then sends an L4L3mCONNECT_REQUEST to the module, asking that the call be connected; the host can consider the call connected at this point unless an error indication is received.
6. Upon receipt of the L4L3mCONNECT_REQUEST message, the module sends a CONNECT message to the network.
7. The network immediately responds with a CONNECT ACKNOWLEDGE message and the call is connected.

This scenario does not use B-channel negotiation, which would require additional message exchanges between the module and host. B-channel negotiation is a feature that allows you to specify on which B-channel you want an incoming call to be established.

Notification of incoming calls comes via a SETUP_IND message which contains the B-channel number the network would like to set the call up on. By turning on negotiation, you can specify a different (specific) B-channel on which to establish rather than the one requested by the network. In this case, a CALL PROCEEDING message will not get sent to the network and it is your responsibility to notify the network of the desired B-channel via any one of the following three messages: L4L3mCALL_PROCEEDING_REQUEST, L4L3mALERTING_REQUEST, and L4L3mCONNECT_REQUEST.

Note: If using more than one of these messages, you should be consistent on the B-channel value.

You might want to do B-channel negotiation in situations where each of the channels is hard-code mapped to different locations depending on the type of call (data, voice, fax) received. If an incoming call to a channel is not the type of call that is handled by that (network-assigned) channel, then it must be (re)established on a channel that can handle the type of that call, or else the call won't be handled properly.

This feature is turned on by setting:

```
l43.data.enable_protocol.level3.cnfg.q931.b_chan_negot=1
```

in the ENABLE_PROTOCOL message. In the case where the network indicates that the B-channel it is offering is non-negotiable, and the **l43.data.enable_protocol.level3.cnfg.q931.proc_on_exclusv** field was set when the protocol was enabled, the calling process will proceed with a CALL PROCEEDING message being sent to the network without requiring you to indicate one. The network indicates that the B-channel it is offering is non-negotiable. The user had agreed to proceed by having set the proc_on_exclusv field.

Connect Acknowledge is reported to the module but is not automatically reported to the host via the Bfv API. In order to retrieve L3L4mCONN_ACK_IND, set the field below to 1 in your ENABLE_PROTOCOL message:

```
l43.data.enable_protocol.level3.cnfg.q931.subscribe_connack=1
```

The stack can consider the call connected after sending the L4L3mCONNECT_REQUEST message or wait until it receives the CONNECT ACKNOWLEDGE from the network. If you do not want to wait for the network response, set the field below to 1 in your ENABLE_PROTOCOL message:

```
l43.data.enable_protocol.level3.cnfg.q931.no_rx_conn_ack=1
```

It is also possible to configure BSMI to generate a SETUP_ACK rather than a CALL_PROCEEDING message upon arrival of a setup message.

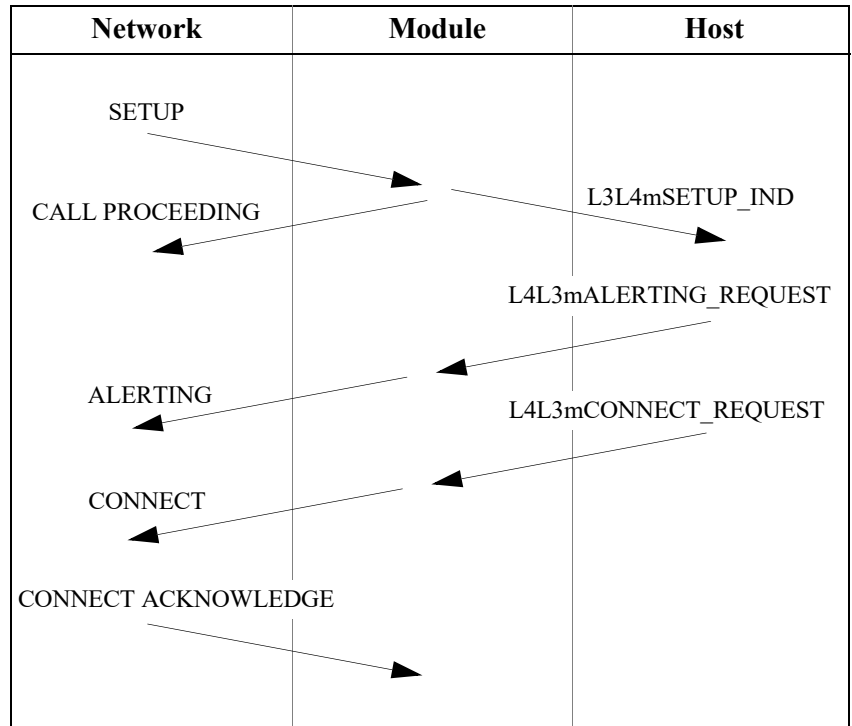


Figure 49. ISDN Incoming Call

As in the case of a call, BSMI also handles the call model where a SETUP_ACK generated instead of a call proceeding.

Making an ISDN Outgoing Call

A typical outgoing ISDN call is illustrated in [Figure 50](#). In this call scenario, the module is making an outgoing call. During an outgoing call, if the phone number is more than 20 digits long, the application automatically sends the number using overlapped dialing (See [ISDN Overlapped Dialing](#) on [page 323](#) for more information). The following message exchange is relative to the module:

1. The host initiates an outgoing call by sending an L4L3mCALL_REQUEST message to the module; this message must contain information about the call (calling party and called party numbers, and so on).
2. Upon receipt of the L4L3mCALL_REQUEST message, the module issues a SETUP message to the network; this message must contain information about the call (calling party and called party numbers, and so on).
3. The network responds to the SETUP with a CALL PROCEEDING message indicating the call request is being processed.
4. Following the CALL PROCEEDING, the network sends an ALERTING message indicating the called party is ringing.
5. Upon receipt of the ALERTING message, the module generates an L3L4mALERTING message to the host.
6. The network then sends a CONNECT message to the module indicating it is connecting the call.
7. The module generates two messages upon receipt of the CONNECT.
 - ♦ L3L4mCONNECT message to the host; the host can consider the call connected at this point
 - ♦ CONNECT ACKNOWLEDGE message to the network indicating the CONNECT message has been acknowledged

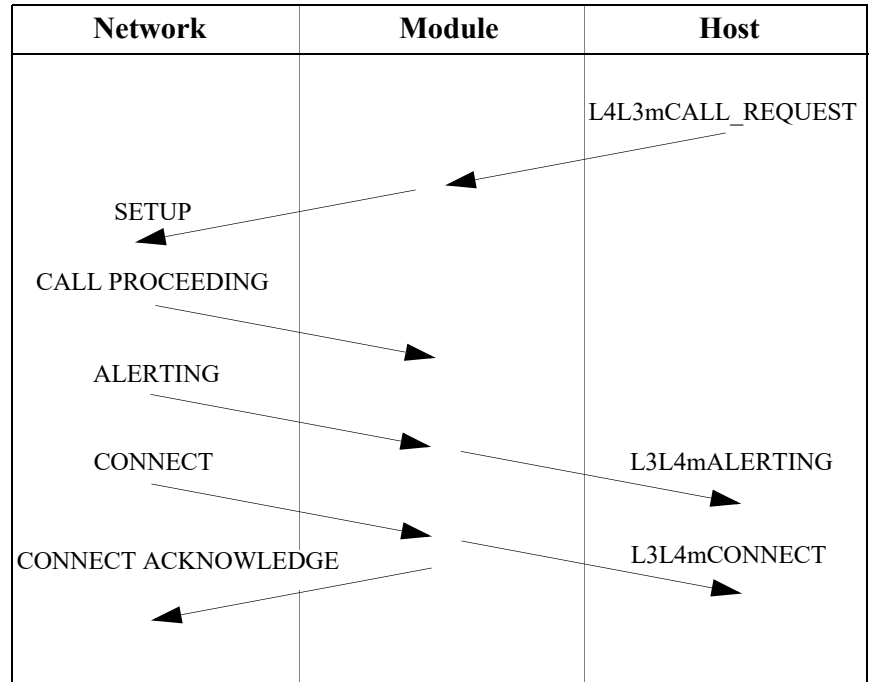


Figure 50. ISDN Outgoing Call

Call Proceeding (L3L4mCALL_PROCEEDING) is reported to the module but is not automatically reported to the host via the Bfv API. In order to retrieve this message, set the field below to 1 in your ENABLE_PROTOCOL message:

```
143.data.enable_protocol.level3.cnfg.q931.report_incoming_callproc=1
```

The default is that the stack sends a CONNECT ACKNOWLEDGE upon receiving a CONNECT. If you do not want to automatically respond in this manner, set the field below to 1 in the ENABLE_PROTOCOL message:

```
143.data.enable_protocol.level3.cnfg.q931.no_tx_conn_ack = 1
```

You can specify that, if a desired B-Channel to establish a call on is unavailable, another available B-Channel, which is chosen by the module, will automatically be used. To do this, set `143.data.call_req_data.preferred = 1` to enable this feature in the CALL_REQUEST message.

ISDN Overlapped Dialing

The Euro-ISDN protocol only allows 20 digits to be sent as a group (en-block) when placing a call. For longer phone numbers, the application must use overlapped dialing— the process of sending extra digits after the initial call setup. This process allows:

- The application to dial very large phone numbers
- The remote end to start answering a call before it receives all the digits

The Bfv API automatically performs overlapped dialing on Euro-ISDN outgoing calls using E1 or BRI lines when the phone number is longer than 20 digits. Users will be able to place a call with up to 255 digits in the dial string on Euro-ISDN without requiring changes in the application.

Applications automatically send overlapped digits when the user calls *BfvCallSetup*, *BfvLineDialString* or *BfvLineOriginateCall* and provides a dial string of more than 20 digits when Euro-ISDN is being used.

The feature divides the dial string into banks of 20 digits and uses the overlapped dialing feature in Euro-ISDN to send one block of digits at a time.

ISDN Call Clearing - Initiated by Module

A typical module-initiated call clearing (tear down) sequence is illustrated in [Figure 51](#). The following message exchange is relative to the module:

1. The host initiates the call clearing procedure by sending an L4L3mCLEAR_REQUEST message to the module.
2. Upon receipt of the L4L3mCLEAR_REQUEST message, the module sends a DISCONNECT message to the network.

3. The network responds to the DISCONNECT with a RELEASE message.
4. The module generates two messages upon receipt of the RELEASE:
 - ♦ RELEASE COMPLETE message to the network, indicating the call has been cleared
 - ♦ L3L4mCLEAR_REQUEST message to the host, indicating the call has been disconnected

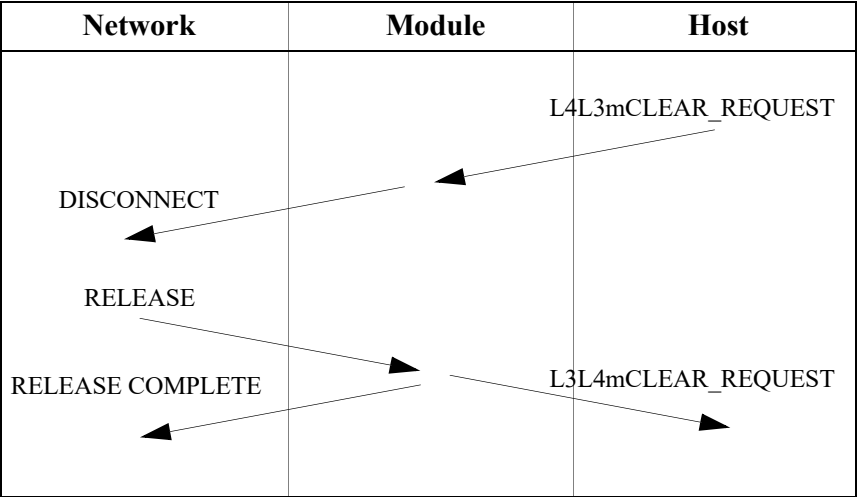


Figure 51. ISDN Call Clearing - Initiated by Module

If you want manual control over the sending of RELEASE COMPLETE, set the field below to 1:

```
l43.data.enable_protocol.level3.cnfg.q931.release_complete_control = 1
```

An L4L3mCLEAR_REQUEST message will then send the RELEASE COMPLETE message.

The host cannot consider a call disconnected and the channel available for another call until the L3L4mCLEAR_REQUEST message has been received from the module.

If the call is made using Switched 56 robbed bit signaling (not ISDN), there is an additional waiting period between sending the L4L3mCLEAR_REQUEST and receiving the L3L4mCLEAR_REQUEST indicating the channel is available. This interval is equal to the value of the *guard_interval_timer* and is usually 100 ms. Refer to [Robbed Bit Signaling](#) on [page 266](#) for more information on robbed bit signaling timers.

ISDN Call Clearing - Initiated by Network

A typical ISDN call clearing is illustrated in [Figure 52](#). In this call scenario, the module responds to the network's request to disconnect a call. The following message exchange is relative to the module:

1. The network initiates the call clearing procedure by sending a DISCONNECT message to the module.
2. Upon receipt of the DISCONNECT, the module generates an L3L4mDISCONNECT message to the host indicating the network is clearing the call.
3. The host responds with an L4L3mCLEAR_REQUEST message indicating the call should be cleared.
4. Upon receipt of the L4L3mCLEAR_REQUEST, the module generates a RELEASE message to the network.
5. The network responds to the received RELEASE by generating a RELEASE COMPLETE message.
6. Upon receipt of the RELEASE COMPLETE message, the module informs the host the call has been cleared using the L3L4mCLEAR_REQUEST message.

It should be noted that the host cannot consider a call disconnected and the channel available for another call until the L3L4mCLEAR_REQUEST message has been received from the module.

If the call is made using Switched 56 robbed bit signaling (not ISDN), there is an additional waiting period between sending the L4L3mCLEAR_REQUEST and receiving the L3L4mCLEAR_REQUEST indicating the channel is available. This interval is equal to the value of the *guard_interval_timer* and is usually 100 ms. Refer to [Robbed Bit Signaling](#) on [page 266](#) for more information on robbed bit signaling timers.

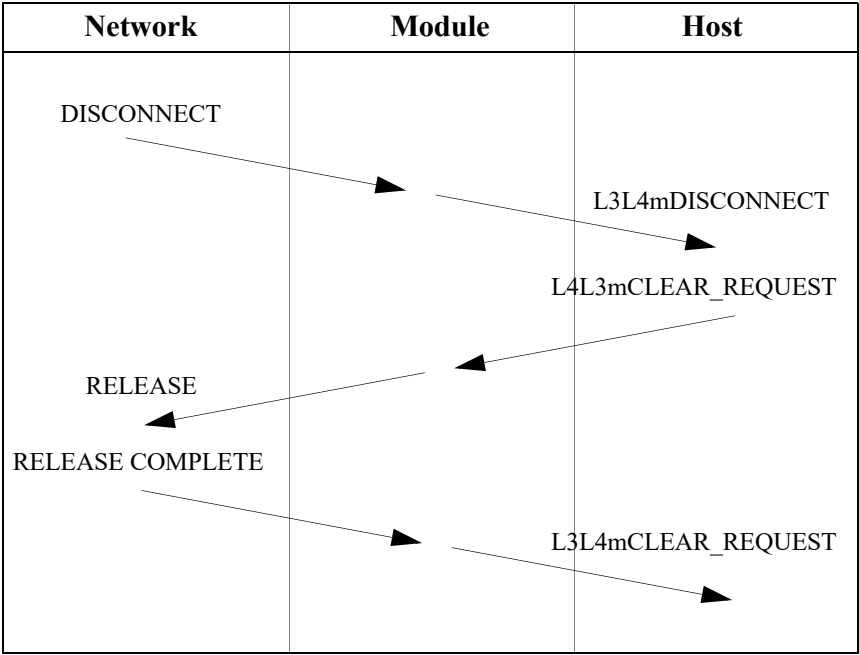


Figure 52. ISDN Call Clearing - Initiated by Network

Translating Q.931 to Simple Message Interface

Table 14 translates Q.931 messages to the appropriate Simple Message Interface (SMI) message and compares them to their corresponding message.

Table 14. Q.931 Message Comparison Table

Q.931Message	BSMI Message(L4L3m)	BSMI Message (L3L4m)
ALERTing	ALERTING_REQUEST	ALERTING
CALL PROCEEDing	CALL_PROCEEDING_REQUEST	CALL_PROCEEDING
CONNect	CONNECT_REQUEST	CONNECT
CONNect ACKnowledge	nothing	CONN_ACK_IND
PROGress	PROGRESS_REQUEST	PROGRESS
SETUP	CALL_REQUEST	SETUP_IND
SETUP ACKnowledge	SETUP_ACK_REQUEST	SETUP_ACK
RESume	RESUME_REQUEST	RESUME_REQUEST
RESume REJect	RESUME_REJECT	RESUME_REJECT
SUSPend	SUSPEND_REQUEST	SUSPEND_REQUEST
SUSPend ACKnowledge	SUSPEND_ACK	SUSPEND_ACK
SUSPend REJect	SUSPEND_REJECT	SUSPEND_REJECT
USER INFOmation	USER_INFO	USER_INFO
DISConnect	CLEAR_REQUEST	DISCONNECT
RELease	CLEAR_REQUEST	CLEAR_REQUEST
RELease COMPLETE	nothing	CLEAR_REQUEST Note: If <code>l43.data.enable_protocol.level3.cnf.g.q931.release_complete_control</code> is set, CLEAR_REQUEST will send this message.
REStart	RESTART	CLEAR_WITH_RESTART_REQUEST or RESTART

Table 14. Q.931 Message Comparison Table (Continued)

Q.931Message	BSMI Message(L4L3m)	BSMI Message (L3L4m)
REStart ACKnowledge	nothing	nothing
SEGMENT	nothing	nothing
CONGestion CONtrol	nothing	nothing
INFOmation	INFO_REQUEST	INFO
NOTIFY	nothing	UNIVERSAL
STATus	nothing	STATUS_IND
STATus ENQuiry	nothing	nothing
FACILITY	UNIVERSAL	UNIVERSAL

Using the *overlap_rcv* feature of L4L3mENABLE_PROTOCOL

What is Overlap Receive?

When an ISDN call is received from the Public Telephone Network, the caller information arrives in a Q.931 SETUP message. Typically, this SETUP message contains all the information necessary for the user to properly route, switch and/or complete the call: information about the type of call (voice, modem, data, and so on), caller ID information, and so on.

Overlap Receive mode occurs when the incoming SETUP message contains no CALLED party information, or incomplete CALLED party information. The CALLED party information is necessary to complete a circuit-switched call, because it identifies the complete number that the originator is dialing.

The ITU Q.931 Recommendation defines a SENDING_COMPLETE Information Element as an indicator of when a SETUP message has complete CALLED party number information. If a SETUP message is received and there is no SENDING_COMPLETE info element in the message, the user is expected to enter Overlap Receive mode.

BSMI Reference Notes

Support for Overlap Receive mode are enabled using the *overlap_rcv* flag in the BSMI_Q931_CNFG structure of L4L3mENABLE_PROTOCOL messages.

If the host application expects to support Overlap Receive calling models (more typical of E1 and EURO-based BRI installations than domestic T1 installations), it should set the *overlap_rcv* flag to “1” when enabling the Q.931 D-channel. Example:

```
L4_to_L3_struct  l43msg;

memset (&l43msg, 0, sizeof(l43msg));
l43msg.lapdid = 0;
l43msg.msgtype = L4L3mENABLE_PROTOCOL;
l43msg.data.enable_protocol.level1.l1_mode = IISDNl1modHDL;
l43msg.data.enable_protocol.level2.l2_mode = IISDNl2modLAP_D;
l43msg.data.enable_protocol.level2.dce_dte = IISDNdirUSER_SIDE;
l43msg.data.enable_protocol.level3.l3_mode = IISDNl3modQ931;
l43msg.data.enable_protocol.level3.cnfg.q931.switch_type =
    IISDNstUNKNOWN;
l43msg.data.enable_protocol.level3.cnfg.q931.variant = IISDNvarCCITT;
l43msg.data.enable_protocol.level3.cnfg.q931.overlap_rcv = 1;

BSMIControlWrite (fd, l43msg);
```

How Overlap Receive Mode Changes Call Control Events Presentation

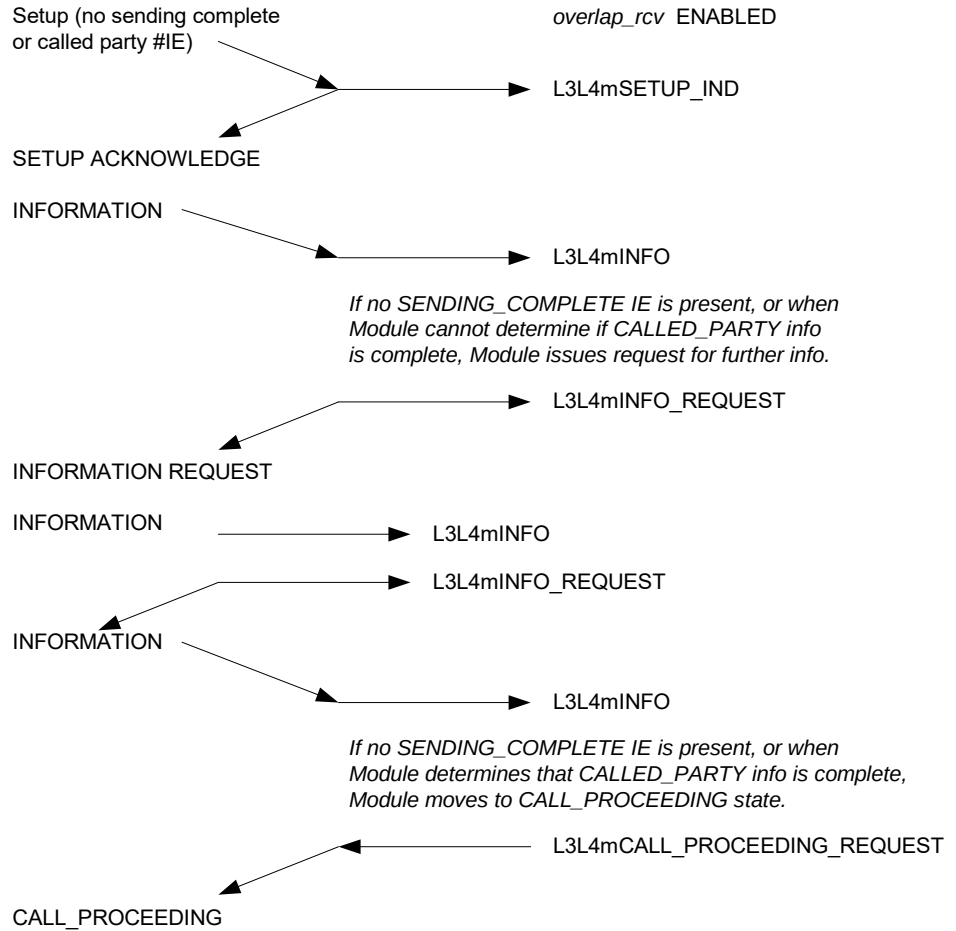
Volume 5, Dialogic® Brooktrout® Bfv API Reference Manual describes the normal call control events associated with ISDN Call Processing and Management. These events occur when incoming SETUP messages contain all the appropriate CALLED_PARTY information elements and SENDING_COMPLETE information elements.

When the *overlap_rcv* flag is enabled and the SETUP message arrives with incomplete CALLED_PARTY information, the call control events change as follows:

1. Network presents SETUP message to the module, with either incomplete CALLED_PARTY information element present, or no SENDING_COMPLETE info element present.
2. Instant ISDN activates the T302 timer, sends a SETUP_ACKNOWLEDGE message to the Network, and enters the Overlap Receive state. Instant ISDN also presents the initial call information to the host in an L3L4mSETUP_IND message.

Note: The T302 timer is under host control (see IISDN_Q931_TIMERS in the IISDN.H header file) with a default timer value of 15 seconds.

3. The Network sends the remainder of the call information (if any) in one or more INFORMATION messages. Each information message arrives as an L3L4mINFO event to the host application, which contains CALLED_PARTY information elements, as well as an indicator for SENDING_COMPLETE information elements. Each time an INFORMATION message is received without a SENDING_COMPLETE info element present in the message, Instant ISDN restarts the T302 timer.
4. Presence of the SENDING_COMPLETE information element is optional. Some ISDN switches and dialing plans will present all of the CALLED_PARTY information and never send a SENDING_COMPLETE info element in the final INFORMATION or SETUP message.
5. When the final INFORMATION message has been received (indicated by presence of the SENDING_COMPLETE information element, or when the host has determined that the CALLED_PARTY information is sufficient to complete the call), the host application issues an L4L3mCALL_PROCEEDING and the call control events will then transpire normally.



Q.921/Q.931 Timers

An application can change the default behavior in Instant ISDN Software Level 2 and Level 3 parameters by issuing ENABLE_PROTOCOL.

L4L3mENABLE_PROTOCOL contains an IISDN_L2_CONST structure that defines Level 2 parameters and an IISDN_Q931_TIMERS structure that configure Level 3 timers. The IISDN_Q931_TIMERS structure is only valid when the Level 3 mode (l3_mode) value in the message is IISDNl3modQ931. [Table 15](#) and [Table 16](#) list the parameters for both structures.

When altering these values, refer to the system specification for the switching system to which you are connecting the Brooktrout Controller for valid timer and window values. Level 2 timers are specified in 1 millisecond “ticks;” Level 3 timers in 100 millisecond “ticks.”

Note: Dialogic strongly recommends that you use the default values for Level 2 and Level 3 parameters and internal buffers. Failure to do so might result in a system irregularity. If you do need to change the configurations and buffer sizes, call Dialogic Technical Services and Support first and seek consultation. Internal buffer sizes for different Brooktrout Controller configurations are found in IISDN.h.

Table 15. Q.921 Timers (Level 2 Parameters)

Data Type	Mnemonic	Definition
IISDN_L2_LAP_CONSTS	12	Setting a field to 0 causes the default value to be used. Some timers have two possible default values depending on whether or not the line is a BRI (set by the q931_cfg.basic_rate flag).
unsigned short	t200	Maximum retransmission time. Default value is 1000 (1 second).
unsigned short	t201	Minimum retransmission TEI Identity check. Default value is 1000 (1 second).
unsigned short	t202	Minimum TEI Identity request. Default value is 2000 (2 seconds).
unsigned short	t203	Maximum link inactivity. Default value is 30 seconds. For NET-5 signaling, this timer should be set to 10 seconds. BRI default is 10000 (10 seconds).

Table 15. Q.921 Timers (Level 2 Parameters) (Continued)

Data Type	Mnemonic	Definition
unsigned short	n200	Maximum number of retransmissions. Default value is 3.
unsigned short	n201	Maximum number of octets in an I frame. Currently the maximum is 240.
unsigned short	n202	Maximum number of transmissions of a TEI request message. Default value is 3.
unsigned short	K	Maximum transmit window. Default value is 7; maximum value is 127. BRI default is 1.

Table 16. Q.931 Timers (Level 3 Parameters)

Data Type	Mnemonic	Definition
IISDN_Q931_TIMERS	q931	Structure that includes changes to the Q.931 timer values, if any. A value of 0xFFFF disables the timer. Setting a field to 0 causes the default to be used. Some timers have two possible default values depending on the connection type/variant used. For example, if the variant field value is IISDNvarCCITT in an L4L3mENABLE_PROTOCOL message, or the conn_type field contains an IISDNctCCITT value in an L4L3mENABLE_D_CHANNEL message, the CCITT default value below is used. For all other connection type or variant values, the standard default value is used.
unsigned short	t302	Q.931 Overlap Receiving timer. Default value is 150 (15 seconds).
unsigned short	t305	Q.931 Disconnect Request state timer. Default value is 50 (5 seconds) and the ITU-T default value is 300 (30 seconds).
unsigned short	t308	Q.931 Release Request state timer. Standard default value is 50 (5 seconds) and the ITU-T default value is 40 (40 seconds).
unsigned short	t313	Q.931 Connect Request state timer. Default value is 50 (5 seconds) and the ITU-T default value is 40 (4 seconds).
unsigned short	t314	Q.931 Segment message time. Default value is 40 (4 seconds).
unsigned short	t316	Q.931 Restart Request state timer. Default value is 1200 (120 seconds).

Table 16. Q.931 Timers (Level 3 Parameters) (Continued)

Data Type	Mnemonic	Definition
unsigned short	t318	Q.931 Resume Request state timer. Default value is 1200 (120 seconds).
unsigned short	t319	Q.931 Suspend Request state timer. Default value is 1200 (120 seconds).
unsigned short	t3m1	Q.931 Maintenance SERVICE ACK timer. Default value is 1200 (120 seconds). Default value <i>must</i> be used for NFAS configurations.
unsigned short	t321	Q.931 NFAS D-channel backup timer. Default value is 400 (40 seconds).

If layer 2 has been enabled, you will receive the data (termed payload) you expect with the layer 2 headers stripped off and all the data reliability checks already performed. If layer 2 has not been enabled, you will receive intact layer 2 packets, which will contain the payload. These packets will need to be processed, and a layer 2 stack will have to be implemented all the while extracting the payload.

HDLC packetization does not need to be enabled, instead you can choose the raw mode option. To do this, both HDLC headers will need to be either stripped off or recognized, and the layer 2 packet headers inside the HDLC headers to get to the payload.

9 - Using the BSMI R2 Signaling Capability

This chapter describes R2 signaling as used with BSMI-level call control.

The chapter has the following sections:

- [CPE Signaling Model on page 336](#)
- [Enabling the R2 Protocol on page 341](#)

Brooktrout boards offer E1 CAS signaling for customer premise equipment (CPE).

The protocols supported are:

- ITU's R2 (Line Signaling and Inter-register Signaling) and national variants
- LEC protocols (Wink Start, Delay Dial, Immediate Start, FXO Loop Start, FXS Loop Start, FXO Ground Start, FXS Ground Start)

Dialogic's implementation of the R2 protocol contains a number of parameters that are used to configure the module to conform to the ITU recommendations (refer to ITU Blue Book, Signaling System R2, Volume VI Fascicle VI.4, Q.421 - Q.424, digital line signaling, and Q.440 - Q.458, compelled inter-register signaling). The fully embedded R2 protocol stack consists of digital line signaling (R2

Line Signaling) and MF compelled (MFC) (R2 Inter-register Signaling). Signaling is controlled via the BSMI interface in a manner similar to Q.931 and LEC protocols.

Note: This chapter applies only to BSMI (low-level call control) users. Bfv API call control users do not need to be concerned with this information.

The LEC protocols are described in [Robbed Bit Signaling on page 266](#).

Please contact Dialogic Technical Services and Support for a list of supported variants of the R2 protocol. See the *Dialogic® Brooktrout® Bfv API Reference Manual, Volume 5* for details.

CPE Signaling Model

This section describes the general signaling model that is implemented by the Brooktrout R2 stack. This model is valid for a large number of national variants that are found worldwide. Customization via the BSMI will not be possible for variants that are not described by this model.

The line signaling part of the protocol is illustrated in [Table 17](#). The Brooktrout implementation assumes that trunks are configured for “both-way” working, that is outbound and inbound calls are possible on the same trunk. Outbound-only and inbound-only trunk configuration is not currently supported. In the discussion to follow, “forward” is synonymous with outbound and “backward” is synonymous with inbound.

Table 17. Line Signaling Model

State	CAS Bits		Notes
	Outbound AB	Inbound AB	
Idle	10	10	Both outbound and inbound channels are sending idle signal.
Seize	00	10	Outbound channel seizes the line.
Seize Acknowledge	00	11	Inbound channel initializes MFC inter-register signaling and sends the seize acknowledgment. When the seize acknowledgment is recognized by the outbound channel, inter-register signaling begins with the transmission of the first DNIS digit.
If the call is accepted by the inbound side via the inter-register signaling, the answered state is entered.			
Answer	00	01	
Answer	00	11	Metering pulses are possible. In this example the pulse is transmitted on the A bit. The outbound side must not confuse this with a clear back signal.
Answer	00	01	Metering pulse over, inbound resumes transmission of the answer signal.
If the call is rejected by the inbound side via the inter-register signaling, the outbound side is responsible for clearing the call.			
Clear forward	10	11	Outbound side sends clear forward.
Idle	10	10	Inbound side sends the idle signal.
Disconnection initiated by the inbound side.			
Clear back	00	11 or 00	The hang-up might be signaled by a clear back (AB=11) or by a forced release (AB=00), as indicated by network specifications.
Clear forward	10	11 or 00	Outbound side sends the clear forward signal.

Table 17. Line Signaling Model (Continued)

State	CAS Bits		Notes
	Outbound AB	Inbound AB	
Idle	10	10	Inbound side sends the idle signal.
Disconnection initiated by the outbound side.			
Clear forward	10	01	Outbound side sends the clear forward signal.
Release guard (Optional; refer to network specifications)	10	11	This state is optional. The inbound side responds to the clear forward with a clear back signal and holds it for a specified length of time.
Idle	10	10	Inbound side sends the idle signal.
Channel blocking is implemented according to ITU recommendations for “both-way” trunks. The side performing the blocking is considered the inbound or backward channel.			
Idle	10	10	Both ends of the channel are idle.
Blocked	10	11	Blocking signal is transmitted.
Idle	10	10	Blocking signal is removed and both ends are in the idle state.

The inter-register signaling performs the exchange of address information via R2 MF tones. The split-band forward and backward tone sets allow the definition of sets of forward and backward protocol signals. The exchange is performed in a compelled manner with the outbound channel sending a forward address signal that is acknowledged by a backward signal that itself directs the transmission of the next piece of address information. Information passed from the outbound side to the inbound side is:

- DNIS digits (Dialed Number Identification Service) representing the called party
- ANI digits (Automatic Number Identification) representing the calling party
- DNIS category (a.k.a. Call, Group II, or Toll category)
- ANI category (a.k.a. Caller, User, or Group I category)
- Circuit type (terrestrial or satellite)
- Half echo-suppressor to be used or not used

Information passed from the inbound side to the outbound side is:

- Called line condition (for example, free with change, busy, etc.)
- Network congestion

ITU recommendations specify a set of 6 frequencies to be used for forward signals, and another 6 for backward signals. Each signal consists of 2 frequencies, thus providing 15 forward signals and 15 backward signals. The recommendations allow countries to use only 5 frequencies (10 signals) for forward signaling and 4 frequencies (6 signals) for backward signaling. Both the forward and backward signals can assume a number of meanings depending on the signaling state. Transition from one set of meanings (called a 'group') to another is controlled by transmission of specific signals. ITU recommendations define 2 groups for forward signals (groups I and II) and 2 for backward signals (groups A and B). Some countries chose to define 3 groups in each direction (I, II and III for forward signals and A, B and C for backward signals). The concept of 'groups' allows more information to be conveyed (2 groups containing up to 15 signals results in 30 signals in each direction) without having to increase the number of signal generators and detectors, which are expensive resources.

The forward channel initiates inter-register signaling with the transmission of the first DNIS digit. Subsequent forward transmission is controlled by the last backward channel signal response.

1. DNIS transmission processing
2. ANI transmission processing
3. Called line condition processing

The backward channel controls inter-register signaling in the sense that after the first DNIS digit transmission, each subsequent forward channel signal is a response to the last backward channel signal. Following the first DNIS digit, the backward channel directs the collection of the address information according to the inbound application parameters associated with the call. The backward protocol is abstracted into four processing states that correspond to the four blocks of information that are sent by the forward side:

1. DNIS number collection
2. DNIS category
3. ANI number collection
4. ANI category

Figure 53 illustrates an example inter-register exchange in which the inbound protocol is configured to collect DNIS digits first, followed by collection of ANI information, and concluding with the exchange of DNIS category and called line condition. However, variations on this exchange are possible, resulting from different settings of the inbound protocol control parameters and the absence of certain signals in a given variant. For instance, the inbound protocol is configured to request ANI information following reception of N DNIS digits. Following completion of ANI collection, the DNIS collection is resumed. Another example is that a certain variant might not define the forward signal that indicates the end of the DNIS string. In this case, the inbound protocol must count the DNIS digits and signal a request for other information when the required number of DNIS digits has been collected.

Enabling the R2 Protocol on page 341 provides further detail on the protocol control parameters, protocol signals, and protocol processing actions provided by the Brooktrout R2 stack.

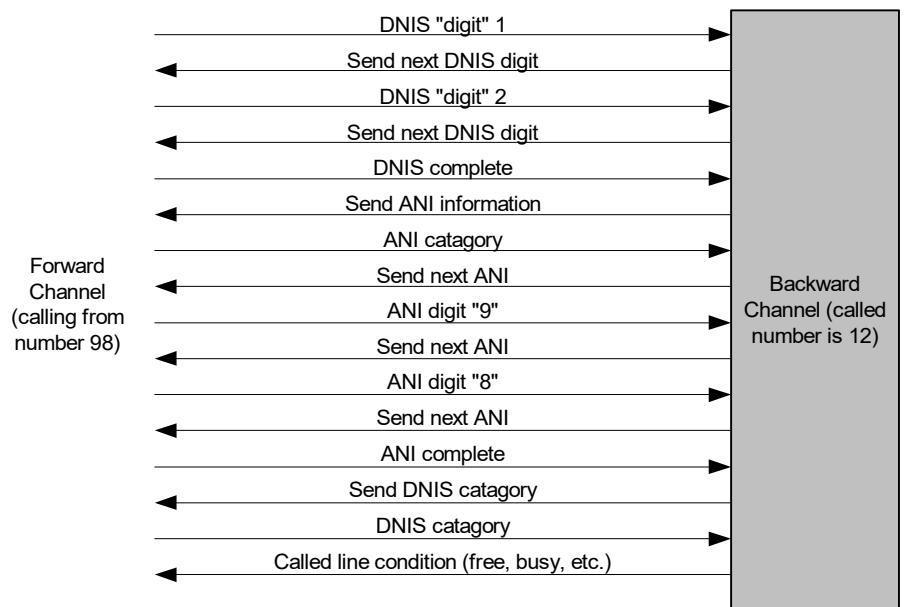


Figure 53. Inter-register Signaling

Enabling the R2 Protocol

Using the R2 protocol is similar to using the LEC protocols. Each individual channel is enabled using an L4L3mENABLE_CAS message. When the module receives this message, it will initialize all timers and data structures associated with the specified channel, put the line in idle state “onhook” and respond with an L3L4mCAS_STATUS message.

The application must select the following signaling type:

```
l43msg.data.cas_data.signalling_type = IISDNsigtypeR2_CAS;
```

This mode operates the inter-register protocol stack on the DSP processors. The iisdn.h file also identifies CAS signaling type IISDNsigtypeR2_MF. This is an IISDN provision for operating R2 signaling on certain controller modules that incorporate modem chips that provide basic MF generation and detection services to IISDN. This capability is not currently released.

For all types of CAS signaling, the lapdid value (zero-based) in the common message header indicates the network interface (line) on the module receiving the message. The call reference value (field call_ref) is a 16-bit value that must have the lapdid value in the most significant 8 bits and the B-channel in the least significant 8 bits (if no B-channel needs to be specified, use 0 for the least significant byte). Since not all BSMI messages involve a B-channel, there is no field in the common message header to specify the B-channel. For some messages the B-channel is specified in the message-specific data structure, and for others the B-channel is inferred from the call reference value. Line and B-channel numbering is 0-based, and the B-channel numbering is line-specific (that is, the first B-channel on the second line is numbered 0).

All of the necessary trunk customization parameters are contained in the IISDN_E1_CAS_R2_DATA portion of the L4L3mENABLE_CAS structure. It is important that all parameters be filled. Most of the parameters are network specific and cannot be modified without resulting in protocol failure. Others are modifiable according to the requirements of the user's application. Dialogic supplies C header files that specify the parameter settings for certain national variants.

BSMI does not support varying any parameter settings from channel to channel on a single trunk. Trunk level configuration is performed upon receipt of the first L4L3mENABLE_CAS message on that trunk. However, L4L3mENABLE_CAS must be sent for every channel in order for the channel to initialize and go on-hook. Therefore, the IISDN_E1_CAS_R2_DATA structure should be filled out identically for each channel's L4L3mENABLE_CAS.

The IISDN_E1_CAS_R2_DATA structure itself contains two structures. [Table 18](#) identifies the IISDN_R2_DIGITAL_LINE_SIG_PARAMS sub-structure that contains the parameters required for configuring the E1 CAS line protocol. None of these parameters is modifiable once correctly specified for a particular network.

Table 18. R2 Digital Line Signaling Parameters

Name	Description	Units	Range
<i>r2OutSeizeTimer</i>	Time outbound side waits for seize acknowledgement signal.	ms	Refer to network specification, however, this is typically set to 100-200 ms for a terrestrial circuit, or 1-2 seconds for a satellite circuit, plus approximately 32 ms to account for internal detection latency.
<i>r2OutAnswerTimeOut</i>	Maximum time between reception of MF call acceptance signal and inbound answer.	ms	Refer to network specification.
<i>inboundReleaseGuardTime</i>	Certain networks require the inbound side to hold a “release guard” state (clear-back) for a certain amount of time following detection of the outbound clear-forward signal.	ms	0: disable release guard >0: refer to network specification
<i>inboundLineQualTimerIdle</i>	Inbound line signaling event qualification time during the idle state.	ms	0: qualify bits immediately >0: refer to network specification

Table 18. R2 Digital Line Signaling Parameters (Continued)

Name	Description	Units	Range
<i>c_d_cas_bits</i>	Specification of the CAS C bit and D bit settings.		0: CD=00 1: CD=01 2: CD=10 3: CD=11
<i>ClearbackControl</i>	Used by the outbound side to determine whether the network will use a release guard (AB=11) or a forced release (AB=00) signal to tear down a call. The selected signal is handled accordingly, while the other is simply ignored.		0: release guard 1: forced release

[Table 19](#) identifies the IISDN_R2_INTERREGISTER_PARAMS sub-structure containing the parameters required used to configure the MFC inter-register signaling.

Table 19. R2 MFC Inter-register Signaling Parameters

Name	Description	Units	Range
The Following Parameters Can Be Modified According to Application Requirements:			
<i>dnisMaxNumDigits</i>	Maximum number of DNIS digits required.		[0-IISDN_MAX_DIGITS]
<i>aniMaxNumDigits</i>	Maximum number of ANI digits required.		[0-IISDN_MAX_DIGITS]
<i>dnisNumDigitsBeforeANI</i>	The number of DNIS digits to collect before requesting ANI.		[1...N], where N = dnisMinNumDigits
<i>interForwardToneTimeOut</i>	Period associated with the inbound channel “T3” timer that supervises the interval elapsing between recognition of two consecutive forward tones. (Ref. ITU Q.476).	Seconds	Typically, set to [8-24], but refer to network specification.
<i>prePulseToneDelay</i>	The interval from the end of the last backward signal and the start of a backward pulse signal (Ref. ITU Q.442).	ms	Set to >= 100

Table 19. R2 MFC Inter-register Signaling Parameters (Continued)

Name	Description	Units	Range
<i>pulseToneDuration</i>	Duration of a backward pulse signal (Ref. ITU Q.442).	ms	Typically [100 - 200]
<i>Outbound MFC Timers</i>			
<i>forwardToneMaxOnTime</i>	Time associated with the outbound “T1” timer that supervises the interval between start of a forward tone and cessation of the forward tone (Ref. ITU Q.476).	Seconds	This is typically in the range [12-18], but refer to network specification.
<i>forwardToneMaxOffTime</i>	Time associated with the outbound “T2” timer that supervises the interval when no forward tones are sent. This interval consists of the period waiting for the backward tone to stop, as well as any additional time needed for the next forward tone to be known.	Seconds	Typically set to > 24 (Q.476), but refer to network specification.
<i>forwardGroup2MaxOnTime</i>	Not used.	Seconds	Not used.
Protocol Control			
<i>dnisTimeOutAction</i>	Not used.		Not used.
Protocol State Transition Specification Each of the following is an array of length 16. Element zero is invalid and elements 1-15 correspond to R2 MF signals 1-15. These allow the Call Setup Service Pack software to construct the proper state transition tables for a given trunk.			
<i>Forward Channel Signal Definitions</i>			
<i>endOfDNIS</i>	Signal end of DNIS pulsing; NOT defined in some variants.		Table 20
<i>endOfANI_Available</i>	Signal end of ANI identification when ANI digits are available.		Table 20
<i>InfoNotAvailable</i>	Indicates that ANI information is restricted. There is usually no need to distinguish from signal “aniRequestNotAccepted”.		Table 20
<i>aniCategoryDefault</i>	Default Group 1 category. Used if category is not specified in a given call request.		Table 20

Table 19. R2 MFC Inter-register Signaling Parameters (Continued)

Name	Description	Units	Range
<i>Backward Channel Signal Definitions</i>			
<i>sendNextDigitDNIS</i>	Signal request for next DNIS digit (state DNIS).		Table 20
<i>sendLastButOneDigitDNIS</i>	Signal request for last-but-one DNIS digit (state DNIS). This signal is not currently used.		Table 20
<i>sendCallCategoryAndSwitchToGroupB_DNIS</i>	Requests transmission of the Call Category and switch of the forward channel to reception of the Group B line condition (state DNIS).		Table 20
<i>congestion</i>	Signals congestion (state DNIS).		Table 20
<i>sendCallingPartyCategory</i>	Requests transmission of the calling party information starting with the Calling Category (state DNIS).		Table 20
<i>callComplete_SetUpSpeechPath</i>	Signals call acceptance, with charge, without the need for the Group II/Group B exchanges (state DNIS).		Table 20
<i>sendLastButTwoDigitDNIS</i>	Refer to ITU A-7. This signal is not currently used.		Table 20
<i>sendLastButThreeDigitDNIS</i>	Refer to ITU A-8. This signal is not currently used.		Table 20
<i>sendFirstDigitDNIS</i>	This signal is not currently used.		Table 20
<i>sendNextDigitANI</i>	Request transmission of the next ANI digit (state ANI).		Table 20
<i>changeFrom_ANI_To_DNIS_SendNextDigit</i>	Signal return to DNIS digit collection, requesting the next DNIS digit (state ANI).		Table 20
<i>changeFrom_ANI_To_DNIS_SendLastDigit</i>	Signal return to DNIS digit collection, requesting the last DNIS digit. This signal is not currently used. (State ANI).		Table 20

Table 19. R2 MFC Inter-register Signaling Parameters (Continued)

Name	Description	Units	Range
<i>groupB_LineConditions</i>	Structure of Group B called line conditions indexed by enumeration IISDN_R2MFCP_GROUP_B_CALLED_LINE_CONDITIONS. Only a subset of these signals are used in a given protocol variant. Set those that are not used to the “invalid” tone code.		Refer to Table 22
Call Progress Signal Generation			
<i>cpSignals</i>	Array of IISDN_CPGEN_MF_PARAMS structures that define the characteristics of the RING and BUSY call progress signals. RING must be defined as cpSignals[0] and BUSY must be defined as cpSignals[1].		Refer to Table 24

The MFC inter-register protocol is specified through a subset of the parameters in [Table 17](#) on [page 337](#) that define the following:

1. Signal meaning definitions (events).
2. Actions associated with a given signal event. These specifications are used to create a set of state transition tables for the desired variant.
3. Protocol control parameters.

The parameter set might require extensions as support is added for variants unidentified at this time. In addition, for a given variant, certain signals and actions might not be defined. This restricts the way in which the protocol can move through the processing states.

Protocol Parameter Mechanics

The set of protocol parameters is specified according to a simple procedure. Individual signal meanings are set to the appropriate MF tone code. R2 MF tone codes for both forward and backward channels are defined according to the enumeration in [Table 20](#). If a signal is not defined for a particular variant, its value is set to zero.

Table 20. IISDN_R2MF_SIGNAL_CODES Enumeration

Name Forward (Backward) Channel	Value
<i>IISDN_R2F_INVALID / IISDN_R2B_INVALID</i>	0
<i>IISDN_R2F_01 / IISDN_R2B_01</i>	1
<i>IISDN_R2F_02 / IISDN_R2B_02</i>	2
<i>IISDN_R2F_03 / IISDN_R2B_03</i>	3
<i>IISDN_R2F_04 / IISDN_R2B_04</i>	4
<i>IISDN_R2F_05 / IISDN_R2B_05</i>	5
<i>IISDN_R2F_06 / IISDN_R2B_06</i>	6
<i>IISDN_R2F_07 / IISDN_R2B_07</i>	7
<i>IISDN_R2F_08 / IISDN_R2B_08</i>	8
<i>IISDN_R2F_09 / IISDN_R2B_09</i>	9
<i>IISDN_R2F_10 / IISDN_R2B_10</i>	10
<i>IISDN_R2F_11 / IISDN_R2B_11</i>	11
<i>IISDN_R2F_12 / IISDN_R2B_12</i>	12
<i>IISDN_R2F_13 / IISDN_R2B_13</i>	13
<i>IISDN_R2F_14 / IISDN_R2B_14</i>	14
<i>IISDN_R2F_15 / IISDN_R2B_15</i>	15

Forward Channel

The set of forward channel actions to backward channel signals is defined in [Table 21](#).

Actions #2 - #8 might be used when processing DNIS or ANI digits according to a particular protocol variant. The processing changes state according to the current state and the particular event. For instance, when in the ANI state, if a backward signal event requires processing action `PROCESS_NEXT_DNIS_DIGIT_REQUEST`, then that action occurs with a return to the DNIS state. In certain protocol variants, a particular action might not be possible in a particular state. For instance, a particular China PRC variant uses `R2B_01` to signal both the request for the next ANI digit and the request for the next DNIS digit. Since there is no other, DNIS related signal defined, the protocol cannot return to DNIS collection until all ANI digits have been delivered.

Table 21. IISDN_R2MFC_FORWARD_ACTIONS

Name	Description	State
<i>PROCESS_INVALID_BACKWARD_SIGNAL</i>	Process a backward signal not defined; results in protocol termination.	Any
<i>PROCESS_NEXT_DNIS_DIGIT_REQUEST</i>	Process request for next DNIS digit; if invoked from the ANI state, then a state change to DNIS occurs.	DNIS or ANI
<i>PROCESS_LAST_BUT_1_DNIS_DIGIT_REQUEST</i>	Process request for N-1 DNIS digit.	DNIS or ANI
<i>PROCESS_LAST_BUT_2_DNIS_DIGIT_REQUEST</i>	Process request for N-2 DNIS digit.	DNIS or ANI
<i>PROCESS_LAST_BUT_3_DNIS_DIGIT_REQUEST</i>	Process request for N-3 DNIS digit	DNIS or ANI
<i>PROCESS_RESTART_DNIS_REQUEST</i>	Process request for DNIS restart.	DNIS or ANI
<i>PROCESS_CALL_COMPLETE_CHANGE_TO_GROUP_B</i>	Send Group II call category and change to Group B reception.	DNIS or ANI

Table 21. IISDN_R2MFC_FORWARD_ACTIONS (Continued)

Name	Description	State
<i>PROCESS_CONGESTION_SIGNAL</i>	Process congestion signal.	DNIS or ANI
<i>PROCESS_NEXT_ANI_DIGIT_REQUEST</i>	Process request for next ANI digit.	ANI
<i>PROCESS_CALLING_PARTY_CATEGORY_REQUEST</i>	Process request for calling party (ANI) information.	DNIS
<i>PROCESS_CALL_ACCEPTED_NO_GROUP_B</i>	Process call accepted without need for Group B line condition.	DNIS
<i>PROCESS_NATURE_OF_CIRCUIT_QUERY</i>	Process request for circuit nature.	DNIS
<i>PROCESS_ECHO_SUPPRESSOR_QUERY</i>	Process request for use of echo suppressor.	DNIS
<i>PROCESS_GROUP_B_LINE_CONDITION</i>	Process received (Group B) called line condition.	Line condition

Backward Channel

[Table 19](#) identifies the set of backward inter-register signals. The tones defined are those used to support DNIS and ANI services appropriate for customer premise equipment. Not all tones that are possible under the ITU recommendations are required nor implemented at this time. Certain tones might not be defined in a particular variant. In such a case, the values of those tones should be set to R2B_INVALID. In addition, certain signals are normally expected to have the same absolute signal code whether the forward channel is in state DNIS or state ANI. For example, this is true for the congestion signal. However, distinct definitions are provided in these cases to allow customization for an arbitrary variant.

Table 22 identifies the enumeration of Group B called line conditions. This enumeration forms the set of indices to array groupB_LineConditions (*Table 19*). This allows the mapping of the invariant enumeration values to and from the corresponding backward signal codes that vary from trunk to trunk. A number of user defined spares are provided.

Table 22. IISDN_R2MFCP_GROUP_B_CALLED_LINE_CONDITIONS Enumeration

Name	Description	Value
<i>IISDN_R2MFCP_LINE_FREE_CHARGE</i>	Line is free, charged.	0
<i>IISDN_R2MFCP_LINE_FREE_NO_CHARGE</i>	Line is free, no charge.	1
<i>IISDN_R2MFCP_LINE_ALTERNATE_ANSWER</i>	Line is free, alternate answer.	2
<i>IISDN_R2MFCP_LINE_BUSY</i>	Line busy.	3
<i>IISDN_R2MFCP_LINE_OUT_OF_ORDER</i>	Line is out of order.	4
<i>IISDN_R2MFCP_LINE_UNALLOCATED</i>	Line is unallocated.	5
<i>IISDN_R2MFCP_LINE_CONGESTION</i>	Circuit congestion.	6
<i>IISDN_R2MFCP_LINE_SPARE_CONDITION_1</i>	Spare	7
<i>IISDN_R2MFCP_LINE_SPARE_CONDITION_2</i>	Spare	8
<i>IISDN_R2MFCP_LINE_SPARE_CONDITION_3</i>	Spare	9
<i>IISDN_R2MFCP_LINE_SPARE_CONDITION_4</i>	Spare	10
<i>IISDN_R2MFCP_LINE_SPARE_CONDITION_5</i>	Spare	11
<i>IISDN_R2MFCP_LINE_SPARE_CONDITION_6</i>	Spare	12
<i>IISDN_R2MFCP_LINE_SPARE_CONDITION_7</i>	Spare	13
<i>IISDN_R2MFCP_LINE_SPARE_CONDITION_8</i>	Spare	14
<i>IISDN_R2MFCP_LINE_SPARE_CONDITION_9</i>	Spare	15

Table 23 identifies the set of backward channel actions to forward signals.

Table 23. IISDN_R2MFCP_BACKWARD_ACTIONS

Name	Description	Valid State
<i>PROCESS_INVALID_FORWARD_SIGNAL</i>	This action is performed when the protocol cannot recover and the register must immediately release. This action is used in any protocol state.	Any
<i>PROCESS_DNIS_DIGIT</i>	Process received DNIS digit.	DNIS
<i>PROCESS_DNIS_END_OF_PULSING</i>	Process end-of-pulsing (DNIS) signal.	DNIS
<i>PROCESS_CALL_CATEGORY_AND_SEND_LINE_STATE</i>	Saves the call category and requests the line state.	DNIS Category
<i>PROCESS_CALL_CATEGORY_AND_REJECT_CALL</i>	Processes an invalid call category, rejecting the call.	DNIS Category
<i>PROCESS_CALLING_CATEGORY_AND_DONT_REQUEST_ANI</i>	Processes a calling category for which there is no request for ANI digits.	ANI Category
<i>PROCESS_CALLING_CATEGORY_AND_REQUEST_ANI</i>	Processes calling category and requests ANI digits.	ANI Category
<i>PROCESS_CALLING_CATEGORY_AND_REJECT_CALL</i>	Processes a calling category that results in the rejection of the call.	ANI Category
<i>PROCESS_CALLING_CATEGORY_REQUEST_DENIED</i>	Process denied request for the calling category.	ANI Category
<i>PROCESS_ANI_DIGIT</i>	Process received ANI digit.	ANI
<i>PROCESS_ANI_END_OF_ID</i>	Process end of ANI signal when digits are available.	ANI
<i>PROCESS_ANI_NOT_AVAILABLE</i>	Process signal indicating that ANI digits are not available.	ANI
<i>PROCESS_ANI_RESTRICTED</i>	Process signal indicating that ANI is restricted.	ANI

Inbound calls require generation of call progress tones. If a call has been accepted, BSMI will generate a finite number of RING tone cycles before entering the answered state. If a call has been rejected, BSMI will generate the BUSY signal until the call is cleared by the outbound side. [Table 24](#) identifies the IISDN_CPGEN_MF_PARAMS structure that contains the parameters required to define a call progress signal for generation by a DSP resource.

Table 24. IISDN_CPGEN_MF_PARAMS

Name	Description	Units	Range
<i>freqTone1</i>	Frequency of tone #1	Hz	[0-4000]
<i>powerTone1</i>	Power of tone #1	0.5 dB, relative to power0dBm Output	
<i>freqTone2</i>	Frequency of tone #2	Hz	[0-4000]
<i>powerTone2</i>	Power of tone #2	0.5 dB, relative to power0dBm Output	
<i>numCadences</i>	Number of distinct cadences to generate (on/off pairs)		[1,2,3]
<i>makeTime1</i>	Duration of first ON cadence	ms	[1 - 8191]: tone plays for this duration 0: no generation >0: tone played continuously
<i>breakTime1</i>	Duration of first OFF cadence	ms	[1 - 8191]: tone is off for this duration 0: silence not generated between ON cadences < 0: silence is played continuously
<i>makeTime2</i>	See makeTime1; not used if numCadences = 1		
<i>breakTime2</i>	See breakTime1; not used if numCadences = 1		

Table 24. IISDN_CPGEN_MF_PARAMS (Continued)

Name	Description	Units	Range
<i>makeTime3</i>	See makeTime1; not used if numCadences = 2		
<i>breakTime3</i>	See breakTime1; not used if numCadences = 2		
<i>numCycles</i>	Number of cadence cycles to generate.		0: pattern repeats indefinitely >0: finite number of cycles

R2 Call Control

This section presents an overview of R2 outbound and inbound call setup and tear down. LEC protocols and ISDN use a similar mechanism.

Once the trunk is configured, a particular channel that is enabled is ready to dial an outbound call or process an inbound call from the network. The R2 protocol stack automatically selects a DSP channel (Boston channel) to perform the tone detection/generation operations. The first B-channel on the first line uses the first DSP channel, then each subsequent B-channel selects the next DSP channel.

The first B-channel on the next line selects the DSP channel immediately following the DSP channel selected for the last B-channel of the previous line, thus ensuring there are no gaps in the DSP channel usage caused by different number of timeslots in T1 and E1 lines.

During call setup - either inbound or outbound - the application must not issue any commands to the Tone Generation and Tone Detection facilities, as these would interfere with the R2 signaling and cause the call to be abnormally terminated. If the application connecting DSP channels and B-channels through the Telephony Bus (H.100), it must ensure the appropriate DSP channel is free and connected to the corresponding B-channel when it is ready to receive/place calls. Once the call is connected, the DSP and B-channels are remapped.

Outbound Call Setup

An outbound call request sequence that results in call acceptance by the remote side is illustrated in [Figure 54](#). The host application issues the module an L4L3mCALL_REQUEST message with the following R2 payload entries:

```
l43msg.msgtype = L4L3mCALL_REQUEST;
l43msg.lapdid  = 0x1;
l43msg.data.call_req_data.bchannel = 1;

l43msg.data.call_req_data.called_party.num_digits
l43msg.data.call_req_data.called_party.dnis_category
(set to an IISDN_R2MF_SIGNAL_CODES value)
l43msg.data.call_req_data.called_party.digits (ASCII
string)
l43msg.data.call_req_data.calling_party.num_digits
l43msg.data.call_req_data.calling_party.ani_category
(set to an IISDN_R2MF_SIGNAL_CODES value)
l43msg.data.call_req_data.calling_party.presentation_ind
(set to an IISDN_R2_ANI_XXXX value)
l43msg.data.call_req_data.calling_party.digits (ASCII
string)
l43msg.data.call_req_data.call_type (set to
IISDNcalltypR2_GND for a terrestrial circuit or
IISDNcalltypR2_SAT for a satellite circuit)
```

Call acceptance by the remote end is determined by BSMI during MFC inter-register signaling. At this point, BSMI issues an L3L4mALERTING message to the host containing the IISDN_R2_CALL_STATUS structure. The structure element

```
l34msg.data.al_con_data.r2_call_status.call_status,
```

provides indication (IISDN_R2_CALL_STATUS_CODES) of whether the inbound side provided a Group B line condition (IISDNR2statGROUP_B_AVAILABLE), or whether the call was accepted without such indication (IISDNR2statGROUP_A_CALL_ACCEPTED). In the case of the former, element

```
l34msg.data.al_con_data.r2_call_status.group_B
```

indicates the Group B call acceptance code.

When the inbound side answers the call, BSMI issues an L3L4mCONNECT message to the host.

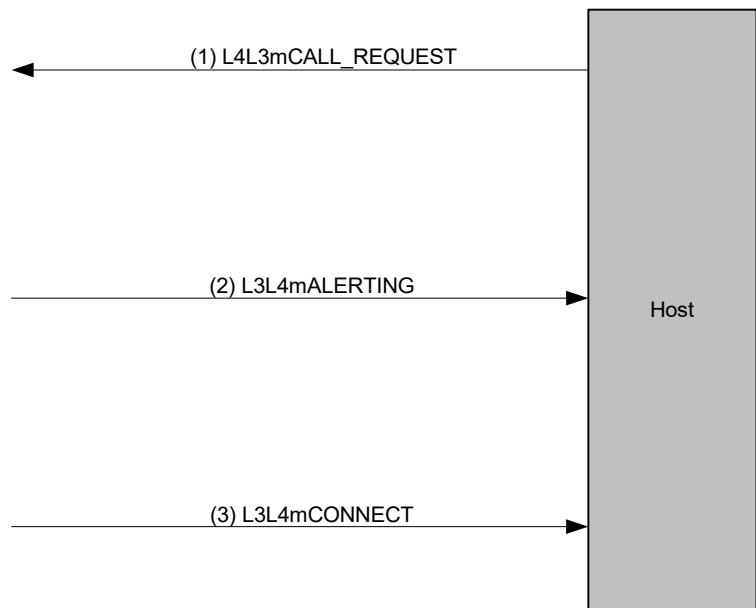


Figure 54. Outbound Call Accepted

An outbound call request sequence that results in call rejection by the remote side is illustrated in [Figure 55](#). In this case, BSMI determines that the call has been rejected during inter-register signaling and automatically clears the call request. BSMI then provides the host indication of the failed call attempt with the L3L4mCLEAR_REQUEST message. As in the case of call acceptance, this message contains the IISDN_R2_CALL_STATUS structure that is examined to determine the precise reason for call failure. Normally, the call has been rejected by the remote side and a status code of IISDNR2statGROUP_B_AVAILABLE is indicated with the associated Group B reason for the rejection. However, other failures are possible during abnormal operation and are indicated by other values in enumeration IISDN_R2_CALL_STATUS_CODES.

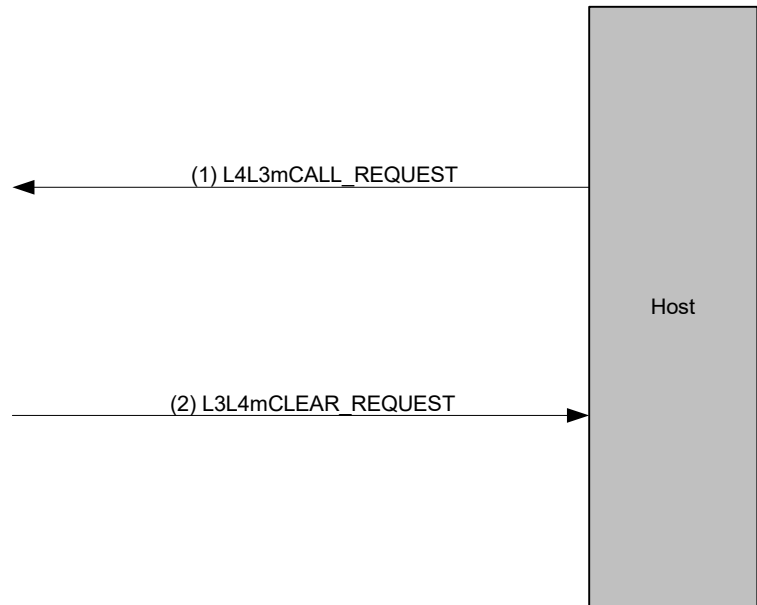


Figure 55. Outbound Call Rejected

Inbound Call Setup

Inbound call setup is illustrated in [Figure 56](#). The host optionally receives an L3L4mPRE_SEIZE message when BSMI detects an incoming seizure. When all DNIS and ANI address information has been collected, BSMI issues an L3L4mSETUP_IND message. After examination of the address information, the host application might wish to accept or reject the call.

In the case of call acceptance, the host provides an appropriate Group B line condition in the L4L3m_CONNECT_REQUEST message,

```
l43msg.data.al_con_data.r2_call_status.group_B = IISDN_R2MFCP_LINE_FREE_NO_CHARGE;
```

Also, the number of ring cycles might be varied from the default setting in [Table 19](#) by setting the following parameter to a non-zero value,

```
l43msg.data.al_con_data.r2_call_status.numberRings = 2;
```

The trunk can also be configured to accept all calls via the inter-register parameter “addressCompleteMode.” In this case, the L4L3m_CONNECT_REQUEST message is still sent, although the Group B code is not required. Following completion of R2 MF register signaling, BSMI will automatically play the RING signal for a finite number of cycles. When the ring signal completes, BSMI enters the answered state and issues an L3L4mCONN_ACK_IND message to the host.

In the case of call rejection, the host provides an appropriate Group B line condition in an L4L3m_CLEAR_REQUEST message,

```
l43msg.data.clr_data.r2_call_status.group_B = IISDN_R2MFCP_LINE_BUSY;
```

Following completion of R2 MF register signaling, BSMI will automatically play the BUSY signal for an indefinite number of cycles. When the call has been cleared by the outbound side, BSMI will issue an L3L4m_CLEAR_REQUEST message to the host after both sides have assumed the idle state.

In the event that a failure occurs after the optional L3L4mPRE_SEIZE message, but before the L3L4mSETUP_IND message, BSMI will issue an L3L4mSTATUS_IND message containing the IISDN_R2_CALL_STATUS structure that identifies the reason for the call failure. No host action is required in this case, but the status indication is logged for informational purposes.

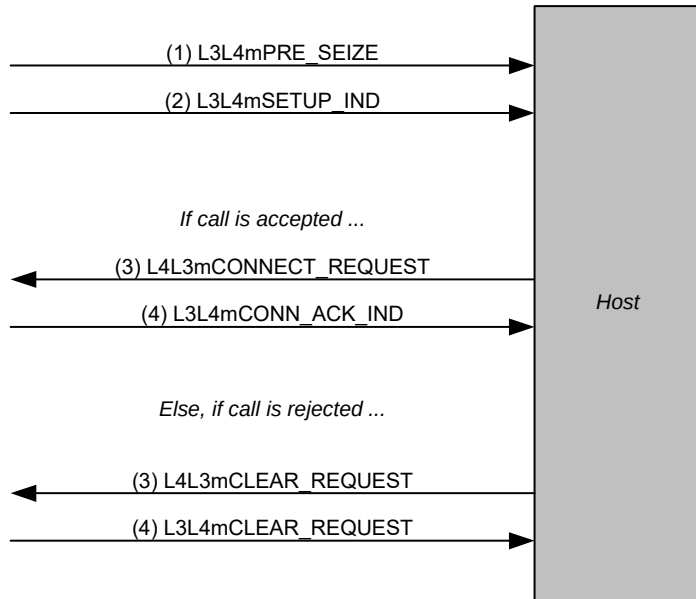


Figure 56. Inbound Call Setup

Call Tear Down

Figure 57 illustrates a call disconnection initiated by the network. BSMI issues an L3L4mDISCONNECT message. The host responds with a clear request that is followed with an L3L4mCLEAR_REQUEST message when the channel has returned to the idle state.

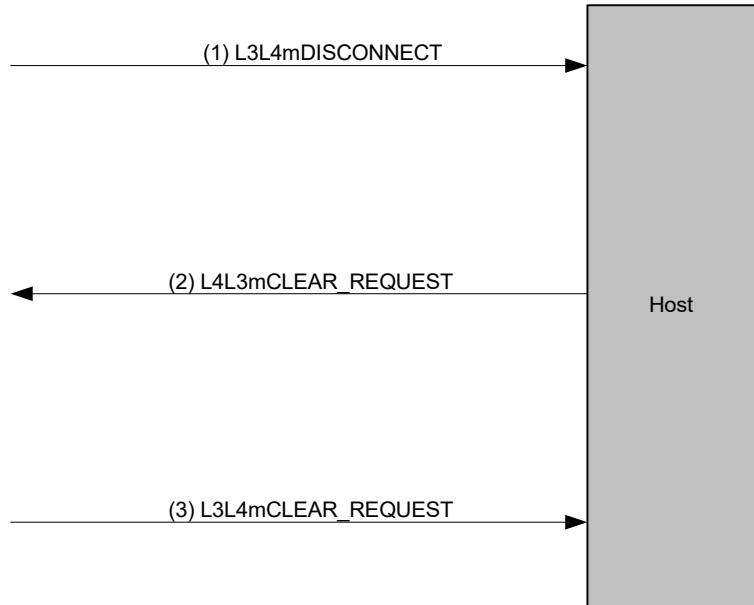


Figure 57. Call Cleared by the Network

Figure 58 illustrates a call disconnection initiated by the host application. The host issues an L4L3mCLEAR_REQUEST message. BSMI responds with an L3L4mCLEAR_REQUEST message when the channel has returned to the idle state.

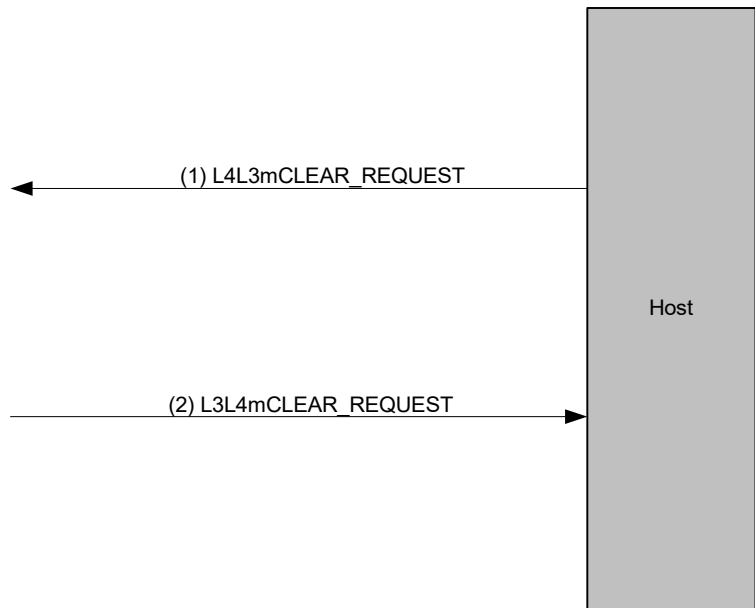


Figure 58. Call Cleared by the Module

Channel Blocking

BSMI provides a mechanism for the host application to block and unblock individual channels as well as to receive indication that the far end has blocked or unblocked a particular channel. Although the R2 protocol remains “enabled”, a blocked channel is not available for outbound or inbound calls. The BSMI message sequences for local blocking and blocking by the network are illustrated in [Figure 59](#).

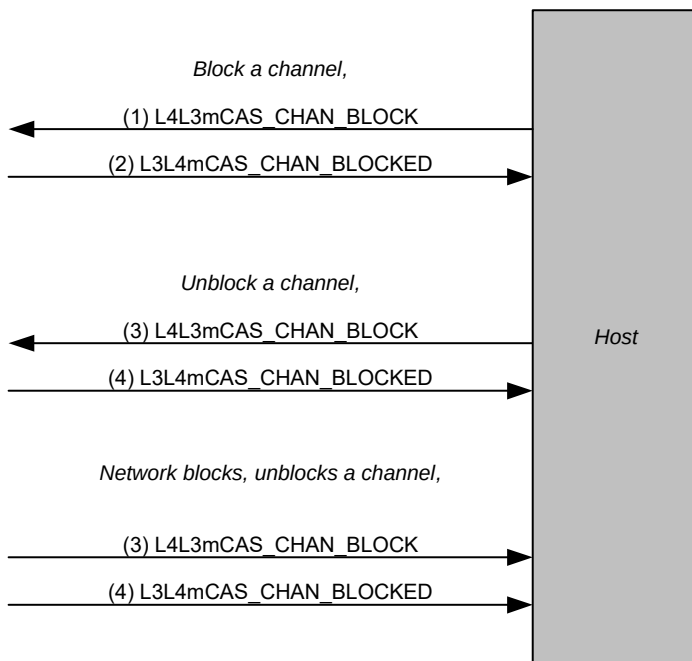


Figure 59. Channel Blocking

10 - Packaging Your Application for Windows

This chapter describes how to package Dialogic® Brooktrout® software so that you can deliver it to your customers as part of your product.

The chapter has the following sections:

- [*Package Options on page 365*](#)
- [*Installation on page 366*](#)
- [*About Plug and Play Components on page 392*](#)
- [*Modifying Configuration Files on page 397*](#)
- [*Including the Brooktrout Configuration Tool on page 398*](#)
- [*Downloading Firmware Files on page 400*](#)
- [*Removing Software on page 401*](#)
- [*Removing the Plug and Play Driver on page 402*](#)

The Dialogic® Brooktrout® software includes an installation package (*boston.msi*) to help you easily package your application for deployment on a Windows based platform. This package is referred to as the Dialogic® Brooktrout® Runtime Software. This installation package provides you with the option to distribute Brooktrout runtime software with your application by spawning it from your own installation program.

You can also select appropriate components of the Dialogic® Brooktrout® SDK installation package (*sdk_windows.exe*) and create your own installation package by:

- Integrating the required merge modules (*.msm file extension) into your own installation package a merge module consists of a component such as a .dll file and its related files, resources, registry entries, and setup logic. To use the merge modules packaging option of Windows Installer, Dialogic grouped features of the Dialogic® Brooktrout® SDK into selectable modules that allow you to choose:
 - ◆ The features you want to include in your package
 - ◆ The location of the installed features
 - ◆ The conditions governing when to install the features
- Selecting your application's required runtime objects and including these Brooktrout files inside your own package as you can do with other operating systems.

The Brooktrout Runtime Software consists of a collection of software components in a simple installation package known as *boston.msi* that installs the basic drivers and other required runtime components to support modules. These components include:

- Library (dll) files excluding those required to develop your application
- Firmware files
- Protocol files
- Configuration files
- Brooktrout Configuration Tool
- Additional supporting utilities

You can also select appropriate components of Brooktrout Fax Software and create your own installation package by:

- Integrating the required merge modules (*.msm file extension) into your own installation package
- Selecting your application's required runtime objects and including these Brooktrout files inside your own package as you can do with other operating systems.

Windows Server 64-bit

Installing the Dialogic® Brooktrout® Runtime Software installs a second package called Visual Studio 2005 Components. This package appears in the Add/Remove Programs list of packages installed. This package contains the Visual Studio 2005 64-bit libraries needed by the driver to run on a Windows 64-bit platform. To remove this package, you need to remove the main package, Dialogic® Brooktrout® Runtime Software. The main package removes all of its dependencies, including the Visual Studio 2005 Components package. If you try to remove Visual Studio 2005 Components independently the system will return an error.

If you have developed an application using the 64-bit Bfv API, you will also need to install the 64-bit Brooktrout components separately. The 64-bit Brooktrout components can be installed using the Brooktrout_x64.msi installation package.

Note: The Brooktrout_x64.msi installation package is not required if you are running a 32-bit application on a 64-bit Windows operating system.

This chapter helps you to understand your distribution options, how to implement them, and how to customize plug and play installation. Using this information to guide you, you can create an installation package that will help your customers successfully install Brooktrout Fax Software and hardware.

Package Options

To redistribute Brooktrout Fax Software with your installation package, choose one of these options:

- Spawn the Dialogic® Brooktrout® Runtime Software installation (*boston.msi*) from your installation program (see [page 367](#))
- Select the software feature modules to merge into your own MSI installation package using the merge modules feature of Windows Installer (see [page 375](#)). The merge modules are available from the Dialogic® Brooktrout® SDK installation (*sdk_windows.exe*)

-
- Choose software files from the Dialogic® Brooktrout® SDK and create your own installation package

Dialogic created its installation package using InstallShield Developer Version 2009 that supports Windows Installer Version 2.0.

Installation

This section provide instructions for the following:

- Installing boards and virtual modules (SR140)
- Installing software

Installing Modules

Your customers can install either the software first or the module first. Brooktrout's application supports both methods. In your instructions to customers, tell them to do the following if they install the module before installing software:

1. Turn off the computer.
2. Install the module.
3. Restart the computer.
4. Place the CD in the computer.
5. At this point, the Found New Hardware application starts up.
6. Answer screen questions.

When the software asks for a driver location, the customer must select the CD.

Note: For computer systems that need plug and play drivers, it helps to store the WinPnP folder at a readily available location on the CD.

Note: If you choose to use *boston.msi* for deploying Brooktrout Fax Software with your product, the recommended method of installation is to first install the software, i.e. *boston.msi*, and then install the hardware.

Installing Virtual Modules (SR140)

If you purchased a SR140 virtual module, follow the instructions in the next section to install the software and then run the license manager to activate your virtual module.

Installing Software

Depending on how you choose to deliver your package to your customers, Dialogic provides its software as an MSI package (*boston.msi*) or as merge modules that you can merge into an existing MSI package. This section discusses these options and provides instructions to implement them.

Installing the Brooktrout Runtime Software

Dialogic delivers this installation package as a Microsoft Software Install (*msi* file extension) module to allow you to distribute Brooktrout runtime software and also to integrate the package within your setup (installation) program. In this case, integration means the ability to launch this package from another program. Our package can either be spawned by:

- Creating a link to *boston.msi* in your autorun application, or
- Double clicking on the *boston.msi* file.

Options for Spawning MSI

Use one of the following Windows Installer Options to spawn the Dialogic® Brooktrout® Runtime Software (*boston.msi*) installation package so that you can use it or adapt it for your own setup (installation) program.

1. Use the */I* option to install the product.

Example:

```
C:\WINDOWS\System32\msiexec /I D:\boston.msi
```

where, "C:\WINDOWS\System32\msiexec" is the executable name including the path.

/I D:\boston.msi are the command line arguments.

2. Use the */qn* option to run the *boston.msi* package in silent mode.

Example: `msiexec /I boston.msi /qn`

3. Use the `/L` option to print the output of the install to a log file.

Example: `msiexec /I boston.msi /qn /L boston_msi.log`

4. Use the following syntax to set the public properties of the `boston.msi` package:

```
msiexec /I boston.msi INSTALLDIR=C:\ProductFolder REBOOT=1
```



If you use your own installation program instead of *boston.msi* and the Brooktrout *INF* file to install Brooktrout files, check for the presence of Brooktrout Fax Software by examining the registry entries (see [page 372](#)). Remove the entries before installing the new Brooktrout version.

The *boston.msi* package only installs the runtime components such as the firmware, configuration files, and configuration tools. This collection installs under *[INSTALLDIR]\bin*.

INSTALLDIR is the directory where you choose to install the Brooktrout Fax Software. The default is *[WINDOWS_VOL]\Program Files\Brooktrout* where *WINDOWS_VOL* is the drive on which your Operating System is installed.

The device driver is installed under *windows\system32\drivers*.

The *bostsrv.exe* service is installed under *windows\system32* directories.

When you install this package, it creates a *bin* directory under an *INSTALLDIR* folder that contains all the runtime objects required for the application you developed using the Brooktrout SDK.

The Dialogic® Brooktrout® Runtime Software provides the files listed in [Table 25](#).

Note: Note: Do not install the Dialogic® Brooktrout® Runtime Software and the Dialogic® Brooktrout® SDK on the same machine. If you install the two packages, files will be duplicated on different locations because the Dialogic® Brooktrout® Runtime Software package contains a subset of the files in the Dialogic® Brooktrout® SDK. Managing the

duplicate files may cause issues due to the installation folder structure being different. For example the location of the Dialogic® Brooktrout® Configuration Tool.

Table 25. Brooktrout Fax Software System Files

Install Location	File Name	Purpose
<i>INSTALLDIR\</i> : All other folders are under that folder.	Brooktrout Windows End User Guide.pdf	Brooktrout license agreement
bin\	AxisClient.dll	License Manager file
bin\	AxisTransport.dll	License Manager file
bin\	AxisXMLParser.dll	License Manager file
bin\	axis_notice.txt	License notice text file
bin\	bostvb.dll	Host based firmware dll
bin\	brktcctrace.exe	Call Tracer utility file
bin\	brktlicmgr.exe	License Manager file
bin\	brktlicmgrhelp.zip	License Manager Help files
bin\	btver.exe	Brooktrout Fax Software and firmware version information
bin\	confighelp.zip	Help files for the Brooktrout Configuration Tool
bin\	configtool.exe	Brooktrout Configuration Tool
bin\	connect.gif	License Manager file
bin\	cp.bin	Firmware
bin\	dh.exe	Dump History stand-alone utility
bin\	dsp1000.hex	Firmware
bin\	dsp1000_ld.hex	Firmware
bin\	dsp1000_ud.hex	Firmware
bin\	dsp1000_v34.hex	Firmware
bin\	dsp1034_ud.hex	Firmware
bin\	feature.exe	Utility to query or download the modules features
bin\	filtersettings.cfg	Call Tracer Utility Filter Settings configuration file
bin\	firm.exe	Firmware flash update file
bin\	firmload.exe	Firmware update utility file
bin\	FulfillResources.properties	License Manager file

Table 25. Brooktrout Fax Software System Files (Continued)

Install Location	File Name	Purpose
bin \	libeay32.dll	License Manager file
bin \	modinfo.exe	Module information utilities
bin \	SSLeay32.dll	License Manager file
bin \	vttty_tracer.exe	ISDN Message and Network Layer Tracing utility
bin \	xerces-c_2_2_0.dll	License Manager file
bin \	xerces_license.txt	License Manager text file
config \	analog_loopstart_europe.lec	Protocol file
config \	analog_loopstart_us.lec	Protocol file
config \	<i>btcall.cfg</i>	User-defined configuration file
config \	BT_CPARM.CFG	Country-specific configuration parameters file
config \	<i>callctrl.cfg</i>	Call control configuration file
config \	ctr21.qslac	Protocol file
config \	epsonec.fnt	Font file
config \	epsonec.fz8	Font file
config \	epsones.fnt	Font file
config \	epsones.fz8	Font file
config \	epsonpc.fnt	Font file
config \	epsonpc.fz8	Font file
config \	epsonps.fnt	Font file
config \	epsonps.fz8	Font file
config \	fxo_groundstart.lec	Protocol file
config \	fxo_loopstart.lec	Protocol file
config \	fxs_groundstart.lec	Protocol file
config \	fxs_loopstart.lec	Protocol file
config \	ibmpcps.fnt	Font file
config \	ibmpcps.fz8	Font file
config \	immediatedial.lec	Protocol file
config \	itu_argentina.r2	Country specific configuration file
config \	itu_brazil.r2	Country specific configuration file
config \	itu_china.r2	Country specific configuration file
config \	itu_egypt.r2	Country specific configuration file

Table 25. Brooktrout Fax Software System Files (Continued)

Install Location	File Name	Purpose
config\	itu_honduras.r2	Country specific configuration file
config\	itu_korea.r2	Country specific configuration file
config\	itu_mexico.r2	Country specific configuration file
config\	us600.qslac	Protocol file
config\	winkstart.lec	Protocol file
driver\install\	install.exe	Installation file for 32-bit OS
driver\install\	installx64.exe	Installation file for 64-bit OS. This file is only present when the package is installed on a 64-bit OS.
driver\pnp\	brooktrout.cat	Security catalog file for the pnp driver components
driver\pnp\	trxstream.inf	INF file for installing pnp boston
driver\pnp\ TRxStream\x86\	boston.pdb	Driver symbol file for 32-bit OS
driver\pnp\ TRxStream\x86	boston.sys	Plug and Play driver for 32-bit OS
driver\pnp\ TRxStream\x86	bfktBdevpp.pdb	Property Page symbol file for 32-bit OS
driver\pnp\ TRxStream\x86	brktBdevpp.dll	Device Property Page dll for 32-bit OS
driver\pnp\ TRxStream\x64\	boston.pdb	Driver symbol file for 64-bit OS
driver\pnp\ TRxStream\x64	boston.sys	Plug and Play driver for 64-bit OS
driver\pnp\ TRxStream\x64	bfktBdevpp.pdb	Property Page symbol file for 64-bit OS
driver\pnp\ TRxStream\x64	brktBdevpp.dll	Device Property Page dll for 64-bit OS
%System Root%\ system32\	bostdll.dll	Bfv API library file for 32-bit OS
%System Root%\ system32\	bostsrv.dll	Boston Host Service dll for 32-bit OS
%System Root%\ system32\	bostsrv.exe	Boston Host Service for 32-bit OS
%System Root%\ system32\	brkth323.dll	H.323 library file for 32-bit OS
%System Root%\ system32\	brktsip.dll	SIP library file for 32-bit OS

Table 25. Brooktrout Fax Software System Files (Continued)

Install Location	File Name	Purpose
%System Root%\syswow64\	bostdll.dll	Bfv API library file for 64-bit OS
%System Root%\syswow64\	bostsrv.dll	Boston Host Service dll for 64-bit OS
%System Root%\syswow64\	bostsrv.exe	Boston Host Service for 64-bit OS
%System Root%\syswow64\	brkth323.dll	H.323 library file for 64-bit OS
%System Root%\syswow64\	brktsip.dll	SIP library file for 64-bit OS

In addition to copying the files to the destination folders, this option registers (service installation) *bostsrv.exe* with the host system. The *bostsrv.exe* service registers with a dependency on the *boston.sys* device driver.

Because the *boston.msi* also installs the plug and play boston device driver, you do not have to manually install the pnp driver using the Found New Hardware Wizard. This installation occurs automatically.

Dialogic recommends that all the above tools/files that are provided from Dialogic be installed with your software. If your customer ever needs assistance from Dialogic to configure or troubleshoot a problem, Dialogic Technical Services and Support might ask your customer to run or view these files to ensure the configuration is set up properly.

Registry Entries

The installation creates one of the following keys:

■ *For a 32-bit system:*

```
HKEY_LOCAL_MACHINE\SOFTWARE\Brooktrout Technology  
  \Brooktrout System Software
```

■ *For a 64-bit system:*

```
HKEY_LOCAL_MACHINE\SOFTWARE\wow6432Node  
  \Brooktrout Technology\Brooktrout System Software
```

and adds the following entries:

Name	Value
Install Home	[INSTALLDIR] where INSTALLDIR is the directory you selected to install the Brooktrout Fax Software. Default location value: [WindowsVolume]\Program Files\Brooktrout where WindowsVolume is the drive where the operating system is installed.
Runtime Configtool Path	[INSTALLDIR]\bin where INSTALLDIR is the directory you selected to install the Brooktrout Fax Software. Default location value: [WindowsVolume]\Program Files\Brooktrout where WindowsVolume is the drive where the operating system is installed.
Version	Brooktrout Software Version (for example, 6.4)

Shortcuts

The installation creates the following shortcut under Start -> Programs:

Name	Location
Brooktrout Configuration Tool	[INSTALLDIR]\bin\configtool.exe Where INSTALLDIR is the directory that you selected to install the configuration tool. The Brooktrout Configuration Tool is a utility that is used to create and modify the configuration files, edit and update the driver parameters, and configure and initialize both physical and virtual modules. See Including the Brooktrout Configuration Tool on page 398.
Brooktrout License Manager	[INSTALLDIR]\bin\brktlicmgr.exe Where INSTALLDIR is the directory that you selected to install the Brooktrout Fax Software. The Brooktrout License Manager is a utility that is used to obtain, install, update, and remove the SR140 license.

Reboot Options

When the driver is installed, the system must be rebooted but only under certain circumstances which are detected during the driver installation.

The MSI installs the driver during installation and removes it during the un-installation. Sometimes the installation or removal of the driver requires the need for a system reboot for the changes to take effect.

The system creates the following registry keys and values if the package detects that a reboot is needed during the installation and un-installation. These key entries will be present only if a reboot is required.

For 64-bit system

Key Generated

HKEY_LOCAL_MACHINE\Software\brkttmp

Value Generated

HKEY_LOCAL_MACHINE\Software\brkttmp\reboot

For 64-bit system

Key Generated

HKEY_LOCAL_MACHINE\Wow6432Node\Software\brkttmp

Value Generated

HKEY_LOCAL_MACHINE\Wow6432Node\Software\brkttmp\reboot

During the full installation of the User Interface, a dialog appears indicating that a reboot is necessary. However, if you are running a quiet installation, all dialog boxes are suppressed including the reboot dialog box.

You will have to refer to the reboot values above stored in the registry to know if a reboot is necessary.

A REBOOT public property is available in the package to automatically reboot the package at the end of the installation. Use it as follows:

- If the property is set to 1 and a reboot is required, the package reboots automatically.
- If the property is set to 0 and a reboot is needed, the package will not reboot automatically and the registry key needs to be used.

If you run *boston.msi* in quiet mode without changing the REBOOT property, the package will not automatically reboot - even if required. You must refer to the registry to know if a reboot is required.

Installation

Enter the following from a command line to install the package.

```
msiexec /i boston.msi /qn
```

Enter the following from a command line to remove the package:

```
msiexec /x boston.msi /qn
```

If you change the REBOOT property to 1 and a reboot is required, the package will automatically reboot when the installation is completed.

The following are examples of installing and removing the package using the REBOOT property above.

Install:

```
msiexec /i boston.msi /qn REBOOT=1
```

Remove:

```
msiexec /x boston.msi /qn REBOOT=1
```

In these cases, if a reboot is needed (install or remove) the package will automatically reboot.

About the Merge Module Feature

A Microsoft Software Merge (MSM) module file consists of a simplified MSI database created to deliver components to an MSI application package. Like an MSI package, it contains instructions, components, and setup logic. Unlike an MSI package, it cannot be separately installed and must be merged into an existing MSI package. The merging process permanently alters the original MSI package by adding the merge module's components and logic to it. However, the merge module itself is not changed by the merging process and is reused when no components in the module need updating.

Each merge module contains unique version information that the Windows Installer database maintains for each application to prevent premature removal of a component that the application needs. After you have included a module in your install package, any incremental version of a component in the module makes it necessary to create a new merge module. When Dialogic supplies an updated merge module, you must remove the old module and merge the new module into your existing MSI package. Merging in a new module eliminates problems such as version conflicts, missing registry entries, and improperly installed files.

To allow you to implement the merge modules feature of Windows Installer, Dialogic separated the files into several sets of features and created one module for each set. You can merge one or more of these modules into your installation package depending on the features you want to include. The following paragraphs indicate when to use each module and what each module includes.

The Brooktrout merge modules consist of:

- [Dynamically Linked DLLs \(dynamic_dlls.msm\)](#) on [page 376](#)
- [Dynamically Linked 64-bit DLLs \(dynamic_dlls_x64.msm\)](#) on [page 377](#)
- [Configuration and Protocol Files \(configdata.msm\)](#) on [page 378](#)
- [Firmware \(firmware.msm\)](#) on [page 379](#)
- [SR140 Firmware \(Firmware_sr140.msm\)](#) on [page 379](#)
- [Brooktrout Configuration Tool \(configtool.msm\)](#) on [page 380](#)
- [License Manager \(softwarelicense.msm\)](#) on [page 382](#)
- [Utility Programs \(utilities.msm\)](#) on [page 383](#)
- [Boston Host Service \(bostsrv.msm\)](#) on [page 384](#)
- [TECUpdate \(TECUpdate.msm\)](#) on [page 381](#)

Dynamically Linked DLLs (dynamic_dlls.msm)

Merge this module to install the dynamically linked version of the Bfv API DLLs and supporting files.

Files Installed

The module contains the following files:

File Name
bostdll.dll
brkth323.dll
brktsip.dll
osidlld.dll
bsmidlld.dll

Registry Entries

None

Environment Variables

None

Shortcuts	None
Services	None
Dependency	None

Dynamically Linked 64-bit DLLs (dynamic_dlls_x64.msm)

Merge this module to install the dynamically linked version of the 64-bit Bfv API DLLs and supporting files.

Files Installed The module contains the following files:

File Name
bostdll.dll
osidll.dll
bsmidll.dll

Registry Entries	None
Environment Variables	None
Shortcuts	None
Services	None
Dependency	None

Configuration and Protocol Files (configdata.msm)

Merge this module to install the configuration, protocol, and font files.

Files Installed

The module contains the following files:

File Names	
<i>analog_loopstart_europe.lec</i>	<i>fxo_loopstart.lec</i>
<i>analog_loopstart_us.lec</i>	<i>fxs_groundstart.lec</i>
<i>btcall.cfg</i>	<i>fxs_loopstart.lec</i>
<i>BT_CPARM.CFG</i>	<i>ibmpcps.fnt</i>
<i>callctrl.cfg</i>	<i>ibmpcps.fz8</i>
<i>ctr21.qslac</i>	<i>immediatedial.lec</i>
<i>epsonec.fnt</i>	<i>itu_argentina.r2</i>
<i>epsonec.fz8</i>	<i>itu_brazil.r2</i>
<i>epsones.fnt</i>	<i>itu_china.r2</i>
<i>epsones.fz8</i>	<i>itu_egypt.r2</i>
<i>epsonpc.fnt</i>	<i>itu_honduras.r2</i>
<i>epsonpc.fz8</i>	<i>itu_korea.r2</i>
<i>epsonps.fnt</i>	<i>itu_mexico.r2</i>
<i>epsonps.fz8</i>	<i>us600.qslac</i>
<i>fxo_groundstart.lec</i>	<i>winkstart.lec</i>

Registry Entries

None

Environment Variables

None

Shortcuts

None

Services

None

Dependency

None

Firmware (firmware.msm)

Merge this module to install the firmware files.

Files Installed

The module contains the following files:

File Name
cp.bin
dsp1000.hex
dsp1000_ld.hex
dsp1000_ud.hex
dsp1000_v34.hex
dsp1034_ud.hex

Registry Entries

None

Environment Variables

None

Shortcuts

None

Services

None

Dependency

None

SR140 Firmware (Firmware_sr140.msm)

Merge this module to install the SR140 firmware files.

Files Installed

The module contains the following files:

File Name
bostvb.dll

Registry Entries

None

Environment Variables

None

Shortcuts

None

Services

None

Dependency

None

Brooktrout Configuration Tool (configtool.msm)

Merge this module to install the Brooktrout Configuration Tool and its help files.

Files Installed

The module contains the following files:

File Name
configtool.exe
confighelp.zip

Registry Entries

RuntimeConfigtoolPath

configtool.exe

Environment Variables

None

Shortcuts

Creates an advertised shortcut under Start Menu → Program files.

Shortcut Name	Target
Brooktrout Configuration Tool	configtool.exe

Services

None

Dependency

➤ **Depends on the following modules:**

1. Dynamically linked DLL (*dynamic_dlls.msm*)
2. Configuration and Protocol files (*configdata.msm*)
3. Firmware (*firmware.msm*)
4. SR140 Firmware (Firmware_sr140.msm) only if using SR140.

TECUpdate (TECUpdate.msm)

Distribute TECUpdate to your end users even if you do not plan to distribute the Brooktrout Configuration Tool so that your customers can update their systems as needed. Merge this module to install TECUpdate utility files.

Files Installed

The module contains the following files:

File Name
bostdll210.dll
bostdll301.dll
bostdll303.dll
bostdll310.dll
bostdll320.dll
bostdll330.dll
TECUpdate.exe
TECUpdate_Guide.pdf
TECUpdateHelp.zip

Registry Entries

None

Environment Variables

None

Shortcuts

Creates an advertised shortcut under Start Menu -> Program files.

Shortcut Name	Target
TECUpdate License Upgrade Tool	TECUpdate.exe

Services

None

Dependency

None

License Manager (softwarelicense.msm)

Distribute the License Manager to all end users so that your customers can activate a software license on a system. The License Manager validates the license activated by the customer and turns on the Brooktrout product functionality.

Merge this module to install the license program files.

Files Installed

The module contains the following files:

File Name
axis_notice.txt
AxisClient.dll
AxisTransport.dll
AxisXMLParser.dll
brktlicmgr.exe
brktlicmgrhelp.chm
connect.gif
FulfillResources.properties
libeay32.dll
SSLeay32.dll
xerces_license.txt
xerces-c_2_2_0.dll
dh.exe
feature.exe

Registry Entries

brktlicmgr.exe

Environment Variables

BRKTD_LICENSE_FILE set to C:\Program Files\Common Files\Brooktrout

Shortcuts

Creates an advertised shortcut under Start Menu -> Program files -> Brooktrout.

Shortcut Name	Target
License Manager	brktlicmgr.exe

Services

None

Dependency

None

Utility Programs (utilities.msm)

Merge this module to install the utility programs used for querying hardware and software information, downloading firmware and performing other related tasks.

Files Installed

The module contains the following files:

File Name
brktcctrace.exe
btver.exe
dh.exe
feature.exe
filtersettings.cfg
firm.exe
firmload.exe
modinfo.exe
vttty_tracer.exe

Registry Entries

None

Environment Variables

None

Shortcuts

None

Services

None

Dependency

➤ **Depends on the following modules:**

1. Dynamically linked DLL (*dynamic_dlls.msm*)
2. Configuration and Protocol files (*configdata.msm*)
3. Firmware (*firmware.msm*)
4. SR140 Firmware (Firmware_sr140.msm) only if using SR140.

Boston Host Service (bostsrv.msm)

Merge this module to install and register the Boston Host Service.

Files Installed

The module contains the following files:

File Number	File Name	Default Location
1	bostsrv.exe	System Folder (Windows\System32)
2	bostsrv.dll	System Folder

Registry Entries

The *bostsrv.msm* module creates the following registry entry.

HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet\
Services\bostservice\Parameters

Set values for the following properties in your installer program to provide the Boston Host Service module with the location of these files:

Name

Value

btcall

[BTCALLLOCATION] where BTCALLLOCATION identifies where your installation application installs the *btcall.cfg* configuration file.

Note: This property holds the path and file name of the *btcall.cfg* configuration file.

firmware_path

[FIRMWARELOCATION] where FIRMWARELOCATION identifies where your installation application installs the folder containing the firmware files.

Environment Variables

None

Shortcuts

None

Services

The module installs the following service:

Display Name	Service Name	Default Start Mode
Boston Host Service	bostservice	Demand start

Dependency

➤ Depends on the following modules:

1. Dynamically linked DLL (*dynamic_dlls.msm*)
2. Configuration and protocol files (*configdata.msm*)
3. Firmware (*firmware.msm*)
4. SR140 Firmware (Firmware_sr140.msm) only if using SR140.

Installing the Merge Module Feature

The Brooktrout SDK (*sdk_windows.exe*) includes the Merge Module feature that installs all the *.msm files. When you launch *sdk_windows.exe*, the files (merge modules) install under [INSTALLDIR] \mergemodules where INSTALLDIR identifies the location that you select for storing the Brooktrout SDK package.

Integrating the Modules

Microsoft Software Merge (*.msm extension) module files cannot be directly installed on a system. The feature requires you to merge modules into an installer for each application that uses the component. This merging process ensures that the application installs components consistently, and it eliminates problems such as version conflicts, missing registry entries, and improperly installed files.

After installing your Brooktrout SDK, copy the merge modules to a location accessible by your install project.

The next step requires you to add all the dependent merge modules to your project and set their destination paths.

Note: The design of the Brooktrout merge modules excludes any dependencies due to a defect in the InstallShield Developer 8.0 IDE. This defect prevents a developer (Dialogic (formerly Brooktrout) as the developer of the merge modules) from setting the destination path of dependent merge modules. Without a specified destination, InstallShield uses a default destination that installs the merge module files under:

[WINDOWS_VOLUME]

where WindowsVolume is the drive where the operating system is installed.

The Brooktrout merge modules consist of:

- *dynamic_dlls.msm* (independent module)
- *dynamic_dlls_x64.msm* (independent modules)
- *configdata.msm* (independent module)
- *firmware.msm* (independent module)
- *Firmware_sr140.msm* (independent module)
- *configtool.msm* (module with dependencies)
- *softwarelicense.msm* (independent module)
- *utilities.msm* (module with dependencies)
- *bostsrv.msm* (module with dependencies)
- *TECUpdate.msn* (independent module)

You can only install the independent modules as individual features that do not require any other component or feature. The modules with dependencies require the *dynamic_dlls.msm*, *configdata.msm*, and *firmware.msm* (*Firmware_sr140.msm* also if using SR140) modules.

➤ **Using [Figures 60](#) and [61](#) for guidance, perform the following steps to integrate a merge module into your install project:**

1. Create a feature or features for the merge module.
2. Associate the merge module with the feature.
3. Set the install location for the merge module.

This step is mandatory for all merge modules except *bostsrv.msm*.

4. To install the Boston Host Service (*bostsrv.msm*) merge module, create `BTCALLLOCATION` and `FIRMWARE` location properties.
5. Create custom actions to set these properties to appropriate values.

Perform these custom actions after you select the destination folder.

Note: If you do not install the files contained in the *dynamic_dlls.msm* module in the same location as your application, you must add the destination location of the files installed by *dynamic_dlls.msm* to the *Path* system environment variable. Failure to set this system environment variable prevents your application from

linking to the DLLs. With the system variable set, you can simply append the absolute path to these files to the *Path* environment variable.

You can also add the destination location of *.exe files within the merge modules to the system path so that they are executed from anywhere on your system.

Rather than relying on the Path variable, it is strongly recommended that the destination of *dynamic_dlls.msm* and *boostsrv.msm* be set to [SystemFolder], and the destination of *dynamic_dlls_x64.msm* be set to [System64Folder]. This is especially important on a 64-bit version of Windows using 64-bit applications.

Examples

When merging modules into your installation program, you can choose from the following cases:

CASE I

Associate multiple merge modules with a single feature (see [Figure 60](#)). Set the install location of each of the merge modules to that feature's folder property so that changing the install location of the feature also affects the associated merge modules. For example:

1. Create Feature 1 with a folder property of {FEATURE1_INSTALL_LOCATION}.
2. Set the locations of all the merge modules that this feature contains to:
{FEATURE1_INSTALL_LOCATION}

These settings apply any change to this folder location to all the modules set to the original location. If you intend to install one of the merge modules in a different location, you must create a second feature and associate the module with the feature as shown in [Figure 61](#).

CASE II

Create a feature for each merge module. [Figure 61](#) shows an example of this case.

CASE III

Install the module as a required hidden feature when you don't want to expose the merge module to your user.

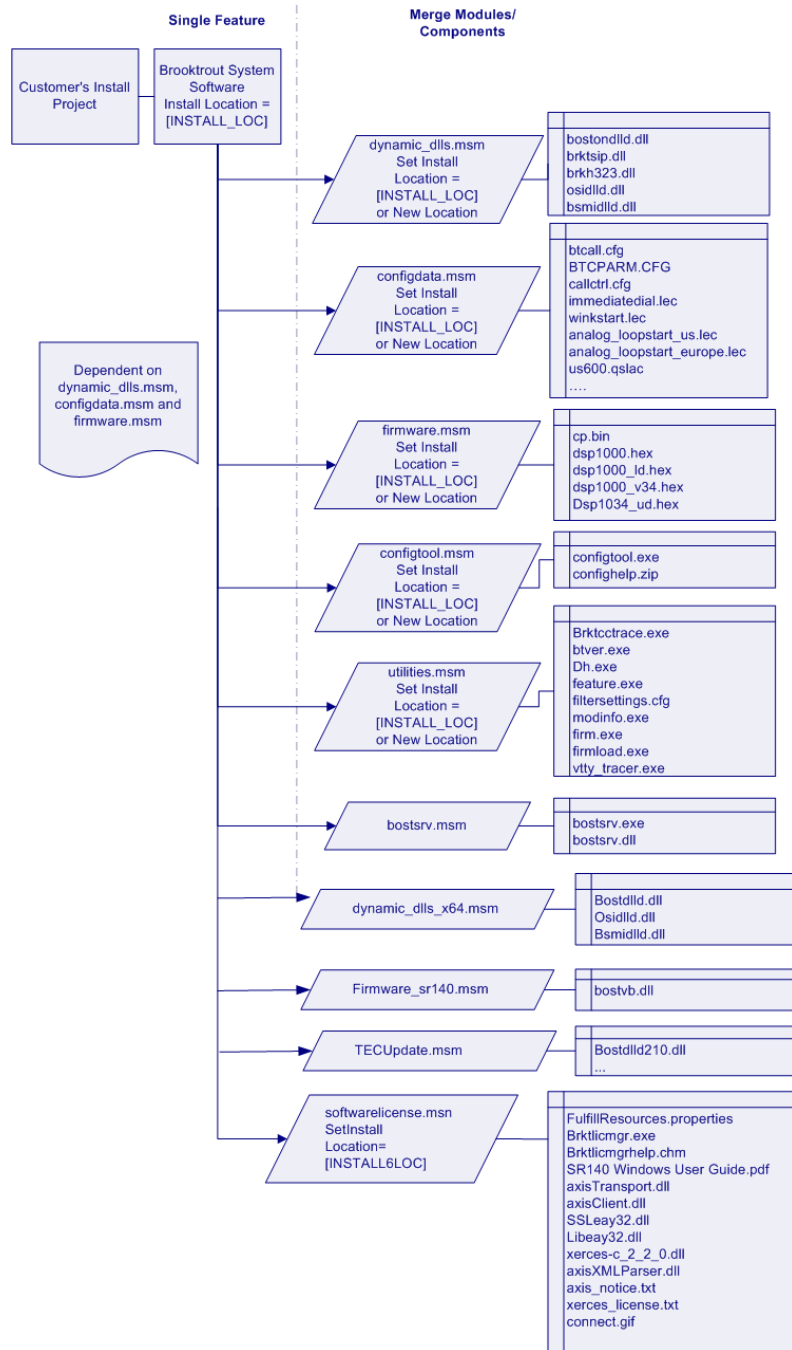


Figure 60. Merging Modules into a Single Feature

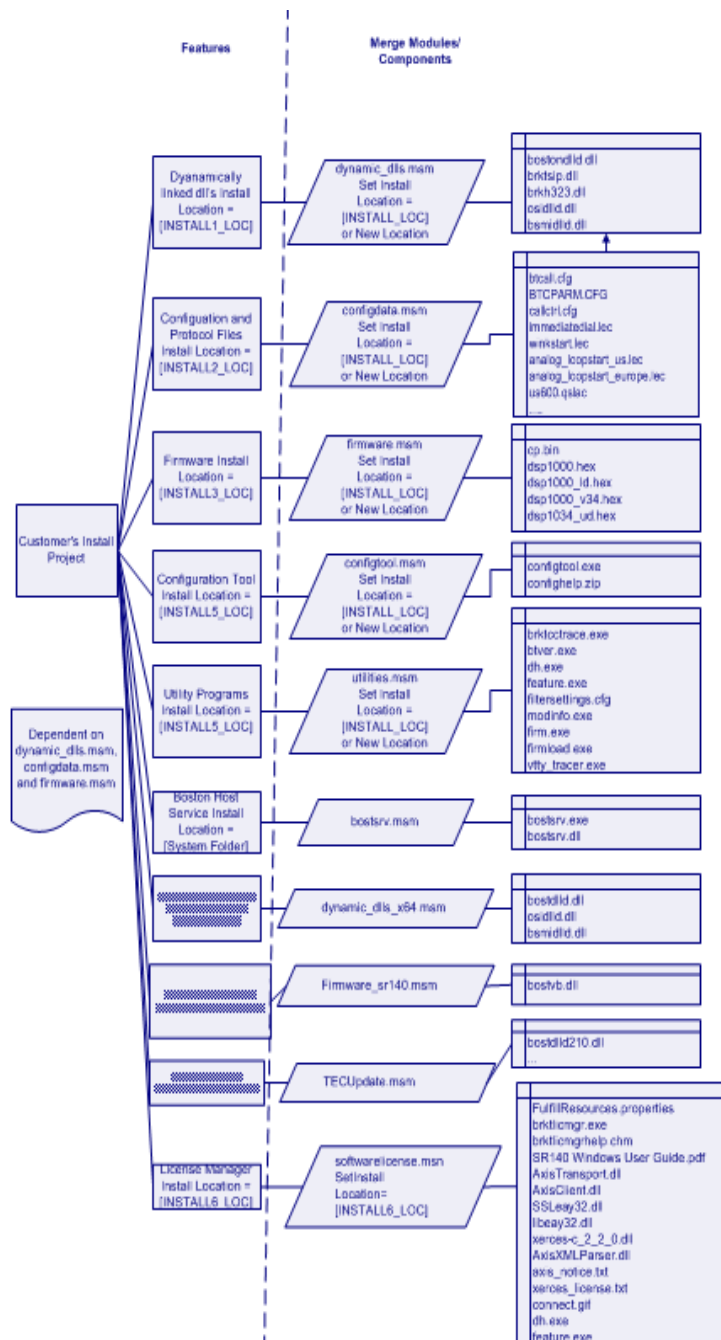


Figure 61. Merging Modules into Multiple Features

Determining Versions of Microsoft Visual C Runtime Components

The Dialogic® Brooktrout® installation packages for the SDK (EXE) and runtime components (MSI) include and install, via redistributable merge modules (MSM) from Microsoft, a version of the MS Visual C 2005 runtime components.

If you want to install components of the Dialogic® Brooktrout® SDK manually, you must also make sure to install the MS Visual C 2005 runtime components. While it is always safe to use the latest version of the runtime component redistributables available for download from Microsoft, it may be desirable to determine the version that is included with the Dialogic® Brooktrout® installation packages.

The Microsoft tool called Orca can be used to determine this. Install and launch Orca, and load the `boston.msi` file. Under the list of Tables, go to the one named `Files`. There will be many, such as `msvcr80.dll.<something>`, which are listed. These all have the same version number, such as 8.0.50727.6195. Use Orca to determine various information about such packages.

If your own CRT redistributable version is earlier than this, then you must make sure to update them for use with the Dialogic® Brooktrout® SDK components. Note that while redistributables can normally be found on a Microsoft web site for download, if a Windows development system has VS 2005 installed, and the Windows Updates are configured to install for other Microsoft products, then it will update your local copy of the runtime component merge modules commonly located in "`c:\program files\common files\merge modules`".

The following is an example of an Orca screen.

boston.msi - Orca									
File Edit Tables Transform Tools View Help									
Tables	File	Comp...	FileNa...	FileSi...	Version	Langu...	Attribu...	Seque...	
Condition	catalog.8.0.50727.6195.97F81AF1_0...	downlevel...	a3t3z6j5...	7318			16384	87	
Control	ATL80.dll.8.0.50727.6195.97F81AF1_...	downlevel...	d3t3z6j5...	97280	8.0.50727.6195	1033	16384	88	
ControlCondition	manifest.8.0.50727.6195.97F81AF1_...	downlevel...	93t3z6j5...	466			16384	89	
ControlEvent	nosxs_ATL80.dll.97F81AF1_0E47_DC9...	nosxs.97...	rwfvltq.l...	97280	8.0.50727.6195	1033	16384	90	
CreateFolder	ul_manifest.66332652_9C28_58B1_FF...	uplevel.6...	n4auwzcy...	804			16384	91	
CustomAction	catalog.8.0.50727.6195.66332652_9...	downlevel...	m4auwzcy...	7340			16384	92	
Dialog	manifest.8.0.50727.6195.66332652_9...	downlevel...	l4auwzcy...	804			16384	93	
Directory	ul_catalog.66332652_9C28_58B1_FF1...	uplevel.6...	o4auwzcy...	7340			16384	94	
DrLocator	ul_msxcr80.dll.98CB24AD_52FB_DB5F...	uplevel.9...	k2rg91xw...	632656	8.0.50727.6195	1033	16384	95	
Environment	ul_msxcr80.dll.98CB24AD_52FB_DB5...	uplevel.9...	m2rg91x...	479232	8.0.50727.6195	1033	16384	96	
Error	ul_catalog.98CB24AD_52FB_DB5F_FF...	uplevel.9...	g2rg91xw...	7318			16384	97	
EventMapping	manifest.8.0.50727.6195.98CB24AD_...	downlevel...	d2rg91xw...	1870			16384	98	
Feature	ul_manifest.98CB24AD_52FB_DB5F_F...	uplevel.9...	f2rg91xw...	1870			16384	99	
FeatureComponents	ul_msxcr80.dll.98CB24AD_52FB_DB5F...	uplevel.9...	l2rg91xw...	554832	8.0.50727.6195	1033	16384	100	
File	msxcr80.dll.8.0.50727.6195.98CB24...	downlevel...	h2rg91xw...	632656	8.0.50727.6195	1033	16384	101	
ISComponentExtended	catalog.8.0.50727.6195.98CB24AD_5...	downlevel...	e2rg91xw...	7318			16384	102	
ISCustomActionReference	msxcr80.dll.8.0.50727.6195.98CB24...	downlevel...	j2rg91xw...	479232	8.0.50727.6195	1033	16384	103	
Icon	msxcr80.dll.8.0.50727.6195.98CB24...	downlevel...	l2rg91xw...	554832	8.0.50727.6195	1033	16384	104	
InstallExecuteSequence	nosxs_msxcr80.dll.98CB24AD_52FB_D...	nosxs.98...	37hpravq...	632656	8.0.50727.6195	1033	16384	105	
InstallUISequence	nosxs_msxcr80.dll.98CB24AD_52FB_...	nosxs.98...	47hpravq...	554832	8.0.50727.6195	1033	16384	106	
LaunchCondition	nosxs_msxcr80.dll.98CB24AD_52FB_...	nosxs.98...	57hpravq...	479232	8.0.50727.6195	1033	16384	107	
ListBox	ul_manifest.63E949F6_03BC_5C40_FF...	uplevel.6...	zxs54we...	804			16384	108	
ListView	catalog.8.0.50727.6195.63E949F6_03...	downlevel...	yxgs54w...	7340			16384	109	
Media	manifest.8.0.50727.6195.63E949F6_0...	downlevel...	xxgs54w...	804			16384	110	
ModuleComponents	ul_catalog.63E949F6_03BC_5C40_FF1...	uplevel.6...	0ygs54w...	7340			16384	111	
ModuleDependency	ul_manifest.9BAE13A2_E7AF_D6C3_F...	uplevel.9...	b21v718o...	2372			16384	112	
ModuleSignature	ul_mfc80.dll.9BAE13A2_E7AF_D6C3...	uplevel.9...	h21v718o...	69632	8.0.50727.6195	1033	16384	113	
MsiAssembly	mfc80.dll.8.0.50727.6195.9BAE13A2...	downlevel...	e21v718o...	1093120	8.0.50727.6195	1033	16384	114	
MsiAssemblyName	catalog.8.0.50727.6195.9BAE13A2_E...	downlevel...	a21v718o...	7318			16384	115	
MsiFileHash	mfc80.dll.8.0.50727.6195.9BAE13A2...	downlevel...	d21v718o...	69632	8.0.50727.6195	1033	16384	116	
MsiSFCBypass	manifest.8.0.50727.6195.9BAE13A2_...	downlevel...	921v718o...	2372			16384	117	

Tables: 55

File - 177 rows

[Version - String[72], Nullable

About Plug and Play Components

Plug and Play happens outside the process of the application that uses the drivers. The Brooktrout SDK includes Plug and Play compatible drivers and the *INF* file. The *INF* file contains essential information needed for the Windows Class Installers to correctly identify components in an *INF* file and install them.

The user installs the hardware in the computer by following the instructions in the hardware installation card that comes with the module the hardware in the computer by following the instructions in the hardware installation guide that comes with the module. When Windows starts and the user logs on, the Found New Hardware Wizard screen appears.



➤ **Follow the steps below:**

1. Select No, not this time and press Next.
2. Select "Install from a list or specific location (Advanced)" then press Next.
3. Select Search for the best driver in these locations. Place the software CD in the system (this is a Brooktrout CD or your application CD with the Brooktrout Plug and Play driver) and press Next.

-
4. Optionally you can select the **Include this location in the search:** and browse to the location of the INF files. The Windows Plug and Play manager finds the driver and installs it on the computer.

Note: Existing Dialogic® Brooktrout® Fax Products SDK

Developers: You cannot start or stop the Plug and Play driver using commands like the `net start/stop` commands that you might be using in your application. The user cannot restart the driver since it is now a Windows Device Manager Plug and Play driver. For any change requiring a driver restart, such as change in history size and debug options, you must:

- ◆ Check for the current state of the service
- ◆ Tell the user to restart Windows if the driver is not started

You can install and uninstall the PnP driver using *install.exe* application that is already installed with the Brooktrout Fax Software.

Dialogic does recommend that the only driver settings that you do change are the *History Enable*, and *History Size*. However, your changes will only take effect by rebooting the system or by using the Brooktrout Configuration Tool. You can run the Brooktrout Configuration Tool in silent mode for the changes to take effect. You do not need to run it in Advanced mode.

As stated earlier, Microsoft leaves few interfaces for the hardware vendor to implement. [Table 26](#) summarizes the components for these interfaces.

Table 26. Brooktrout Plug and Play Components

Reference	Brooktrout Component
Device	Brooktrout hardware
Function Driver	<i>boston.sys</i>
INF File	<i>trxstream.inf</i>
Property Page Symbol File	<i>bfktBdevpp.pdb</i>
Device Property Page	<i>brktBdevpp.dll</i>
Brooktrout Catalog File	<i>brooktrout.cat</i>
Driver Symbol File	<i>boston.pdb</i>

Structure of the Brooktrout PnP Folder

The Brooktrout PnP folder has the following structure:

PnP

```
| brooktrout.cat
| trxstream.inf
|- TRxStream
|   |- x86
|       | boston.sys
|       | boston.pdb
|       | brktBdevpp.dll
|       | brktBdevpp.pdb
|   |- x64
|       | boston.sys
|       | boston.pdb
|       | brktBdevpp.dll
|       | brktBdevpp.pdb
```

Note: It is very important for this folder structure to be maintained when installed or copied for the Plug and Play to work correctly. If you change the structure, the driver will not install correctly.

About the INF File

Brooktrout-supplied *INF* files (such as *trxstream.inf*) should be available in a location readily accessible to Windows. The *INF* file contains important information for Windows Plug and Play to work. The primary function is to copy over a driver suitable for the device that has been discovered. Windows also offers several extension activities implemented using the *INF* file. These activities can allow the creation of a vendor specific device class, registration of co-installers, and providing user mode services and device property pages. We make use of all these features offered by Windows.

- **Once you identify the *INF* file (specifically *trxstream.inf*), the following actions occur.**

For an authoritative description of the Plug and Play install process always use information from Microsoft (available at MSDN).

1. During the hardware installation, Windows creates a new device under Computer Telephony class. The Device Manager lists all Brooktrout hardware devices under Computer Telephony class/node:

Class = Computer Telephony

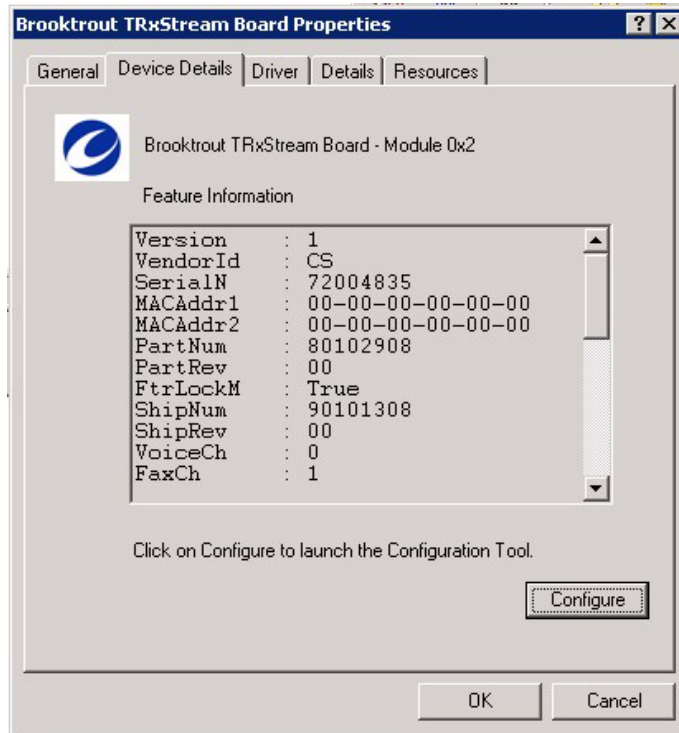
Class GUID = {8CF4CA66-A2CC-48FA-BC1D-6A64E47F6D27}

2. Copies the driver files for the identified device to the *system32\drivers* folder.
3. Copies the device property page to the *system32* folder.
4. Registers the device property page on the host system.
5. Installs the device driver(s) if not already installed.
6. Starts the device driver(s).

About the Device Property Page

The device property pages are dynamic-link libraries (DLLs) that work in conjunction with the Device Manager. They typically display property sheets for examining and modifying the settings and configuration of a device. The Device Manager provides a default property page that gives access to typical device properties. The Device Manager also gives the property page provider an opportunity to add a custom property page. The custom provider must be registered in order for the Device Manager to use it.

The Brooktrout Device Manager Property Page is a custom DLL (*brktBdevpp.dll*) that provides basic integration with Microsoft Management Console. Its primary function displays feature information and its secondary function provides the option to launch the Brooktrout Configuration Tool.



Modifying Configuration Files

This section describes each of the configuration files. You can edit the configuration files with a standard text editor or you can use the Brooktrout Configuration Tool to make changes (see the installation and configuration guide that came with your software).

Assuming the default installation, the configuration files are located in *Brooktrout\Boston\config*. Sample configuration files are located in *Brooktrout\Boston\config\samples.cfg*.

For details about parameters and valid values, see the *Dialogic® Brooktrout® Bfv API Reference Manual, Volume 6, Appendix A, Configuration Files*.

User-Defined Configuration File (*btcall.cfg*)

The user-defined configuration file contains parameters that set values such as specific fax formatting. The Brooktrout SDK supplies a default configuration file named *btcall.cfg*.

If you have a *btcall.cfg* file created for a previous release of the Bfv API, delete the following parameters from the file. These parameters have been removed or moved to another configuration file as indicated:

Parameter	Description
<i>did_digits</i>	This DID digit detection parameter has been modified and moved to the <i>callctrl.cfg</i> file.
<i>did_variable</i>	This DID digit detection parameter has been modified and moved to the <i>callctrl.cfg</i> file.
<i>digital</i>	The call control (<i>callctrl.cfg</i>) configuration file replaces the configuration file defined by the <i>digital</i> parameter.
<i>isdn</i>	The call control (<i>callctrl.cfg</i>) configuration file replaces the configuration file defined by the <i>isdn</i> parameter.
<i>line_encoding</i>	The call control configuration file (<i>callctrl.cfg</i>) that replaces the <i>teleph.cfg</i> file (see Volume 6 of the Dialogic® Brooktrout® Bfv API Reference Manual) does not use this parameter.
<i>nrings</i>	This parameter has been renamed <i>num_rings</i> and moved to the <i>callctrl.cfg</i> file.

Parameter	Description
<i>switch_hook</i>	This parameter has been renamed <i>flash_hook_duration</i> and moved to the <i>callctrl.cfg</i> file.
<i>teleph</i>	The call control (<i>callctrl.cfg</i>) configuration file replaces the configuration file defined by the <i>teleph</i> parameter.

Parameters are listed in any order and typed in either uppercase or lowercase or both. Only one parameter per line is permitted. Parameters must be separated from their values: a decimal integer, a hexadecimal integer, or a character string by one or more spaces. Commas, colons, and dashes are not valid parameter separators. The default value is automatically supplied for each missing parameter; and parameters that do not match any of the valid keywords are ignored. If a parameter appears more than once, the last occurrence is the one that will take effect.

For details about parameters and valid values, see the *Dialogic® Brooktrout® Bfv API Reference Manual, Volume 6, Appendix A, Configuration Files*.

Call Control (*callctrl.cfg*) Configuration File

This file contains configuration parameters that define how you want the Bfv API to configure the modules for call control.

For details about parameters and valid values, see the *Dialogic® Brooktrout® Bfv API Reference Manual, Volume 6, Appendix A, Configuration Files*.

Including the Brooktrout Configuration Tool

When you decide to include the Brooktrout Configuration Tool in your application, you can launch it in different modes depending on your customer needs. The tool provides the following modes:

- Advanced Mode
- Offline Mode
- Silent Mode
- Timer Mode

■ Wizard Mode

Note: The Brooktrout Configuration Tool initializes the default location of the configuration, protocol, and firmware files using the Registry entries created by the installation. See [Registry Entries](#) on [page 372](#). However, if you decide to ship the tool separately, then you need to create a custom *settings.cfg* file that the Brooktrout Configuration Tool can use. Refer to the install and configuration guide for more information.

For details on these modes and running the Brooktrout Configuration Tool in the various modes, see the installation and configuration guide that came with your software).

Downloading Firmware Files

In addition to installing the software, you must:

- Update the boot ROM flash memory
- Download the firmware files
- Optionally configure the call control parameters.

The Configuration Tool can only be used to configure the call control parameters and start the Boston Host Service; this utility cannot be used to update the boot ROM flash memory.

Use the instructions in your installation and configuration guide to manually update the boot ROM flash memory and download the firmware files.

Use the Brooktrout Configuration Tool to configure the call control parameters (see your installation and configuration guide), or manually edit the *callctrl.cfg* file included in your Brooktrout SDK. For details about the call control parameters and valid values, see the *Dialogic® Brooktrout® Bfv API Reference Manual, Volume 6, Appendix A, Configuration Files*.

Removing Software

The Add/Remove Programs selection on the Windows Control Panel screen provides options to initiate the following on the installed components for the package (see *Uninstalling or Modifying the Software* in the installation and configuration guide that came with your software):

Modify – Selecting this option displays a custom setup dialog that allows the user to select the features to be installed or removed. This option is helpful if the user wants to install features that were not selected to be installed during the first installation.

Repair – Selecting this option should do the following:

- Repair all runtime components that are currently installed on the user system
- Re-install some of the files that might have been deleted, renamed, or moved
- Replace missing registry entries and shortcuts.

Note: Windows MSI framework can only detect file corruption if the file is tagged as a KEY file. Only critical files in the installation package are tagged as such. All the runtime binaries that are installed by Brooktrout SDK and are tagged as KEY files.

Remove – This option should remove the package, including all the files that were copied over by the installation. This option only deletes the files that were created or copied by the installation. It should remove *bostsrv.exe* service and delete all registry entries created for this service. It should also remove all the registry entries that were created by applications such as the Brooktrout Configuration Tool.

On Plug and Play operating systems, you must completely remove the Plug and Play driver by following the instructions in [Removing the Plug and Play Driver](#) on [page 402](#).

Note: The group responsible for maintaining the install project *MUST* be notified of any registry entries, files, folders, services and shortcuts that should be removed through the *Add/Remove Programs* process.

Removing the Plug and Play Driver

Installing a Brooktrout device using the Plug and Play Manager (Found New Hardware Wizard) creates some backup files and registry entries. The Windows Plug and Play Manager uses this information to automatically install the device on rebooting.

To completely remove the Plug and Play driver from the system, you must perform a complete cleanup after you remove the device from the Device Manager.



Make sure you stop all your applications and the Boston Host Service before uninstalling the device driver.

➤ **To remove the Plug and Play driver:**

1. Open Windows Device Manager.
2. Expand the Computer Telephony node.
3. Right-click the **Brooktrout TRxStream** board node and select **Uninstall**.
4. Uninstall all the Brooktrout boards listed under the Computer Telephony node.
5. Open Command Prompt and type "**net stop boston**" to stop the driver that might still be running.
6. Delete the following files:

`C:\WINNT\SYSTEM32\DRIVERS\boston.sys`

`C:\WINNT\SYSTEM32\brktBdevpp.dll`

7. Examine the registry value **InfPath** located under `HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet\Control\Class\{8CF4CA66-A2CC-48FA-BC1D-6A64E47F6D27}`.
8. Search for the first key which contains the following values:
`DriverDesc=Brooktrout TRxStream Board`
This key should contain the name of the backup copy of the *trxstream.inf* file that Windows created during the device installation under `C:\winnt\INF`
9. Delete the *INF* file of this name from `C:\WINNT\INF` along with the corresponding *PNF* file.

Except for the extension, the *INF* file and the *PNF* file have the same names. For example, if the *INF* file is *oem11.inf*, the name of the *PNF* file is *oem11.pnf*.

10. Delete the following registry keys:

For 32-bit operating system:

```
HKEY_LOCAL_MACHINE\SOFTWARE\Brooktrout Technology  
  \DevicePropertyPage
```

```
HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet  
  \Services\boston
```

For 64-bit operating system:

```
HKEY_LOCAL_MACHINE\SOFTWARE\WOW6432Node  
  \Brooktrout Technology \DevicePropertyPage
```

```
HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet  
  \Services\boston
```

11. Reboot the system and verify that the Brooktrout TRxStream node(s) is not listed under Computer Telephony class node in the Device Manager.

Appendix A

G3 Legacy Utilities

This appendix describes legacy utilities that help manipulate raw G3 fax files.

The utilities described in this chapter permit you to manipulate raw G3 fax files (rather than infopkt-formatted files) from the command line. These are legacy utilities that are provided to users who need to manipulate raw G3 files. They have been tested with previous versions of the Bfv API and are expected to work, but they are no longer tested/supported.

Output files will usually contain a 128-byte Brooktrout header. When you create an infopkt file using one of these output files, you must either use the BTG3 type infopkt or remove the header.

For input files, the 128-byte Brooktrout header is optional. However, a header can provide crucial information to the utilities about resolution and image width. This information is particularly important to the Print utility.

The G3 utilities only support the resolutions 200H x 100V (Normal) and 200H x 200V (Fine). They do not support any of the higher resolutions.

All utilities except for the G3 conversion utility use file names with the .3XX extension. The first page of a fax has the extension .301, the second .302, etc. Many of these utilities will ignore actual extensions supplied and add or convert to these extensions.

	The G3 utilities include:
ASCII to Fax Conversion Utility	Converts ASCII text files to Brooktrout G3 fax image files.
Cut and Paste Utilities	Used in conjunction, removes parts of fax images and stores them in a separate file and recombines them into another file.
Epson to Fax Conversion Utility	Converts Epson print files to Brooktrout G3 fax image files.
Fax Display and Edit Utility	Displays fax files and provides a visual interface to the cut utility.
G3 Conversion Utility	Converts raw fax files between any of these formats – MH, MR, MMR, PCX, and bitmap.
Print Utility	Prints fax files to any of several printers.
	An example of how to use each utility and its parameters is included with each description.
	All utility executable programs are found in the <i>Brooktrout/Boston/bfv.api/<platform>/utils/bin</i> .
	Note: For the operating system you are using, substitute its name for <i><platform></i> in the following instructions and use the appropriate location where the installed files are located (that is, <i>/usr/sys</i> for Linux; and <i>C:</i> for Windows).

ASCII to Fax Conversion Utility (asctog3)

Converts ASCII text files to Brooktrout Group 3 fax image files. Although this utility is included, in most cases ASCII file transmission is better accomplished utilizing the ability of the Brooktrout fax modules to convert ASCII to G3 on-the-fly. This is a legacy utility.

Command Syntax

asctog3 argument-list

Where:

- i (input file) Specifies the input file to convert to Group 3.
- o (output file) Specifies the output file name. If omitted, an output file is created with the same input file name but is given a .3XX file extension.
- mfine (fine resolution) Specifies fine (200 vertical lines/inch) resolution. The default is normal resolution (100 lines/inch).
- margin# (margin) Adds a left margin specified in tenths of an inch (#). Width is optional and defaults to either 0.5 inch, if the -margin option is specified, or to 0 if the -margin option is not specified.
- f (font file) Species the font file to use for the conversion.
- lines# (page length) Specifies the page length in terms of text lines. Defaults to 66 lines/page for standard 11" paper.
- nopad (no pad) Prevents padding short pages to standard 11" page length. The default pads short pages to 11" length, regardless of the input file.
- skip# (letterhead) Makes the first page of output shorter by # text lines. Leaves room for a letterhead.
- w# Makes lines of the specified width: 0 = A4, 1 = B4, 2 = A3.

The ASCII to Fax Utility displays help information on ASCII to Group 3 file conversion when no arguments are specified. For example:

asctog3 -imemo.txt -fibmpcps.fnt

converts a text file named *memo.txt* to a Group 3 fax file using a standard 80 column font.

Cut and Paste Utilities

The next two utility commands, *g3chop* and *g3combin*, permit cutting and pasting of fax images at the command line. These two commands are most commonly used to create letterhead and signature files.

Cut Utility (g3chop)

Removes portions of Brooktrout Group 3 fax files and stores them in a separate file.

Command Syntax

```
g3chop -sx -cy -ifile1.301 -ofile2.301
```

Where:

- s (start chop) Specifies the distance in 1/10" units, from the top of the file, to skip before chopping.
- c (chop size) Specifies the distance in 1/10" units to chop.
- ifile1.301 (input file) Specifies the name of the file to chop.
- ofile2.301 (output file) Specifies the name of the output file which contains the resulting chopped file.
- l (lines) Changes the **-s** and **-c** arguments to work with G3 lines as the units.

The number of lines are calculated in inches using the file resolution if you know how far down in inches you want to start and you use the **-l** argument: to skip 2 inches from the top of the file in normal resolution, skip 200 lines; for fine resolution, skip 400 lines.

The number of lines to chop can also be calculated based on the file resolution and the number of inches to remove from the file.

For example:

```
g3chop -s5 -c20 -isalute.301 -oadvert.301
```

skips the first five lines of the file *salute.301*, chops the next 20 lines, and puts them into the output file *advert.301*.

Paste Utility (g3combin)

Combines portions of Brooktrout Group 3 fax files, and, although it is most commonly used to create letterhead and signature files, it can combine all types of fax files. Although this utility is included, in most cases combining fax files is better accomplished using infopkt files.

To attach a letterhead, specify the letterhead file as *file1*. To attach a signature file, specify the signature file as *file2*. If the resolution of the two files is different, you can specify which resolution to use for the output file.

Command Syntax

```
g3combin <file1.301> <file2.301> <file3.301> [-l] [-s] [-pad]
```

Where:

- l** (*file2* resolution) Specifies the resolution of *file2* for the resolution of the output file when the resolutions of the input files are different.
- s** (*file1* resolution) Specifies the resolution of *file1* as the resolution of the output file when the resolutions of the input files are different.
- file1.301** (top input file) Specifies the name of the file to place on top of the combined document.
- file2.301** (bottom input file) Specifies the name of the file to place on the bottom of the combined document.
- file3.301** The output file containing the combined contents of *file1.301* and *file2.301*.
- pad** (pad short pages) Pads short pages to a standard 11" page length.

For example:

```
g3combin -l lethd.301 sign.301 busrpt.301
```

combines the contents of *lethd.301* with the contents of *sign.301* into the output file *busrpt.301*. The contents of *lethd.301* (letterhead) is placed at the top of the output file, and *sign.301* (signature) is placed at the bottom of the output file. The resolution of *sign.301* is assigned to the output file *busrpt.301*.

Epson to Fax Conversion Utility (epstog3)

Converts Epson print files to Brooktrout Group 3 fax images and provides complete emulation of the Epson-LX80 printer (including bold, italics and graphics).

Command Syntax

epstog3 argument-list

Where:

-i	(input file) Specifies the input printer file (Epson-format) and optional path name.
-o	(output file) Specifies the output fax image file name and optional pathname.
-mfine	(fine resolution) Specifies fine (200 vertical lines/inch) resolution. The default is normal resolution (100 vertical lines/inch).
-132	(132 column) Scales printer files that contain wide (132 column) documents, spreadsheets, or multicolumn word processed documents so they fit on standard 8.5" x 11" paper.
-margin#	(margin) Adds a left margin specified in tenths of an inch. Width is optional and defaults to either 0.5 inch, if the -margin option is specified, or to 0 if -margin option is not specified.
-fibm	(emulation) Changes the default printer emulation from an Epson to an IBM graphics printer. This, in effect, changes the font used when converting text data.
-lines#	(page length) Specifies the page length in terms of text lines. Defaults to 66 lines/page for standard 11" paper.
-nopad	(no pad) Prevents padding short pages to a standard 11" page length. The default pads pages to an 11" page length, regardless of the length of the input file.
-skip#	(letterhead) Makes the first page of output shorter by # text lines. Leaves room for a letterhead.

The *epstog3* utility displays help information on converting graphics files to fax image files, on the command line.

Note: Fonts must be located in the current directory for all platforms except those platforms that use MS-DOS executables (Windows). For these cases, the fonts must be located either in the *font* subdirectory of the directory named by the *FMAIL* environment variable, if it is set, or in the *boston\bfv.api\fonts* directory.

For example:

epstog3 -icap001.epc

converts a graphics file named *cap001.epc* to a Group 3 fax image file.

If the output file name is not specified, it is automatically created using the input file name and given an extension of *.3XX*.

If the input file does not reside in the current or default directory, the path name must also be specified.

If the path name of the output file is not specified, the output file is placed in the same directory as the input file.

Fax Display and Edit Utility: Supershow (ss)

Displays a fax file on screen for viewing and editing. Editing options include scaling, rotating, and reversing video.

Note: *Supershow* does not scale images automatically to maintain aspect ratios or to fit images on the screen. Each pixel in the file corresponds to one pixel on the screen. *Supershow* does support manual scaling with the commands described on the help screen.

To see a complete list of arguments and display options, type `ss` without any arguments at the system prompt. To access the *Supershow* help screen, which lists movement and display commands, press the `?` key during a fax display.

Supershow automatically determines the adapter type, so the `-h` and other display-related arguments are optional on all environments. Use these arguments to select a display mode that differs from the default for the current adapter type.

All arguments are optional, except `-i`.

Command Syntax

`ss [-r] [h|c|e|v|ea|va] [-xs#] [-ys#] [-x#] [-y#] [-w#] [-m#] -ifilename`

Where:

<code>-r</code>	Displays the image in reverse video. You can combine this argument with one of the other arguments: <code>h</code> , <code>c</code> , <code>e</code> , <code>v</code> , <code>ea</code> , or <code>va</code> .
<code>-h</code>	Forces the display type to Hercules graphics.
<code>-c</code>	Forces the display type to IBM CGA.
<code>-e</code>	Forces the display type to IBM EGA.
<code>-v</code>	Forces the display type to IBM VGA.
<code>-ea</code>	Forces the display type to IBM EGA alternate mode.
<code>-va</code>	Forces the display type to IBM VGA alternate mode.
<code>-ifilename</code>	Specifies the name of the fax file to display.
<code>-x</code>	Specifies the X-offset. Units are in tenths of an inch.
<code>-y</code>	Specifies the Y-offset. Units are in tenths of an inch.
<code>-xs</code>	Specifies the X scale factor. Values are 1, 2, or 3.

<code>-ys</code>	Specifies the Y scale factor. Values are 1, 2, or 3.
<code>-w</code>	Specifies the width (0 = A4, 1 = B4, 2 = A3) and overrides the header.
<code>-m</code>	Specifies the number of kilobytes of memory to use for an image. The default is 512K. Using large values might slow response times. (This argument is not available for Windows.)

For example:

`ss -rv -iletsig.301`

displays a file named *letsig.301* on the screen in reverse video on a PC equipped with a VGA adapter.

G3 Conversion Utility (**g3cvt**)

Converts a raw fax file between any of these formats—MH, MR, MMR, PCX, and bitmap. It accepts files that use either MSB or LSB bit order, and (optionally) skips past a Brooktrout 128-byte header. Output from this utility does not contain a 128-byte Brooktrout header.

As it proceeds, *g3cvt* displays dots across the screen. If it detects an error in the input file, *g3cvt* displays an error message. With some input formats, *g3cvt* cannot continue after it encounters an input error.

For bitmap conversion, this utility uses raw bitmap format. The `-w` option determines the number of bits per line, and the `-b` option determines the bit order. Raw bitmap format does not use start-of-image, end-of-line, or end-of-image marks.

When decoding MH or MR files, if this utility encounters an input line with errors, it writes a copy of the previous line in the output file.

To display usage information, type **g3cvt** without any arguments.

Command Syntax

g3cvt `<i_fmt>` `<i_file>` `<o_fmt>` `<o_file>` [`options`]

Where:

<code>i_fmt</code>	Format of the input file—MH, MR, MMR, PCX, or BIT.
<code>i_file</code>	Name of the input file.
<code>o_fmt</code>	Format of the output file—MH, MR, MMR, PCX, or BIT.
<code>o_file</code>	Name of the output file.

Options

<code>-w<width></code>	Specifies the width of the lines; A4 (default), B4, or A3.
<code>-b<b_o></code>	Specifies the bit order of both the input and output files – M (MSB), the default, or L (LSB).

-
- | | |
|----|---|
| -h | Causes <i>g3cvt</i> to copy the first 128 bytes of the input file to the output file, with no conversion. |
| -f | Specifies fine resolution input. Used with MR or PCX output only.

MR output—The program represents the data slightly differently and more efficiently when it knows the input is in fine resolution.

PCX output—Since PCX format is always in fine resolution, the program must know whether the input file is in fine or normal resolution. If the input is in normal resolution, the program doubles each line in the output file.

A PCX input file automatically activates the -f option. |

For example:

g3cvt MMR fax.mmr MH fax.mh

converts a fax file from MMR format to MH format.

Print Utility (p)

Prints the fax file to any one of several printers.

Command Syntax

p **argument-list**

```
p [-ppmode] [-xXOFF] [-yYOFF] [-xsXSCALE] [-ysYSCALE] -ifiles [-ooutfile]
[-paper[A4/LET]] [-w[width]]
```

Where:

-ppmode	(print mode) Indicates the printer and resolution to use for printing; where pmode is the character string that defines the print mode and is one of the following:
psd	Single density: 60 dpi
plsd	Low speed double density: 120 dpi
phsd	High speed double density: 120 dpi
pqd	Quadruple density: 240 dpi
pqds	Low speed quadruple density: 240 dpi
peps	Epson QX-10 mode: 80 dpi
poto	One-to-one plotter: 72 dpi
pd90	90 dpi
pibmprosd	IBM Proprinter single density
pibmprolsd	IBM Proprinter low speed double density
pibmprohsd	IBM Proprinter speed double density
pibmproqd	IBM Proprinter quadruple density
pljl	HP Laser Jet: 75 dpi
pljm	HP Laser Jet: 100 dpi
pljh	HP Laser Jet: 300 dpi
plj2l	HP Laser Jet II: 75 dpi
plj2m	HP Laser Jet II: 100 dpi
plj2h	HP Laser Jet II: 300 dpi
pnec	NEC P6 24-Nadel Drucker: 360 dpi
pnec0	NEC P6 24-Nadel Drucker: 180 dpi
pfx2000	Axiom Edwards FX-2000: 200 dpi
-xXOFF	(x offset) Selects an X direction offset in tenths of inches. Only the portion of the image to the right of this position is displayed.

<code>-yYOFF</code>	(y offset) Selects a Y direction offset in tenths of inches. Only the portion of the image below this position is displayed.
<code>-xsXSCALE</code>	(x scale factor) Selects an X direction scale factor in 5% increments. The image is scaled to this factor of its original size in the X direction.
<code>-ysYSCALE</code>	(y scale factor) Selects a Y direction scale factor in 5% increments. The image is scaled to this factor of its original size in the Y direction.
<code>-ifiles</code>	(input file name) Specifies the Group 3 files to print. The file name and pages to print are specified: file name (first page, last page).
<code>-ooutfile</code>	(output file name) Directs output to the specified file rather than the printer.
<code>-paper</code>	(paper type) Specifies the paper size to use.
<code>-w[width]</code>	Scales horizontally and vertically to fit an image of the specified width: 0 = A4, 1 = B4, 2 = A3. If no width is specified, this utility uses the width from the 128-byte Brooktrout header.
A4	Specifies an A4 feeder and A4 size paper.
LET	8.5" x 11" standard size paper.

The Print Utility displays help information when no arguments are specified.

Note: The Print utility automatically scales the image to maintain the correct size and aspect ratio for each supported printer type and resolution. If the Print utility encounters a 128-byte header at the beginning of the input file, it uses the actual resolution of the fax for scaling; otherwise, it uses normal resolution for scaling.

Note: When using widths larger than the standard A4, you do not need to include the `-w` argument. However, because these larger widths exceed printers' standard width, the Print utility truncates the image horizontally to fit the standard width. If you do include the `-w` argument, the Print utility reduces the size of the image to fit the standard width.

For example:

p -phsd -xs10 -ys10 -idemo.301

prints a file, *demo.301*, on an Epson/IBM compatible printer in high speed, double density mode scaling the image in both the x and y directions to one-half size.

Appendix B

Recompiling On Linux Platforms

Refer to “Recompiling on Linux Platforms” in the *Dialogic® Brooktrout® Fax Products SDK Installation and Configuration Guide*.

Glossary

API	Application Programming Interface
ASR	Automatic Speech Recognition
Channel	A logical channel of operations provided by a Boston module. See <i>logical channel number</i> , <i>ordinal channel number</i> , <i>work channel</i> .
EC	Echo cancellation
External-Telephony Mode	Using the Bfv API directly instead of a speech engine API to perform call control (Mixed Mode)
Facility	A software entity responsible for a set of related functions that provide services to the host, e.g., fax facility and voice facility.
HDLC	High level Data Link Control
ISDN	Integrated Services Digital Network
IVR	Interactive Voice Response

lapdid	The term <i>lapdid</i> has its origins with the LAP-D protocol used for call control, but has an extended meaning for Brooktrout products. For call control for all protocols, trunk 1 = lapdid 0, trunk 2 = lapdid 2, trunk 3 = lapdid 4 and trunk 4 = lapdid 6.
LEC	Local Exchange Carrier
Line	A T1/E1 slot or a single analog slot. Lines are numbered starting at 1.
Logical channel number	A number used with the hardware module number to reference a channel in a system. Channel 0 is reserved and channel 1 is the administrative channel. Other channels are numbered from 2 to n+1 (where n is the number of work channels defined for the module).
Millennium Address	An address of a communicating Boston or Millennium entity such as a facility and channel on a Boston module or the Boston device driver. Each address consists of 4 components, the facility, the channel, the module, and the machine. It is possible to make use of most features of the firmware and the APIs without requiring direct use of these addresses.
Module	A communicating Boston entity that usually represents a CPU on a Brooktrout board, a host application, or the driver. See section TBS for more information about modules and module number assignment.
NVRAM	Non-Volatile RAM. Random Access Memory that is not erased when the board is powered off or when the firmware is downloaded again.
Ordinal channel number	A number in the range 0 to n-1 where n is the total number of work channels on all the boards in the system.
Packet	In the Boston architecture, a sequence of bytes containing a Boston destination address, source address, and one or more commands.
PCI	Peripheral Component Interconnect
PDF	Portable Document Format
Port	A TDM bus (MVIP, H110, etc.) or a DS-1 interface (T1/E1).
RBS	Robbed-Bit Signaling

Stream	A logical data entity that corresponds to a physical data line on a TDM bus.
T1/E1 span	The set of slots that comprises one T1 (24) or one E1 (30) line. The spans are numbered starting at 1.
TCP	Transmission Control Protocol
Time slot	A logical entity that corresponds to one telephone call.
Unit number	1. In telephony configuration files and functions, a unit is a hardware port on a Brooktrout board and is numbered starting at 0 (to denote a TDM bus); from 1 to n (for a specific T1/E1 interface). 2. In some Bfv functions (e.g., <i>BfvLineAttach</i>), an ordinal channel number. Its range is 0...n-1, where n is the number of channels in a system.
VAD	Voice Activity Detection
Work channel	One of the channels on a module that is available for non-administrative purposes as a result of downloading firmware. A module configured for 48 channels when firmware download is performed has 48 work channels and one administrative channel. Logical work channel numbers start at 2 on each module.

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