



## **VoIP Interfaces**

**bintec-Dm 770-I**

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## I Related Documents

bintec-Dm 704-I Configuration and Monitoring

bintec-Dm 722-I Telephony over IP

bintec-Dm 732-I Dial Profile

bintec-Dm 746-I G703 Inteface

bintec-Dm 754-I NSLA

bintec-Dm 772-I Common Configurations for Interfaces

bintec-Dm 781-I Cellular Interface

# Chapter 1 Introduction

## 1.1 Introduction

VoIP cards allow voice to be exchanged between the PSTN network and an IP network. Several PSTN interfaces are available: FXS/FXO for conventional analog telephony, ISDN BRI in both user (TE) and network (NT) mode, E1 interfaces with ISDN PRI or R2 signaling, and two- or four-wire E&M analog interfaces to interconnect with some switchboards that support this interface.

In analog telephone interfaces, the VoIP card carries out the Analog-Digital and Digital-Analog (A/D and D/A) conversion, while in digital interfaces, the voice is received in digital format encoded in G.711 A-law or U-law. On the IP side, the voice can arrive encoded in various codecs and encapsulated in Real Time Protocol (RTP) over UDP.

VoIP cards have one or more Digital Signal Processors (DSP) dedicated exclusively to the task of changing the format between the traditional world and the IP world. Each card has the necessary processing resources to simultaneously support all the available functionality on all the voice channels present on the card. The DSP functions are as follows:

- Encoding/decoding: the supported codecs include G.711 A-law and U-law, G.729 with Annexes A and B, and G.723 at speeds of 6.4 and 5.3 Kbps.
- Silence detection (voice activity detection or VAD).
- Echo cancellation in the switched interface.
- Generation/detection of multifrequency signaling (DTMF).
- FAX and T.38 detection.

The G.729 and G.723 codecs reduce the amount of bandwidth taken up by voice at the expense of a slight decline in audio quality and offer protection against packet loss through redundancy and interpolation techniques. To the voice samples we must add the LAYER2/IP/UDP/RTP header overhead, which is fairly significant, so to reduce the bandwidth, several voice frames can be grouped per packet and header compression techniques used.

The packets received through IP are classified and put into an adaptive delay buffer, allowing any jitter in the IP network to be eliminated and the voice samples to be submitted to the switch interface at the speed they are generated.

## Chapter 2 VoIP Interfaces Configuration

### 2.1 VoIP Interfaces Configuration Commands

To enter the Voice over IP (VoIP) interface configuration, access from the main menu as follows:

- (1) At the star (\*) prompt, enter PROCESS 4 (or P 4).
- (2) At the configuration prompt (Config>), type network voipx/y where x and y are the PCI slot position.  
In the case of VoIP-ISDN interfaces created through the ADD DEVICE VoIP-ISDN command, type voipz where "z" is the number used to create it.
- (3) At the interface configuration prompt (voipx/y config>), use the configuration commands described in this chapter to configure the aforementioned interface parameters.

This chapter lists and describes the configuration commands of a VoIP interface. These commands are to be entered at the interface configuration prompt (voipx/y config>).

Some of the commands are common to all the device's interfaces. You can find these commands described in the following manual: bintec-Dm 772-I Common Configurations for Interfaces.

#### 2.1.1 BASE-INTERFACE

This command is only available for ISDN interfaces. It allows you to configure the ISDN physical interface that executes the relevant calls and their related aspects. When it comes to VoIP interfaces that have a fixed wired base interface (e.g., VoIP ISDN boards), this interface cannot be modified or deleted. This command operates similarly to other types of DIAL interfaces (such as PPP).

*Syntax:*

```
voip3/0 config>BASE-INTERFACE

-- Base Interface Configuration --
voip3/0 Base IFC config>
```

The available options are:

```
voip3/0 Base IFC config>?
  base-interface    Specify a base interface
  list              List current configuration
  no                Negates a command or sets its defaults
  exit
voip3/0 Base IFC config>
```

##### 2.1.1.1 [NO]BASE-INTERFACE

This command allows you to associate a particular base interface with the VoIP interface you are configuring, as well as some DIAL profile parameters (call) and the maximum number of calls that can be carried out over this base interface.

To execute and receive calls, you need to specify the name of the DIAL profile that contains the call parameters (for further information on configuring Call Profiles, please see manual bintec-Dm 732-I Dial Profile). This profile must be audio.

In base interfaces, since the circuit identifier (id circuit) must be 255, you cannot specify the requesting channel.

For VoIP interfaces, configuring a base interface to execute audio calls is the only thing that makes sense.

*Syntax:*

```
voip3/0 Base IFC config>BASE-INTERFACE <isdn interface> <id circuit> [options]
link                add this interface to the dial group
number-of-circuits  number of circuits to request
profile            dial profile to use with this interface
```

<i>link</i>	Adds the base interface to the VoIP interface
<i>number-of-circuits</i>	Maximum number of circuits in the base interface that can be used to execute calls. That is, the maximum number of calls that can be made over the base inter-

	<p>face.</p> <p>By default, this is configured as ALL in interfaces with an ISDN physical interface, i.e., everything that the base interface provides is used. In interfaces without an associated DSP, this value is not configured.</p>
<i>profile</i>	<p>Name of the call profile used in the base interface. This must be audio. If this is not configured, event VoIP.002 is displayed and calls cannot be executed or received.</p> <p>The default is no profile configured.</p>

### 2.1.1.2 LIST

Displays a table showing all the configured base interfaces. An example is given below:

```
voip3/0 Base IFC config>LIST
```

Base Interface	Profile Name	Base Circuit Id	Number of circuits
g703port3/0	VoIP_Calls	255	all

```
voip3/0 Base IFC config>
```

The above corresponds to configuration:

```
voip3/0 Base IFC config>show conf
base-interface g703port3/0 255 link
base-interface g703port3/0 255 profile VoIP_Calls
base-interface g703port3/0 255 number-of-circuits all
```

### 2.1.1.3 NO

Deletes the configured base interface. Base interfaces in VoIP ISDN boards cannot be deleted.

## 2.1.2 [NO] CHANNELS

This command is only available on dummy virtual interfaces. It specifies the number of channels available, i.e., the maximum number of simultaneous calls that can be established through the interface. Unless it is configured, the default value is 1.

*Syntax:*

```
voip1000 config> channels ?
<1..30> Value in the specified range
```

## 2.1.3 [NO] DSP-FIRMWARE

Simultaneous use of G.723 and G.729 codecs is not allowed in some VoIP interfaces. This command specifies which firmware should be loaded in the voice DSP on startup, i.e., whether it should include G.723 or G.729. This is only available for interfaces that require this.

*Syntax:*

```
voip1/0 config>dsp-firmware ?
g723-g711 Load G723 and G711 firmware
g729-g711 Load G729 and G711 firmware
```

## 2.1.4 ISDN

This command configures parameters related to the ISDN interfaces and is therefore only available for ISDN interfaces.

### 2.1.4.1 [NO] ISDN BEARER-CAP 3100HZ

When routing H323 calls over an ISDN interface, the information transfer capacity of the "carrier capacity" element becomes 3100 Hz, instead of that received through the H.323 SETUP message.



By default, this option is not configured. Consequently, the received capacity is transmitted over the ISDN interface.

**Syntax:**

```
voip100 config>ISDN BEARER-CAP 3100HZ
```

#### 2.1.4.2 [NO] ISDN BEARER-CAP SPEECH

When routing H323 calls over an ISDN interface, the information transfer capacity of the “carrier capacity” element matches that of SPEECH, instead of that received through the H.323 SETUP message.

By default, this option is not configured. Consequently, the received capacity is transmitted over the ISDN interface.

**Syntax:**

```
voip100 config>ISDN BEARER-CAP SPEECH
```

#### 2.1.4.3 [NO] ISDN CALLED NUMBER-TYPE

Specifies the type of called number for an incoming or outgoing call over this interface.

By default, this option is not configured. Consequently, the type of received numeration is transmitted over the ISDN interface.

**Syntax:**

```
voip100 config>ISDN CALLED NUMBER-TYPE ?
unknown          Unknown q931 number type (0)
international    International q931 number type (1)
national         National q931 number type (2)
network         Network q931 number type (3)
subscriber       Subscriber q931 number type (4)
abbreviated      Abbreviated q931 number type (5)
reserved         Reserved q931 number type (6)
any              Any q931 number type (7)
voip100 config>
```

#### 2.1.4.4 [NO] ISDN CALLED PLAN-TYPE

Specifies the type of numeration plan the called number pertains to for an incoming or outgoing call over this interface.

By default, this option is not configured. Consequently, the received numeration plan is transmitted over the ISDN interface.

**Syntax:**

```
voip100 config>ISDN CALLED PLAN-TYPE ?
unknown          Unknown plan type (0)
isdn/telephony  Isdn/telephony plan type (1)
data            Data plan type (3)
telex           Telex plan type (4)
national        National plan type (8)
private         Private plan type (9)
reserved        Reserved plan type (15)
voip100 config>
```

#### 2.1.4.5 [NO] ISDN CALLER NUMBER-TYPE

Specifies the type of caller number for an incoming or outgoing call over this interface.

By default, this option is not configured. Consequently, the type of received numeration is transmitted over the ISDN interface.

**Syntax:**

```
voip100 config>ISDN CALLER NUMBER-TYPE ?
unknown          Unknown q931 number type (0)
international    International q931 number type (1)
national         National q931 number type (2)
network         Network q931 number type (3)
```

```

subscriber      Subscriber q931 number type (4)
abbreviated     Abbreviated q931 number type (5)
reserved       Reserved q931 number type (6)
any            Any q931 number type (7)
voip100 config>

```

### 2.1.4.6 [NO] ISDN CALLER PLAN-TYPE

Specifies the type of numeration plan the caller number pertains to for an incoming or outgoing call over this interface.

By default, this option is not configured. Consequently, the received numeration plan is transmitted over the ISDN interface.

#### Syntax:

```

voip100 config>ISDN CALLER PLAN-TYPE ?
unknown          Unknown plan type (0)
isdn/telephony   Isdn/telephony plan type (1)
data             Data plan type (3)
telex            Telex plan type (4)
national         National plan type (8)
private          Private plan type (9)
reserved         Reserved plan type (15)
voip100 config>

```

## 2.1.5 LINE

Allows you to independently configure parameters per line. The number of lines that a VoIP interface contains depends on the type of hardware.

In cases of VoIP E1 or BRI interfaces, the number of available interfaces for configuration purposes is 1 so the configuration for this line is applied over all the calls that can be carried out. The number of calls to execute depends on the minimum value of the switched channels the base interface supports and the number of circuits configured under BASE-INTERFACE in the VoIP interface.

#### Syntax:

```

voip1/0 config>LINE <line-number>
active          Line can receive and make calls

alert voice-message  Substitute the alert tone with a voice message

auto-answer      Answer an fxo line before the call being connected

behavior        voip-dummy audio behavior
  play-loop      play an audio message in a loop
  play-disc      play an audio message and disconnect
  play-fwd       play an audio message and forward
  <0-9 # * + , . T or P allowed> Telephone number
  play-ivr       play an audio message and wait for DTMF
  error-fwd      number to forward on error
  <0-9 # * + , . T or P allowed> Telephone number
  timeout-fwd    number to forward on timeout
  <0-9 # * + , . T or P allowed> Telephone number
  translation    apply a translation to dialed number
  <1..10000>     Value in the specified range

busy voice-message  Substitute the busy tone with a voice message

caller-id        Set the caller id generation standard
  bellcore      Bellcore caller id
  etsi          ETSI caller id
  none          No caller id generation
  gain         caller id detection gain

default          Default line configuration

dial             Dial configuration options

```

```

    level Configure dial tone output level
pulse-detect Use pulse dialing instead of dtmf
voice-message Substitute the dial tone with a voice message

direct-dialing          Call is automatically carried out to this number

echo-cancellation      Echo cancellation is done in the line
  suppression           Suppression factor, default 5
  dtd-threshold         Double talk detector threshold, default 50
  min-loss              Min echo return loss, default 6dB
    0dB                 0dB min echo return loss
    3dB                 3dB min echo return loss
    6dB                 6dB min echo return loss
    9dB                 9dB min echo return loss
  nlp                   Non linear Processor configuration
    adaptive            Default Adaptive operation
    disable             Disable NLP
    mute                Mute background noise

error voice-message    Substitute the error tone with a voice message

e&m-idle                E&M digital idle ABCD signal
  <hex 1..f>           Hexadecimal value in the specified range

e&m-lead-relay         E&M signals relay mode
  all                   all signals are relayed
  none                  no signals are relayed

e&m-mask               E&M mask to apply to Rx ABCD signal
  <hex 1..f>           Hexadecimal value in the specified range

e&m-operation          E&M wiring mode
  2-wires               audio over 2 wires
  4-wires               audio over 4 wires

e&m-seize              E&M digital idle ABCD signal
  <hex 1..f>           Hexadecimal value in the specified range

e&m-signal             E&M signaling modes
  delay-dial            delay-dial
  immediate             immediate
  wink-start            wink-start
  always-on             no signaling, call is always established

e&m-type               E&M working types
  type-i                Type I
  type-ii               Type II
  type-iii              Type III
  type-v                Type V

fork-call              Enable fork call

forward-digits         Send called number as DTMF

forward-no-answer      Telephone to forward the call if the callee doesnt answer

framer                Framer configuration menu
  external-loopback     Set an external loopback at the framer
  internal-loopback     Set an internal loopback at the framer
  input-gain            Set the input gain of the framer
  output-loss           Set the output loss of the framer
  GIN                   1 disable input attenuator
  AX                    1 enable 6 dB input gain
  AR                    1 enable 6 dB output loss
  LRG                   1 enable 6 dB output loss
  X-filter              Configure X filter

```

hold voice-message	Hold voice message
impedance	Impedance
high	High impedance
low	Low impedance
interface-type	Indicate the line type fxs/fxo
fxs	line is connected to a telephone (FXS, Foreign eXchange Station)
fxo	line is connected to a trunk line (FXO, Foreign exchange Office)
ip-calls	Permit making calls over IP
loc-level	Indicate the line output gain to play voice messages
mic-gain	Indicate the line input gain
no	
active	line can receive and make calls
alert	Substitute the alert tone with a voice message
auto-answer	Answer an fxo line before the call being connected
behavior	voip-dummy audio behavior
busy	Substitute the busy tone with a voice message
caller-id	Set the caller id generation standard
dial	Dial configuration options
level	Configure dial tone output level
pulse-detect	Use pulse dialing instead of dtmf
voice-message	Substitute the dial tone with a voice message
direct-dialing	Call is automatically carried out to this number
echo-cancellation	Echo cancellation is done in the line
suppression	Suppression factor, default 5
dtd-threshold	Double talk detector threshold, default 1
error	Substitute the error tone with a voice message
e&m-idle	E&M digital idle ABCD signal
e&m-mask	E&M mask to apply to Rx ABCD signal
e&m-seize	E&M digital seize ABCD signal
fork-call	Enable fork call
forward-digits	Send called number as DTMF
forward-no-answer	Telephone to forward the call if the callee doesn't answer
framer	Framer configuration menu
hold	Hold voice message
ip-calls	Permit making calls over IP
pcm-loop	Loop internal calls at the pcm bus
start-early-media	Start sending Early Media as soon as possible
suspend-mode	lines in FXS mode permit suspended behavior mode
supervised	Supervise FXO line availability
telephone-number	Configures the telephone number associated to the line
track	Track this line
pcm-loop	Loop internal calls at the pcm bus
play	Play a voice-message in a dummy ifc
G711 voice-message	voice-message coded as G.711 samples
G723 voice-message	voice-message coded as G.723 frames
G729 voice-message	voice-message coded as G.729 frames
speaker-gain	Indicate the line output gain
start-early-media	Start sending Early Media as soon as possible
supervised	Supervise FXO line availability
suspend-mode	Suspend call when hanging FXS line
telephone-number	Configure the telephone number associated to the line

timing	Configure timings parameters
delay-duration	Max delay-signal duration (in milliseconds)
delay-start	Time to wait (in milliseconds) before dialing digits in delay-dial mode
dialout-delay	Time to wait before dialing digits
forward-no-answer	Time to wait before forwarding the call if no answer
fxo-answer	Minimum ringing time (in milliseconds) before an fxo line hook-off
hook-flash	Configure hook-flash timings
initial	Max initial time before receiving dialed digits
interdigit-timeout	Time to wait before starting a call between two dtmf
off-ringing-duration	Interval (in milliseconds) for the ring signal being inactive
playout-delay	Time to wait before dialing digits
ringing-duration	Min time (in milliseconds) of ringing to hook-off a FXO line
ringing-timeout	Time to wait for call connection
supervision-interval	Max supervision time to consider a FXO line available
wait-wink	Max time (in milliseconds) to wait for a wink signal after sending outgoing seizure
wink-duration	Wink duration (in milliseconds) for wink start signaling
wink-wait	Time to wait (in milliseconds) before sending wink-signal after detecting incoming seizure
tone-level	Volume to generate DTMF digits
track	Track this line
nsla-advisor	Use an nsla to track this line

<i>line-number</i>	This is the number of the line you are configuring. It accepts values ranging between 1 and the number of interface lines.  E1 and BRI interfaces only allow you to configure line 1 and the configuration is common to all the available channels.
<i>active</i>	Enables a line to make and receive calls.  All lines are active by default.
<i>alert voice-message</i>	Configures a voice message as the alert tone for this line. Accepts values ranging between 1 and 9.  By default, no voice message is enabled.  This cannot be configured in ISDN interfaces.
<i>auto-answer</i>	If this is enabled, the router answers an FXO line as soon as it detects the call current. If it is not enabled and the line has direct dialing with zero call time, the FXO line is not answered until the called picks up and establishes the call. Default is enabled.
<i>behavior</i>	Configures the behavior of a voip-dummy interface when a call is established. This can only be configured in voip-dummy interfaces.  By default this doesn't do anything, it only establishes the call.
<i>- play-loop</i>	Continually reproduces the configured voice message through the <b>line &lt;id&gt; play</b> command in loop mode until the other end hangs up.
<i>- play-disc</i>	Reproduces the configured voice message through the <b>line &lt;id&gt; play</b> command once only and hangs up the call when finished.
<i>- play-forward</i>	Reproduces the configured voice message through the <b>line &lt;id&gt; play</b> command once only and forwards the call to the indicated number. It is possible not to configure the forwarding number, in which case the call is diverted to the current destination number (this may have been changed with the outbound dial-peer <b>outgoing prefix/strip-digits/translation</b> commands).
<i>- play-ivr</i>	Plays the configured voice message through the <b>line &lt;id&gt; play</b> command once. When DTMF digits are received, the device's dial plan is checked and the call is forwarded to the matching destination.
<i>- translation</i>	A translation can be configured. Whenever it is set, it applies to the number received before consulting the dial plan (see manual bintec-Dm 722-I IP Telephony).
<i>- error-fwd</i>	Configures an extension number to divert to on receipt of a wrong number that does not fit the dial plan. The call ends if this extension is not configured.
<i>- timeout-fwd</i>	Configures an extension number to divert to when the maximum time permitted

	without receiving digits, configured through the <b>line &lt;id&gt; timing initial</b> command, expires. The call ends if this extension is not configured.
<i>busy voice-message</i>	Configures a voice message as a busy tone for this line. Accepts values ranging between 1 and 9.  By default, no voice message is enabled.  This cannot be configured in ISDN interfaces.
<i>caller-id</i>	In an fxs interface, this configures the method used for the caller number presentation. In an FXO interface, it activates or deactivates caller number detection.
- <i>bellcore</i>	Bellcore Type 1 standard is used. The caller number is transmitted between the first and second modulated ring (pursuant to the Bell 202 standard).
- <i>etsi</i>	The ETS 300 659-2 norm is used. The caller number is transmitted between the first and the second modulated ring (pursuant to the V23 norm). This is the default value.
- <i>none</i>	Does not show the caller number on FXS lines. Caller number detection is not performed on FXO lines.
- <i>gain</i>	Configures the input gain during caller id detection. The valid range goes from -31 to 31 dB. The default value is 0 dB.
<i>dial</i>	Sets the dialing parameters for FXS, FXO or E&M analogue interfaces.
- <i>level</i>	Configures the dial tone output level. The valid range is -31 to 0 dBm. By default, no value is configured and the tone level depends on the <i>speaker-gain</i> command.
- <i>pulse-detect</i>	Configures dialed number detection to use pulse dialing instead of DTMF tones, which is the default mode. This command is only available on FXS lines.
- <i>voice-message</i>	Configures a voice message in place of the dial tone on this line. Accepts values ranging between 1 and 9. By default, no voice message is enabled.
<i>default</i>	Sets the default values for a line.
<i>direct-dialing</i>	Allows you to configure a direct dial number. If you configure this field, when the phone is picked up a call is automatically made to this number when the time indicated by the second parameter times out. Accepts up to 15 digits (0 to 9) and between 0 and 30 seconds as wait time.  By default this parameter is not configured.  When this command is configured in voip-dummy interfaces, the interface automatically hooks off on startup and makes a call to the configured number. If the call ends, it is relaunched.
<i>echo-cancellation</i>	Allows you to enable/disable the echo cancellation for this line. By default echo cancellation is enabled in interfaces that have a DSP for codification: this is not supported by the voip-dummy interfaces.
- <i>suppression</i>	Echo suppression echo factor, a greater suppression factor means it is capable of suppressing stronger echoes. Default is 5. This is only available in motherboard voip isdn interfaces.
- <i>dtd-threshold</i>	Threshold used to decide if the return audio is an echo or the other interlocutor speaking; the greater the threshold means a greater level of echo can be detected. Default is 100. This is only available in motherboard voip isdn interfaces.
- <i>min-loss</i>	Minimum attenuation that the echo signal must have with respect to the original signal so the echo canceler behaves correctly. This can be configured at 0, 3, 6 and 9 dBs. Default is 6 dBs. Going below this value can slightly deteriorate the audio quality, but help eliminate specific problems of strong echoes. This is only available in VoIP card interfaces.
- <i>nlp</i>	Non Linear Processor configuration. It can be configured as adaptive (default configuration), disabled, or to completely mute the background noise. This is only available in FXS/FXO interfaces.
<i>error voice-message</i>	Configures a voice message as an error tone for this line. Accepts values ranging between 1 and 9.  By default, no voice message is enabled.  This cannot be configured in ISDN interfaces.
<i>e&amp;m-idle</i>	Configures the ABCD code sent by the associated CAS channel to signal the line is idle. This can only be configured in E1 interfaces. Default is 0x1.
<i>e&amp;m-lead-relay</i>	Allows remote reception and sending of the E&M pins state. This possibility only works if the line signaling is "always-on." The exchange of states is executed through RTP packets complying with the RFC28833 norm: hook-on state is rep-

	resented through event 144 and hook-off through event 159. The exchange of these events is only carried out through direct dial-peer (for further information, please see manual bintec-Dm 722-I Telephony over IP). The state of several remote E pins can be collected, meaning the result sent to the physical pin is the result of all received states. This can only be configured in E&M interfaces.  By default, this is configured to none.
- all	Allows E&M pin states exchange with the remote end.
- none	Does not remotely send any control signal for the interface.
e&m-mask	Configures the mask that must be applied to the ABCD signal received by the associated CAS channel. This can only be configured in E1 interfaces. Default is 0xc.
e&m-operation	Configures the type of wiring to be used in an E&M line. When two wires are configured, the interface must use a hybrid coil to pass from 2 to 4 wires. This can only be configured in E&M interfaces. When this is configured for one line, the rest of the lines in the board will automatically follow suit.  By default, this is configured for two wires.
- 2-wires	Configures the line so the audio goes over two wires.
- 4-wires	Configures the line so the audio goes over four wires.
e&m-seize	Configures the ABCD code sent by the associated channel CAS to signal the condition of the line in hook-off. This can only be configured in E1 interfaces. Default is 0xd.
e&m-signal	Configures the signaling mode to be used when a call is established so that the receptor can notify it may receive the number to call. This can only be configured in E&M interfaces.  By default, wink-start is configured.
- delay-dial	The signaling mode is delay. The caller can delay number dialing through a signaling pulse.
- immediate	The signaling mode is immediate. The caller can send the called number after hook-off without waiting for any notification from the call receptor.
- wink-start	The signaling mode is wink-start. The caller cannot send the called number until the wink pulse is received from the receptor.
- always-on	This does not use any signaling. The interface is always hook-off and its state is independent to the M pin level (pin 2). You can only relay the E&M pins status with this type of signaling. This mode can only be used with a direct dial-peer (for further information, please see manual bintec-Dm 722-I Telephony over IP).
e&m-type	Determines the configuration of the power supply, current detectors and type of wiring used at both ends to notify situations of hook-on and hook-off. This can only be configured in E&M interfaces. On configuring this over a line, the rest of the board lines automatically follow suit.  By default, the configuration is type 1.
- type-i	The configuration corresponds to E&M 1.
- type-ii	The configuration corresponds to E&M 2.
- type-iii	The configuration corresponds to E&M 3.
- type-v	The configuration corresponds to E&M 5.
loclevel	This is the volume gain in decibels applied to the voice message reproduction.
mic-gain	Indicates the line input gain in decibels. These values can vary between -31 and 31 dB.  Default value is 10 dB for POTS interfaces and -5 dB for E1 and BRI interfaces.
forward-digits	Enables forwarding the called number through DTMF tones in outgoing calls over an FXS interface when the call is hooked off. The time between the hook off being detected and the forwarding of the digits is 1 second by default and can be configured through the <b>line &lt;id&gt; timing dialout-delay</b> command. This is only available for FXS interfaces.
fork-call	Enables the possibility of establishing a fork call, i.e., admits a call on a line that was previously busy with another call. Consequently what is received from one of the calls is transmitted to the other call.  This option is disabled by default.
framer	Allows you to configure gains, attenuators and loops in analog interface lines (POTS and E&M interfaces). This command is not available for ISDN interfaces.



	Modifying these parameters may significantly alter the audio quality and affect the generation and detection of tones. Therefore, these should not be modified except at the request of our technical service. For further information on how each of the configuration parameters works, please see 'The analog board Framer' in the appendix of this manual.
- <i>external-loopback</i>	Enables the external loop the Framer has at the configuration level. By default, this is disabled.
- <i>internal-loopback</i>	Enables the internal loop the framer has at the configuration level so the audio sent to the framer is returned to the DSP. By default, this is disabled.
- <i>input-gain</i>	Modifies the framer input gain. Accepts values ranging between 0 and 120, resulting in a variation between 0 and 12 dB. Default is 0. For further information on this, please see 'The analog board Framer' in the appendix of this manual.
- <i>output-loss</i>	Modifies the framer output loss. Accepts values ranging between 0 and 120, resulting in a variation between 0 and 12 dB. Default is 0. For further information on this, please see 'The analog board Framer' in the appendix of this manual.
- <i>GIN</i>	Enables/disables the input attenuator known as GIN. Accepts values ranging between 0 (enabled) or 1 (disabled). Default is enabled. For further information on this, please see 'The analog board Framer' in the appendix of this manual.
- <i>AX</i>	Enables/disables the 6.02 dB input gain known as AX. Accepts values 0 (disabled) or 1 (enabled). It is disabled by default. For further information on this, please see 'The analog board Framer' in the appendix of this manual.
- <i>AR</i>	Enables/disables the 6.02 dB output attenuator known as AR. Accepts values 0 (disabled) or 1 (enabled). It is disabled by default. For further information on this, please see 'The analog board Framer' in the appendix of this manual.
- <i>LRG</i>	Enables/disables the 6.02 dB output attenuator known as LRG. Accepts values 0 (disabled) or 1 (enabled). It is disabled by default. For further information on this, please see 'The analog board Framer' in the appendix of this manual.
- <i>X-filter</i>	Enable/disable the digital filter known as X-filter, configured as a high-pass filter. Accepts values 0 (disabled) or 1 (enabled). It is disabled by default.
<i>impedance</i>	Allows you to configure an E&M line impedance. This can only be configured in E&M interfaces. When this is configured on one line, the rest of the lines automatically adopt the same behavior.  By default, lines are configured with low impedance.
- <i>high</i>	The line adopts an impedance of 900 ohm.
- <i>low</i>	The line adopts an impedance of 600 ohm.
<i>interface-type</i>	Allows you to configure the type of interface for POTS lines. Some POTS VoIP boards do not allow this parameter to be configured as it is fixed at a predetermined value.  By default, lines are configured as <b>FXO</b> when this is configurable. The configuration is always displayed through the corresponding text.
- <i>fxs</i>	The line is connected to a telephone (FXS, Foreign eXchange Station).
- <i>fxo</i>	The line is connected to a telephone line (FXO, Foreign eXchange Office).
<i>ip-calls</i>	Specifies if calls to IP destinations are allowed on this line.  It is permitted by default.
<i>forward-no-answer</i>	Telephone where the call is diverted if there is no answer.  Default is no divert configured.
<i>no</i>	Disables the selected option or sets the default values.
<i>play</i>	Allows you to configure the voice message to be played in a voip-dummy interface when behavior is reproduce voice message.  No voice message is configured for any codec by default.
- <i>G711</i>	Indicates the number of the voice messaged to be played when the codec negotiated in the call is G711. Accepts values that range between 1 and 9.
- <i>G723</i>	Indicates the number of the voice messaged to be played when the codec negotiated in the call is G723. Accepts values ranging between 1 and 9.
- <i>G729</i>	Indicates the number of the voice messaged to be played when the codec negotiated in the call is G729. Accepts values ranging between 1 and 9.
<i>pcm-loop</i>	For internal calls in the same card, a local loop is formed between the two lines implicated at a dsp level. Consequently, the audio does not pass through the



	device.
<i>speaker-gain</i>	This is the volume gain in decibels applied at reception. This can vary between -31 and 31.  Default is 0 dB.
<i>start-early-media</i>	Early Media sending is configured even though a RTP packet has not yet been received from the remote device. This option is disabled by default, so Early Media sending only starts when RTP traffic has been received from the other end.
<i>supervised</i>	Enables FXO line supervision, which lets you know if an FXO line is available to make calls. Supervision is executed by detecting the dial tone. This can only be configured in FXO interfaces. For further information, please see the section on "FXO line supervision" in the appendix of this manual.  It is not configured by default.
<i>suspend-mode</i>	Lines configured in FXS mode when receiving a call may behave in suspend mode. This mode allows you to hang up the called number without ending the established call and return to this call by simply picking up the called telephone. This mode is maintained for up to 30 seconds after which the call is considered ended. This command activates this mode.  By default, the device does not operate in this mode.
<i>telephone-number</i>	Configures the telephone number associated with the line. This is the caller telephone (ANI) which goes out in the call packets. Accepts up to 15 digits (0 to 9).  No number is configured by default.
<i>timing</i>	Configures the timings for the various parameters.
- <i>delay-duration</i>	Maximum time in milliseconds the caller expects the "delay" pulse to last in the delay-dial signaling. This can only be configured in interfaces with E & M signaling. Accepts values ranging between 100 and 5000 ms.  Default is 2000 ms.
- <i>delay-start</i>	Time in milliseconds that the calling waits to dial the called number from the moment of hook-off in delay-dial mode. This can only be configured in interfaces with E & M signaling. Accepts values ranging between 20 and 2000 ms.  Default is 300 ms.
- <i>dialout-delay</i>	Time in tenths of a second that a line waits before dialing the called number after hook-off. This can only be configured in FXS/FXO/E&M lines. Accepts values ranging between 1 and 100 tenths of a second.  Default is 10 tenths of a second.
- <i>forward-no-answer</i>	Time waited in seconds to forward a call because the called number has not answered. Accepts values ranging between 1 and 20 seconds.  Default is 10 seconds.
- <i>fxo-answer</i>	Time in milliseconds that the FXO line waits to pick up once the call current has been detected. No waiting time is set by default.
- <i>hook-flash</i>	The minimum and maximum times configured in milliseconds to detect hook-flash. Accepts values ranging between 20 and 1500 ms.  By default, range 30 is configured at 200 ms.
- <i>initial</i>	Maximum time waited to receive the first digit of the dialed number in calls coming in through the FXS, FXO or E&M interfaces. In dummy interfaces with play-ivr behaviour it is the maximum time to wait without receiving DTMF digits. Accepts values ranging between 0 and 180 seconds. If you configure 0, then there is no maximum time. Default is 30 seconds.
- <i>interdigit-timeout</i>	When a variable length string is being dialed, (T) is the time waited between digits. Whenever this time is surpassed, dialing is initiated. Accepts values ranging between 1 and 30 seconds.  Default is 3 seconds.
- <i>off-ringing-duration</i>	Interval in milliseconds in order to consider a ring signal inactive. If the ring signal is inactive for more than the permitted time, the call is hung up. If this is less than the permitted time, it is not considered as a valid period.  This is only logical in POTS FXO interfaces. The default interval is 500-4500
- <i>ringing-duration</i>	Minimum time in milliseconds that the ring signal must be active so the device can

	<p>detect this and pick up. This is only makes sense for FXO POTS interfaces. Accepts values ranging between 100 and 3000 ms. If a too-high value is configured, the FXO line will not pick up.</p> <p>Default is 100 ms.</p>
- <i>ringing-timeout</i>	<p>Maximum time that a line is ringing without connecting the call. If this times out, the call is released.</p> <p>The value is 100 seconds by default.</p>
- <i>supervision-interval</i>	<p>Interval where the dial tone is detected. Once this has timed out and the tone has not been detected, the line is considered unavailable and an error event is produced. This is only used when FXO line supervision is enabled. This can only be configured in FXO interfaces. Accepts values ranging between 2 and 12 seconds. For further information, please see the section on “FXO line supervision” in the appendix of this manual.</p> <p>By default, this is configured at 2 seconds.</p>
- <i>wait-wink</i>	<p>Maximum time the caller waits for the wink pulse to initiate in the wink-start signaling. This can only be configured in interfaces with E&amp;M signaling. Accepts values ranging between 100 and 5000 ms.</p> <p>Default is 550 ms.</p>
- <i>wink-duration</i>	<p>Duration in seconds of the “wink” pulse generated in incoming calls with wink-start signaling. This can only be configured in interfaces with E &amp; M signaling. Accepts values ranging between 50 and 3000 ms.</p> <p>Default is 200 ms.</p>
- <i>wink-wait</i>	<p>Time in milliseconds that the call receptor waits before sending the wink pulse to the caller. This can only be configured in interfaces with E &amp; M signaling. Accepts values ranging between 100 and 5000 ms.</p> <p>Default is 200 ms.</p>
<i>tone-level</i>	<p>This is the volume gain with which the DTMF digits are generated and the level (in dBm) used to detect the DTMF digits. This can vary between -31 and 0 dB.</p> <p>Default is 0 dB.</p>
<i>track</i>	<p>This can only be configured in FXO interfaces. This means line availability depends on a monitored parameter. When this is disabled, the line is in a hook-off state and produces a busy tone. If this is disabled when a call is in progress, the call will continue normally and the line will disable when the call terminates.</p>
- <i>nsla-advisor</i>	<p>Line activation/deactivation depends on the output of an NSLA advisor. Please see manual <b>bintec-Dm 754-I NSLA</b> for further details.</p>

#### Example:

Configures the device so calls executed from line 1 have caller number 123 and suspend mode is disabled in said line.

```
voip1/0 config>LINE 1 telephone-number 123
voip1/0 config>LINE 1 no suspend-mode
voip1/0 config>
```

#### Command history:

Release	Modification
10.08.34.05.09	The option <i>dial level</i> is added
10.08.43	The option <i>dial level</i> is added
10.09.25	The option <i>dial level</i> is added
11.00.00.02.08	The option <i>dial level</i> is added
11.00.04	The option <i>dial level</i> is added
11.01.00	The option <i>dial level</i> is added
11.01.00	The option <i>caller-id gain</i> is added
11.01.00	The option <i>echo-cancellation nlp</i> is added
11.01.00	The option <i>framer X-filter</i> is added
10.09.26	The option <i>caller-id gain</i> is added
10.09.26	The option <i>echo-cancellation nlp</i> is added

10.09.26	The option <i>framer X-filter</i> is added
11.00.05	The option <i>caller-id gain</i> is added
11.00.05	The option <i>echo-cancellation nlp</i> is added
11.00.05	The option <i>framer X-filter</i> is added
11.00.05	The option <i>no</i> is obsoleted. It has been replaced by command <i>no line</i> (see Section 2.1.6). Also, some misprints in help texts have been corrected.
11.01.01	The option <i>no</i> is obsoleted. It has been replaced by command <i>no line</i> (see Section 2.1.6). Also, some misprints in help texts have been corrected
11.00.05	In option <i>off-ringing-duration</i> , the unit used to introduce the value is explained in the help of command.
11.01.01	In option <i>off-ringing-duration</i> , the unit used to introduce the value is explained in the help of command.
11.00.05	In option <i>ringing-duration</i> , the unit used to introduce the value is explained in the help of command.
11.01.01	In option <i>ringing-duration</i> , the unit used to introduce the value is explained in the help of command.
11.00.05	In option <i>fxo-answer</i> , the unit used to introduce the value is explained in the help of command.
11.01.01	In option <i>fxo-answer</i> , the unit used to introduce the value is explained in the help of command.
11.00.05	In option <i>delay-duration</i> , the unit used to introduce the value is explained in the help of command.
11.01.01	In option <i>delay-duration</i> , the unit used to introduce the value is explained in the help of command.
11.00.05	In option <i>delay-start</i> , the unit used to introduce the value is explained in the help of command.
11.01.01	In option <i>delay-start</i> , the unit used to introduce the value is explained in the help of command.
11.00.05	In option <i>wait-wink</i> , the unit used to introduce the value is explained in the help of command.
11.01.01	In option <i>wait-wink</i> , the unit used to introduce the value is explained in the help of command.
11.00.05	In option <i>wink-duration</i> , the unit used to introduce the value is explained in the help of command.
11.01.01	In option <i>wink-duration</i> , the unit used to introduce the value is explained in the help of command.
11.00.05	In option <i>wink-wait</i> , the unit used to introduce the value is explained in the help of command.
11.01.01	In option <i>wink-wait</i> , the unit used to introduce the value is explained in the help of command.
10.09.27	The option <i>e&amp;m-mask</i> is added.
11.00.06	The option <i>e&amp;m-mask</i> is added.
11.01.02	The option <i>e&amp;m-mask</i> is added.

## 2.1.6 NO

Sets the parameters to their default values or disables them. The configurable elements are:

### Syntax:

```
voip2/0 config>NO ?
dsp-firmware      Firmware to load into the dsp
isdn              Isdn configuration parameters
line              Line configuration options

voip2/0 config>NO line <line-number>
active            Line can receive and make calls

alert voice-message  Substitute the alert tone with a voice message

auto-answer       Answer an fxo line before the call being connected
```

behavior	voip-dummy audio behavior
busy voice-message	Substitute the busy tone with a voice message
caller-id	Set the caller id generation standard
gain	set default caller id detection gain
dial	Dial configuration options
level	Configure dial tone output level
pulse-detect	Use pulse dialing instead of dtmf
voice-message	Substitute the dial tone with a voice message
direct-dialing	Call is automatically carried out to this number
echo-cancellation	Echo cancellation is done in the line
<cr>	
suppression	Suppression factor, default 5
dtd-threshold	Double talk detector threshold, default 1
min-loss	Min echo return loss, default 6dB
error voice-message	Substitute the error tone with a voice message
e&m-idle	E&M digital idle ABCD signal
e&m-lead-relay	E&M signals relay mode
e&m-mask	E&M mask to apply to Rx ABCD signal
e&m-operation	E&M wiring mode
e&m-seize	E&M digital seize ABCD signal
e&m-signal	E&M signaling modes
e&m-type	E&M working types
fork-call	Enable fork call
forward-digits	Send called number as DTMF
forward-no-answer	Telephone to forward the call if the caller doesn't answer
framer	Framer configuration menu
external-loopback	Set an external loopback at the framer
internal-loopback	Set an internal loopback at the framer
hold voice-message	Hold voice message
impedance	Impedance
ip-calls	Permit making calls over IP
loc-level	Indicate the line output gain to play
voice messages	
mic-gain	Indicate the line input gain
pcm-loop	Loop internal calls at the pcm bus
play	play a voice-message in a dummy ifc
G711 voice-message	voice-message coded as G.711 samples
G723 voice-message	voice-message coded as G.723 frames
G729 voice-message	voice-message coded as G.729 frames
speaker-gain	Indicate the line output gain

start-early-media	Start sending Early Media as soon as possible
supervised	Supervise FXO line availability
suspend-mode	Suspend call when hanging FXS line
telephone-number	Configure the telephone number associated to the line
timing	Configure timing parameters
delay-duration	Max delay-signal duration (in milliseconds)
delay-start	Time to wait (in milliseconds) before dialing digits in delay-dial mode
dialout-delay	Time to wait before dialing digits
forward-no-answer	Time to wait before forwarding the call if no answer
fxo-answer	Minimum ringing time (in milliseconds) before an fxo line hook-off
hook-flash	Configure hook-flash timings
initial	Max initial time before receiving dialed digits
interdigit-timeout	Time to wait before starting a call between two dtmf
off-ringing-duration	Interval (in milliseconds) for the ring signal being inactive
playout-delay	Time to wait before dialing digits
ringing-duration	Min time (in milliseconds) of ringing to hook-off a FXO line
ringing-timeout	Time to wait for call connection
supervision-interval	Max supervision time to consider a FXO line available
wait-wink	Max time (in milliseconds) to wait for a wink signal after sending outgoing seizure
wink-duration	Wink duration (in milliseconds) for wink start signaling
wink-wait	Time to wait (in milliseconds) before sending wink-signal after detecting incoming seizure
tone-level	Volume to generate DTMF digits
track	Track this line
nsla-advisor	Use an nsla to track this line

#### Command history:

Release	Modification
11.00.05	<i>no line</i> command is added
11.01.01	<i>no line</i> command is added
10.09.27	The option <i>e&amp;m-mask</i> is added.
11.00.06	The option <i>e&amp;m-mask</i> is added.
11.01.02	The option <i>e&amp;m-mask</i> is added.

## Chapter 3 Monitoring VoIP Interfaces

### 3.1 VoIP Interface monitoring commands

#### 3.1.1 BASE-INTERFACE

This allows you to monitor aspects relative to the call request and reception mechanism located between the ISDN VoIP interfaces and the associated base interface. This command is only available in ISDN VoIP interfaces.

*Syntax:*

```
voip2/0+BASE-INTERFACE
Base Interface Console
voip2/0 Base IFC+?
LIST
EXIT
```

##### 3.1.1.1 LIST

Allows you to view information on the channels available to execute and receive calls between VoIP interfaces and the ISDN base interface.

*Example:*

```
voip2/0 Base IFC+LIST
Item: 1                      Base ifc: bri2/0
Destination Addr:           Local Addr:
Circuit id request: 255     Dial Circuit status: DOWN
Circuit id assigned: NOT ASSIGNED

owner: 0
profile: 13bec10
regd_dbnode: 11cb510
regd_pry: 11cb4b8
regd_makecallarg: 13be810
dlci_req: 255
frsize_rq: 80
flags: 2c
status_chn: 0
cookie_chn: 0
dlci_asig: 0
dnet_chans: 0
dnet_speed: 0

Item: 2                      Base ifc: bri2/0
Destitiation Addr:         Local Addr:
Circuit id request: 255     Dial Circuit status: DOWN
Circuit id assigned: NOT ASSIGNED

owner: 0
profile: 13be610
regd_dbnode: 11cb510
regd_pry: 11cb4b8
regd_makecallarg: 13be510
dlci_req: 255
frsize_rq: 80
flags: 2c
status_chn: 0
cookie_chn: 0
dlci_asig: 0
dnet_chans: 0
dnet_speed: 0
```

### 3.1.2 BYPASS

Allows you to control the bypass relay or relays that some VoIP interfaces have between some of their terminals.

**Syntax:**

```
voip2/0+BYPASS ?
Enable 1, Disable 0[0]? 1
voip2/0+
```

The bypass status is displayed through the DEVICE monitoring command if the said interface has one:

```
*process 3
Console Operator
+DEVICE voip2/0

Interface          CSR      Vect      Auto-test   Auto-test   Maintenance
                  F1001000  22        valids     failures    failures
voip2/0            F1001000  22        1          0           0

Board ID: 1714040-1840
Firmware id: 26AE09B4
N Channel: 4-4-4
Type: PSTN / POTS
Support:   1 FXS   2 FXO   3 FXS   4 FXO

Bypass: OFF
+
```

If the interface does not contain any bypasses, this command has no effect. For further information on the DEVICE command, please see manual bintec-Dm 704-I Configuration and Monitoring.

Bypasses, by default, are enabled when the device is off, the routing application inactive or if the board has not initialized correctly.

### 3.1.3 CLEAR

Allows you to delete distinct counters or information registered by the interface related to interface monitoring.

**Syntax:**

```
voip2/0+CLEAR ?
STATISTICS
```

#### 3.1.3.1 CLEAR STATISTICS

Deletes statistics on a line. These statistics refer to the low level counters associated with the DSP.

**Syntax:**

```
voip2/0+CLEAR STATISTICS
Line: [1]?
voip2/0+
```

### 3.1.4 DISPLAY

Dynamically displays the status of the distinct voice lines and other additional parameters.

**Syntax:**

```
voip2/0>DISPLAY ?
LINE
```

#### 3.1.4.1 DISPLAY LINE

Allows you to view the state of a line and whether the codec is currently active.

It also provides information on whether the line is forwarded to another extension unconditionally or when there is no

answer.

When the interface is in MGCP mode, this command lets the user know whether it is being controlled by MGCP or it is in survival mode. Additionally, it lets the user know an active call is ongoing.

*Example:*

```
voip1/0>DISPLAY LINE 1
State: IDLE                      Forward:                      No answer frwd:
Codec: G711-Alaw                 VAD: Disabled

MGCP:   Controlled by MGCP
MGCP connection:                 Active

Rx frames 816, Write erros 0

Driver information:
  last rx delay 3(52/49), buffered rx frames 4, missed rx frames 0 fax errors 0
  (since last print)
  s1 0, s2 2, ints 1647 rx bytes 131760 tx bytes 130240 codec err 3 dup frames
0 cng 0

voip1/0>
```

If the interface supports this option, the command will also display DSP low level statistics related to frames which have been sent or received, as well as errors, etc.

## 3.1.5 LIST

*Syntax:*

```
voip1/0>LIST ?
ACTIVE
RELEASED
STATISTICS
PORTS
voip1/0>
```

### 3.1.5.1 LIST ACTIVE

Lists the active calls associated with this interface. This displays the associated line, the caller and the called, the type of call, the lines involved in the call, the codec, bytes transmitted and received, call start time and date and the time and date when the call was established.

On listing the called telephone, information is displayed on the results of applying the configured numerical expansions, both for incoming and outgoing calls.

*Syntax:*

```
voip1/0>LIST ACTIVE <registers to display>
voip1/0>
```

*Example:*

```
voip1/0>LIST ACTIVE 1
CallId 5, CallerAddress 12, CalledAddress 23, CallType SIP Gw, CallingPeer voip1/0-2,
CalledPeer 172.24.100.133, Codec G723 5.3, TransmitByt
es 0, ReceiveBytes 0, SetupTime 15:13:03 Sun 24 2005, ConnectTime 15:13:06 Sun 24
2005, DisconnectTime
voip1/0>
```

### 3.1.5.2 LIST ENERGY

Allows you to monitor the audio signal energy received by a given line. This command is useful to configure parameters related to detecting hook-on in FXO interfaces.

This command first obtains 200 energy values and then displays the information. The energy is shown in dBm and the position of the displayed data is indicative of their magnitude, i.e., small energy values are displayed close to the left-hand console margin and high values near the right-hand console margin. Finally, the dynamic range for these is shown.



Each value corresponds to the average energy during the interval between active codec samples. The interval for codec G.723 is 30 ms and for G.729 10 ms.

**Syntax:**

```
voip2/0+LIST ENERGY
Line: [1]?
-60
-60
    -40
        -20
...
-60
-60

Audio range (dBm): -60 / -20
voip2/0+
```

For further information, please see Chapter V section “Detecting the busy tone in FXO” in this manual. This command does not work in ISDN interfaces.

### 3.1.5.3 LIST RELEASED

Lists the last released calls associated with this interface. The information displayed is identical to that shown for active calls, but it also includes information on the time, date and cause of disconnection.

On listing the called telephone, information is displayed on the results of applying the configured numerical expansions, both for incoming and outgoing calls. The release cause appears encoded; to obtain information on the meanings, use the **DISPLAY CAUSE RELEASE** command.

**Syntax:**

```
voip1/0>LIST RELEASED <registers to display>
voip1/0>
```

**Example:**

```
voip1/0>LIST RELEASED 1
CallId 5, CallerAddress 12, CalledAddress 23, CallType SIP Gw, CallingPeer voip1/0-2,
CalledPeer 172.24.100.133, Codec G723 5.3, TransmitBytes 0, ReceiveBytes 0, SetupTime 15:13:03 Sun 24 2005, ConnectTime 15:13:06 Sun 24
2005, DisconnectTime 11:11:12.45 Sun 24 2005
voip1/0>
```

## 3.1.6 LOOP

Allows you to activate or deactivate various types of loops in order to check the functionality of various hardware elements. The available loops depend on the type of VoIP interface, as said loops are closely associated with the actual hardware.

Any VoIP board is basically made up of one or various DSPs which carry out the process of audio encoding/decoding and a *framer* that exchanges the digital samples with the external audio interface. Communications between the DSP and the framer are carried out in digital. In cases of analog boards (e.g., POTS and E&M boards), this framer carries out the analog-digital and digital-analog conversion.



**Note**

If you activate the loops for any type of test, don't forget to deactivate them once you have finished, otherwise the hardware functionality will change.

**Syntax:**

```
voip1/0+LOOP ?
FRAMER
DSP
SIGNAL
voip1/0+
```

### 3.1.6.1 LOOP FRAMER

Allows you to activate or deactivate various types of loops the board framer has. This command is only operative in analog boards (e.g., POTS and E&M boards). Loop control is carried out by the line.

#### Syntax:

```
voip1/0+LOOP FRAMER ?
DSP-INTERNAL
EXTERNAL
INTERNAL
NONE
```

#### 3.1.6.1.1 LOOP FRAMER DSP-INTERNAL

Enables a loop in the *framer* so it transmits to the DSP everything that it has received from the DSP. All audio received from the exterior is ignored. The difference between this and the LOOP FRAMER INTERNAL is that the audio is not altered by any filters, amplifiers or attenuators. For further information on this, please see 'The analog board Framer' found in the appendix of this manual.

Requests the line number where the loop is applied. By default, no loops are enabled in the interface.

#### Example:

```
voip2/0+LOOP FRAMER DSP-INTERNAL
Line: [1]? 1
voip2/0+
```

#### 3.1.6.1.2 LOOP FRAMER EXTERNAL

Enables a loop in the framer so it transmits everything received in the analog pins through the analog pins. All audio received from the DSP is ignored. For further information on this, please see 'The analog board Framer' found in the appendix of this manual.

Requests the line number where the loop is applied. By default, no loops are enabled in the interface.

#### Example:

```
voip2/0+LOOP FRAMER EXTERNAL
Line: [1]? 1
voip2/0+
```

#### 3.1.6.1.3 LOOP FRAMER INTERNAL

Enables a loop in the framer so it transmits to the DSP everything that it has received from the DSP.

All audio received from the exterior is ignored. The difference between this and the LOOP FRAMER DSP-INTERNAL is that the audio is altered by filters, amplifiers and attenuators. For further information on this, please see 'The analog board Framer' found in the appendix of this manual.

Requests the line number where the loop is applied. By default, no loops are enabled in the interface.

#### Example:

```
voip2/0+LOOP FRAMER INTERNAL
Line: [1]? 1
voip2/0+
```

#### 3.1.6.1.4 LOOP FRAMER NONE

Deactivates all active loops in the *framer*.

Requests the line number where the loop is applied. By default, no loops are enabled in the interface.

#### Example:

```
voip2/0+LOOP FRAMER NONE
Line: [1]? 1
voip2/0+
```

### 3.1.6.2 LOOP DSP

Allows you to activate or deactivate various types of loops that the board DSP has. These loops are dependent on the type of DSP in the board. Loop control is carried out by the line.

*Syntax:*

```
voip1/0+LOOP DSP ?
CODEC
EXTERNAL
NONE
```

#### 3.1.6.2.1 LOOP DSP CODEC

Enables a loop in the DSP audio output so all audio received from the *framer* is returned to it. The DSP alters the audio so it is affected by the echo cancellation, input and output gains as well as the encoding and decoding processes.

Audio received from the framer is also processed at the IP layer so if communication is established with another interlocutor, said audio is received. Contrariwise, any audio received from this interlocutor is ignored so the DSP only sends the framer the audio received from it.

Requests the line number where the loop is applied. By default, this is disabled.

*Example:*

```
voip2/0+LOOP DSP CODEC
Line: [1]? 1
voip2/0+
```

#### 3.1.6.2.2 LOOP DSP INTERNAL

In voip-cellular interfaces, enables a loop in the PCM audio interface of the module 3G/4G/LTE, so all audio received from the framer is returned to it. In other words, the sound received from the telephone's microphone is returned to the handset. This way, the audio can be tested.

*Example:*

```
voip2000+loop dsp internal ?
<1..30> Line number
voip2000+loop dsp internal 1
```

**Command history:**

Release	Modification
11.00.05	Audio interface can be tested in MC73xx modules
11.01.01	Audio interface can be tested in MC73xx modules

#### 3.1.6.2.3 LOOP DSP EXTERNAL

Enables a loop in the DSP audio output so all audio received from the framer is returned to it. The DSP does not alter the audio in any way so audio hookup in POTS boards is produced.

Audio received from the framer is also processed at the IP layer so if communication is established with another interlocutor, the said audio is received. On the other hand, any audio received from this interlocutor is ignored so the DSP only sends the framer the audio received from it.

Requests the line number where the loop is applied. By default, this is disabled.

*Example:*

```
voip2/0+LOOP DSP EXTERNAL
Line: [1]? 1
voip2/0+
```

#### 3.1.6.2.4 LOOP DSP NONE

Deactivates the loop the DSP is currently executing. Also, in voip-cellular interfaces disables the loop in the PCM audio interface of the module 3G/4G/LTE.

Requests the line number where the loop is applied. By default, the DSP does not execute loops.

**Example:**

```
voip2/0+LOOP DSP NONE
Line: [1]? 1
voip2/0+
```

**Command history:**

Release	Modification
11.00.05	Audio loopback can be disabled in MC73xx modules
11.01.01	Audio loopback can be disabled in MC73xx modules

**3.1.6.3 LOOP SIGNAL**

Controls the E&M signals loop in E&M analog boards. This command is only operational in E&M boards when the line signaling is *always-on* and the *relay* for the said signals is enabled.

The signals *relay* permits remote reception and transmission of the E and M pins states respectively. This possibility only works if the line signaling is *always-on*. The exchange of the states is carried out through RTP packets complying with the RFC2833 norm: the hook-on state is represented by event 144 and hook-off through event 159. The exchange of these events is only executed through a direct dial-peer (for further information, please see manual bintec-Dm 722-I Telephony over IP).

Loop control is carried out by the line. By default, the loops are disabled.

**Syntax:**

```
voip1/0+LOOP SIGNAL ?
INTERNAL
EXTERNAL
NONE
```

**3.1.6.3.1 LOOP SIGNAL INTERNAL**

Enables the E and M signals loop at the RTP layer when signaling is *always-on* and the *relay* for the signals is enabled. The loop is such that the remote M pin state is not reflected in the local E&M interface but returned to the remote interface. The local interface M pin state is ignored. This is an internal loop in the sense that this loop does not externally manifest in the interface executing this.

Requests the line number where the loop is applied. By default, this is disabled.

**Example:**

```
voip2/0+LOOP SIGNAL INTERNAL
Line: [1]? 1
voip2/0+
```

**3.1.6.3.2 LOOP SIGNAL EXTERNAL**

Enables the E and M signals loop at the local layer when the signaling is *always-on* and the *relay* for the signals is enabled. The loop is such that the local interface M pin state is reflected in the E pin in the same interface. The remote M pin state is ignored and the local M pin state is not sent to the remote interface.

Requests the line number where the loop is applied. By default, this is disabled.

**Example:**

```
voip2/0+LOOP SIGNAL EXTERNAL
Line: [1]? 1
voip2/0+
```

**3.1.6.3.3 LOOP SIGNAL NONE**

Disables the loop currently being executed.

Requests the line number where the loop is applied. By default, this does not execute loops.

**Example:**

```
voip2/0+LOOP SIGNAL NONE
Line: [1]? 1
voip2/0+
```

### 3.1.7 TEST

Controls the different types of test which allow you to check if the interface is operating correctly. The tests depend on the type of interface and the type of DSP in the board as they use specific functionalities for these. Some tests will require calls to be established and others to enable loops.



#### Note

Some tests need to be deactivated once they have finished so please remember to do that, otherwise the hardware functionality will change.

#### Syntax:

```
voip1/0+TEST ?
LOCAL
MUTE
REMOTE
TONE
```

#### Command history:

Release	Modification
10.09.26	The option <i>mute</i> is added
11.00.05	The option <i>mute</i> is added
11.01.00	The option <i>mute</i> is added

#### 3.1.7.1 TEST LOCAL

Activates a test in the interface to automatically check line functionality. This test is based on the generation and detection of the hook-on tone (CPT) by the DSP on the indicated line. To execute this test successfully, you need to externally wire the line's audio reception to the transmission. You can also do this by enabling the internal loop in the framer. Given that the generation and detection take place in the same line, the test is considered local.

This test consists of detecting the hook-on tone three times at one second intervals. During this process the partial statistics are shown. Once this has finished the results of whether the test has been passed or not are shown.

Requests the line number where the test is to be executed. By default, the test is deactivated.

If you use a framer loop to execute the test, don't forget to enable it first and disable it once the test has been completed.

#### Example:

```
voip2/0+TEST LOCAL ?
Line: [1]?
    -- Local test based on CPT --
running ... test n: 3 passed 2 failed 1
TEST FAILED

voip2/0+
```

#### 3.1.7.2 TEST MUTE

Allows you to mute one or both audio directions in the card framer, independently for each line.

Option *rx* mutes the audio received in the framer from the DSP, which is the audio being sent to the line.

Option *tx* mutes the audio sent from the framer to the DSP, which is the audio received from the line.

Option *none* disables any active mute.

#### Syntax:

```
voip2/0+test mute ?
rx      Mute framer RX from DSP. TX to line
tx      Mute framer TX to DSP. RX from line
none    Disable framer mute

voip2/0+
```

### 3.1.7.3 TEST REMOTE

Activates a test in the interface to automatically check remote communications functionality. This test is based on the detection of the hook-on tone (CPT) by the DSP on the indicated line. The tone you need to detect is generated from a remote interlocutor so the test must take place in an active conversation. To do this, the remote interlocutor must internally activate the TEST TONE DSP-CPT. To execute this test successfully, you need to externally wire the line's audio reception to the transmission. You can also do this by enabling the internal loop in the framer. Given that the tone generation is remote, the test is considered remote.

This test consists of detecting the hook-on tone three times at one second intervals. During this process the partial statistics are shown. Once finished, the results of whether the test has been passed or not are shown.

Requests the line number where the test is to be executed. By default, the test is deactivated.

If you use a framer loop to execute the test, don't forget to enable it first and disable it once the test has been completed. Also, don't forget to enable tone generation in the remote interlocutor before executing the test and disable it once the test has finished.

*Example:*

```
voip2/0+TEST REMOTE
Line: [1]? 1
    -- Remote test based on CPT --
running ... test n: 3 passed 0 failed 3
TEST FAILED
voip2/0+
```

### 3.1.7.4 TEST TONE

Allows you to activate/deactivate different test tones in order to check the audio levels. To control these, request the line and operation to carry out. By default, there are no active tones.

*Syntax:*

```
voip2/0+TEST TONE ?
DSP-1KHZ-INTERNAL
DSP-CPT
FRAMER-1KHZ-EXTERNAL
NONE
```

#### 3.1.7.4.1 TEST TONE DSP-1KHZ-INTERNAL

Allows you to activate/deactivate a 1KHz test tone in the DSP in reception direction from the DSP. If this tone activates in a conversation, the remote interlocutor will hear it. Audio received from the framer is ignored.

Requests the line number where the loop is applied and the operation to be executed. By default there are no active tones.

*Example:*

```
voip2/0+TEST TONE DSP-1KHZ-INTERNAL
Line: [1]? 1
Mode (0 ->off 1 ->on) [0]? 1
voip2/0+
```

#### 3.1.7.4.2 TEST TONE DSP-CPT

Allows you to activate/deactivate a test hook-on tone (call proceeding tone) in the DSP. You can select the generation direction for this: if it's internal, it is generated in the DSP reception direction so if there is an active conversation, the remote interlocutor will hear it: if it's external, it is generated in the DSP transmission direction and heard in the local telephone interface. It is useful to execute a remote test. For further information, please see the TEST REMOTE command.

Requests the line number where the loop is applied and the operation to be executed. By default there are no active tones.

*Example:*

```
voip2/0+TEST TONE DSP-CPT
Line: [1]? 1
Mode (0 -> turn off 1 ->external 2->internal) [0]? 1
```

```
voip2/0+
```

### 3.1.7.4.3 TEST TONE FRAMER-1KHZ-EXTERNAL

Allows you to activate/deactivate a 1KHz test tone in the framer in the framer reception direction. If this tone activates the local telephone terminal will hear it. Audio received from the DSP is ignored. For further information on this, please see 'The analog board Framer' found in the appendix of this manual.

Requests the line number where the loop is applied and the operation to be executed. By default there are no active tones.

*Example:*

```
voip2/0+TEST TONE FRAMER-1KHZ-EXTERNAL
Line: [1]? 1
Mode (0 ->off 1 ->on) [0]? 1
voip2/0+
```

### 3.1.7.4.4 TEST TONE NONE

Deactivates all tones which are currently active.

Requests the line number where the loop is applied. By default, there are no active tones.

*Example:*

```
voip2/0+TEST TONE NONE
Line: [1]? 1
voip2/0+
```

## 3.1.8 TRACE

*Syntax:*

```
voip2/0+TRACE ?
LEVEL
```

### 3.1.8.1 TRACE LEVEL

For low level debugging tasks, some VoIP boards display traces on what they do. This command controls the detail level for these. By default, low-level traces are enabled. Trace level is disabled when set to 0.

*Syntax:*

```
voip2/0+TRACE LEVEL
Level (0..7), 0->disable[0]? 0
voip2/0+
```

The displayed traces are not associated with any events subsystem. The process of displaying traces overloads the device, so we recommend that they are not enabled unless requested by our technical support team.



#### Note

The trace level is maintained between device startups, if a given level is enabled and you do not want to use it, it must be expressly disabled.

## Chapter 4 Types of VoIP Interfaces

### 4.1 POTS Interface

#### 4.1.1 Description

This interface permits audio to be established and transmitted over a conventional telephone wire.

Depending on the board, a POTS line can be configured either in FXS (Foreign Exchange Station) or FXO (Foreign Exchange Office).

FXS mode permits direct connection to a conventional telephone, a fax or similar device. This supplies the power voltage, the rings, detects when the connected device hooks-on or off and transmits and receives the audio analog signals. In this mode, the interface behaves as if it were the telephone network.

In FXO mode, the line behaves as if it was a conventional telephone and expects the power voltage to be supplied to it. This permits the line to be directly connected to the public telephone network or to a telephone switchboard (PABX).

From the point of view of voice codification, these boards have a signal processor which supports G.711, G.729 annex A and annex B and G.723 at 5.3 and 6.4 kbps. For performance reasons, not all the boards simultaneously support G.723 and G.729 codecs.

#### 4.1.2 Connector

An RJ11 connector is used with the following pin configuration:

Pin 1— Not connected

Pin 2— Not connected

Pin 3— ring

Pin 4— tip

Pin 5— Not connected

Pin 6— Not connected

#### 4.1.3 Types of VoIP POTS Boards



##### Warning

**The specifications and availability of the said boards are subject to change without prior notice. For further information, please contact your regular supplier.**

A line in FXS mode admits the connection of a limited number of telephone terminals which depends on its REN value (Ringer Equivalent Number). As a general rule, this only connects one terminal to each FXS line, especially if the telephone terminals have screens or other devices which need additional requirements from the line; otherwise the behavior will not be what you expect.

##### 4.1.3.1 VoIP board with 4 FXS/FXO lines

The four lines can be independently configured in FXS or FXO mode via the console.





**Important**

Do not connect a line configured as FXS to a telephone network or a terminal behaving as FXS. This could seriously damage the line and/or the board supporting this.

#### 4.1.3.2 VoIP board with 2 FXS/FXO lines

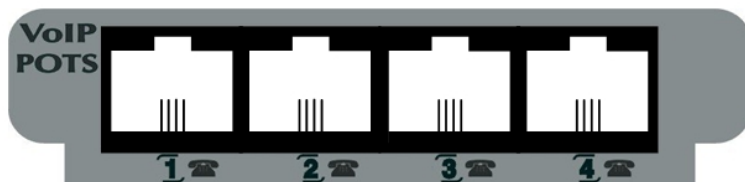
The two lines can be independently configured in FXS or FXO mode via the console.

**Important**

Do not connect a line configured as FXS to a telephone network or a terminal behaving as FXS. This could seriously damage the line and/or the board supporting this.

#### 4.1.3.3 VoIP board with 4 FXS lines

The operating mode for lines configured as FXS is fixed and cannot be changed.

**Important**

Do not connect a line configured as FXS to a telephone network or a terminal behaving as FXS. This could seriously damage the line and/or the board supporting this.

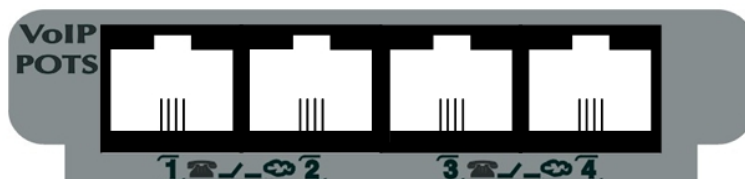
#### 4.1.3.4 VoIP board with 1 FXO line

The operating mode for a line configured as FXO is fixed and cannot be changed.



#### 4.1.3.5 VoIP board with 2 FXS lines and 2 FXO lines with bypass

The operating mode for the lines (FXS/FXO) is fixed and cannot be changed.



Between lines 1 (FXS) and 2 (FXO) there is a bypass relay which connects them when the device is switched off. The same thing occurs between lines 3 (FXS) and 4 (FXO). In the serigraphy this bypass is indicated by the relevant symbol. The relay can be controlled through the BYPASS voip monitoring command.

```
+network voip3/0
-- Voip network monitor --
voip3/0+bypass
Enable 1, Disable 0[0]? 1
voip3/0+
```

The bypass relay state can be monitored through the DEVICE interfaces monitoring command.

```
*p 3
Console Operator
+device voip1/0

Interface          CSR      Vect      Auto-test   Auto-test   Maintenance
                  F0020000  22        valids     failures    failures
voip1/0           F0020000  22         1          0           0

Board ID: 1714040-1840
Firmware id: 26AE09B4
N Channel: 4-4-4
Type: PSTN / POTS
Support:   1 FXS   2 FXO   3 FXS   4 FXO

Bypass: OFF
+
```

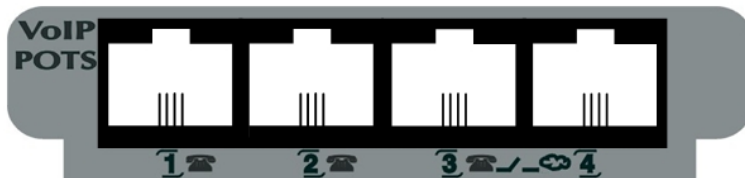


#### Important

**Do not connect a line configured as FXS to a telephone network or a terminal behaving as FXS. This could seriously damage the line and/or the board supporting this.**

#### 4.1.3.6 VoIP board with 3 FXS lines and 1 FXO line with bypass

The operating mode for the lines (FXS/FXO) is fixed and cannot be changed.



Between lines 3 (FXS) and 4 (FXO) there is a bypass relay which connects them when the device is switched off. In the serigraphy this bypass is indicated by the relevant symbol. The relay can be controlled through the BYPASS voip monitoring command.

```
+network voip3/0
-- Voip network monitor --
voip3/0+bypass
Enable 1, Disable 0[0]? 1
voip3/0+
```

The bypass relay state can be monitored through the DEVICE interfaces monitoring command.

```
*p 3
Console Operator
+device voip3/0

Interface          CSR      Vect      Auto-test   Auto-test   Maintenance
                  F0000000  22        valids     failures    failures
voip3/0           F0000000  22         1          0           0

Board ID: 1714040-1440
Firmware id: 26AE09B4
N Channel: 4-4-4
Type: PSTN / POTS
Support:   1 FXS   2 FXS   3 FXS   4 FXO

Bypass: OFF
```

+

**Important**

**Do not connect a line configured as FXS to a telephone network or a terminal behaving as FXS. This could seriously damage the line and/or the board supporting this.**

## 4.1.4 Configuration

From the configuration point of view, each POTS board is presented as a single voip interface over which the various lines are independently configured.

```
Config>LIST DEVICE
```

Interface	Connector	Type of interface
ethernet0/0	LAN1	Fast Ethernet interface
serial0/0	SERIAL0/WAN1	X25
serial0/1	SERIAL1/WAN2	X25
serial0/2	SERIAL2/WAN3	X25
bri0/0	BRI/ISDN1	ISDN Basic Rate Int
x25-node	---	Router->Node
voip2/0	SLOT2	POTS VoIP PCI card

```
Config>
```

The default configuration depends on the type of board. Those permitting default FXS/FXO configuration are configured as **FXO**. This always displays the operating mode.

```
network voip2/0
; -- VoIP interface Configuration --
  line 1 interface-type fxs
;
  line 2 interface-type fxo
;
  line 3 interface-type fxs
;
  line 4 interface-type fxo
;
exit
```

### 4.1.4.1 Example

A voice gateway has a board with 4 FXS ports. It preferably uses the G.729 codec and executes calls with H323 using a gatekeeper associated with the GkZone. The analog telephones are numbered from 1000 to 1003. The remote telephones following the 11xx pattern.

```
log-command-errors
no configuration
set hostname VoIP-GW-POTS
set data-link x25 serial0/0
set data-link x25 serial0/1
set data-link x25 serial0/2
telephony
; -- Telephony configuration -
  dial-peer 1 voice-port
    destination-pattern 1000
    target voice-port voip2/0 1
  exit
;
  dial-peer 2 voice-port
    destination-pattern 1001
    target voice-port voip2/0 2
  exit
;
  dial-peer 3 voice-port
    destination-pattern 1002
    target voice-port voip2/0 3
```

```
exit
;
dial-peer 4 voice-port
  destination-pattern 1003
  target voice-port voip2/0 4
exit
;
dial-peer 100 h323
  codec g729
  destination-pattern 11..
  target gatekeeper
  no vad
exit
;
exit
;
network ethernet0/0
; -- Ethernet Interface User Configuration --
  ip address 192.168.1.10 255.255.255.0
;
;
;
  exit
;
network voip2/0
; -- VoIP interface Configuration --
  line 1 interface-type fxs
  line 1 telephone-number 1000
;
  line 2 interface-type fxs
  line 2 telephone-number 1001
;
  line 3 interface-type fxs
  line 3 telephone-number 1002
;
  line 4 interface-type fxs
  line 4 telephone-number 1003
;
  dsp-firmware g729-g711
exit
;
event
; -- ELS Config --
  enable trace subsystem VoIP ALL
exit
;
;
;
protocol h323
  application address 192.168.1.10
  application gateway
  gatekeeper address 192.168.1.2
  gatekeeper zone GkZone
  gw name Gwbintec
exit
;
dump-command-errors
end
```

## 4.2 BRI ISDN Interface

## 4.2.1 Description

The BRI ISDN interface is a digital interface permitting you to establish two calls over a physical interface at 64 Kbps. The calls can be either audio or data. When dealing with audio, the signal is digitalized complying with the G.711 A-law in Europe and G.711 m-law in the USA.

From a logical point of view, an ISDN interface is composed of at least three different layers: the physical layer, the link layer and the network layer. In order to execute calls, the physical interface must be up and the ISDN interface link layer established. The network layer requests and establishes the calls.

An ISDN interface can operate in user mode (TE) or network mode (NT). This behavior in conjunction with the ISDN interface norm affects the way the different layers exchange information.

In the case of the ITU-T norm, the physical layer follows the I430 norm, the link Q.921 and layer 3 Q.931.

## 4.2.2 Connector

An RJ11 connector is used in S/T interfaces with the following pin configuration:

Pin 1— Not connected

Pin 2— Not connected

Pin 3— Transmit +

Pin 4— Receive +

Pin 5— Receive -

Pin 6— Transmit -

Pin 7— Not connected

Pin 8— Not connected

Internally, depending on the operating mode, network or terminal, the wires are crossed so externally direct wiring can be used.

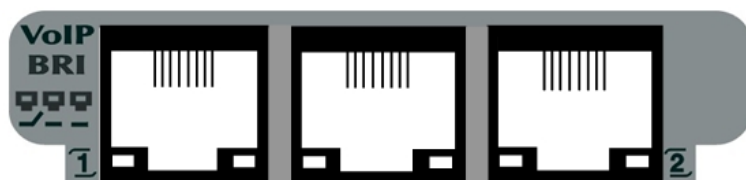
The bypass connector internally crosses the transmission and reception pins facing basic access 1, so that externally the wiring can be directly used.

## 4.2.3 Types of VoIP BRI boards

### 4.2.3.1 VoIP 2 BRI board with bypass

This board contains 2 independent basic accesses (2B+D) with an S/T interface to be used in audio calls. From a behavior point of view, both interfaces can independently operate in TE mode or NT mode; in cases of TE, these can operate by requesting TEI from the network entity or with TEI 0; in cases of NT, these only operate with TEI 0. TE mode does not provide any power so it is not possible to supply power to the terminals. By default, this operates in TE mode.

From the voice encryption point of view, this board has a signal processor which supports G.711, G.729 annex A and annex B and G.723 at 5.3 and 6.4 kbps. For performance reasons, the G.723 and G.729 codecs are not simultaneously supported.



As you can see in the above figure, there is a bypass connector between both basic access connectors, which is associated with line 1. In the serigraphy, the bypass is indicated by the relevant symbol. This bypass is active when the device is switched off. The bypass can be controlled through the voip monitoring command BYPASS. This is only available in the VoIP interface with the lowest order.

```
+network voip3/0
-- Voip network monitor --
```

```
voip3/0+bypass
Enable 1, Disable 0[0]? 1
voip3/0+
```

The bypass relay state can be monitored through the DEVICE interfaces monitoring command.

```
*p 3
Console Operator
+device voip3/0

Interface          CSR      Vect      Auto-test   Auto-test   Maintenance
                  F0000000  22        valids     failures    failures
voip3/0            F0000000  22        1          0           0

Board ID: 1724021-1541
Firmware id: 26AE09B4
N Channel: 2-2-4
Type: ISDN / BRI

Bypass: OFF
+
```

Regarding the wiring to be used, this should be a direct wire, both for the basic access connectors as well as for the bypass connector, for both NT and TE mode.

All connectors have two LEDs: one green and one yellow. These are not used in the bypass connector. For the basic access connectors, only the green LED is used on startup when the physical layer is active (UP).

## 4.2.4 Configuration

From the configuration point of view, the VoIP BRI boards are seen as two interfaces for basic access: one VoIP interface and one BRI interface. A VoIP BRI board with two basic accesses located in slot 3 is shown as displayed below:

```
Config>list device

Interface          Connector      Type of interface
ethernet0/0        LAN1           Fast Ethernet interface
serial0/0           SERIAL0/WAN1  X25
serial0/1           SERIAL1/WAN2  X25
serial0/2           SERIAL2/WAN3  X25
bri0/0             BRI/ISDN1     ISDN Basic Rate Int
x25-node           ---           Router->Node
voip3/0            SLOT3         BRI VoIP PCI card
bri3/0             SLOT3         BRI PCI card
voip3/1            SLOT3         BRI VoIP PCI card
bri3/1            SLOT3         BRI PCI card
Config>
```

Interface voip3/0 is associated with interface bri3/0. Interface voip3/1 is associated with interface bri3/1.

The VoIP interfaces support everything relevant to VoIP audio calls from the ISDN access (in this case, basic access). The BRI interface supports everything relating to the ISDN physical interface.

The VoIP interface configures all the lines with the same parameters. In this type of interface, the number of lines is determined by the base interface depending on the number of available B channels. For this reason, regarding configuration, the ISDN VoIP interfaces only allow you to configure parameters for one line.

The BRI interface is similar to other BRI interfaces used for data.

The default configuration for a VoIP BRI board with two accesses is as follows:

```
network voip3/0
; -- VoIP interface Configuration --
  base-interface
; -- Base Interface Configuration --
  base-interface bri3/0 255 link
  base-interface bri3/0 255 number-of-circuits all
;
```

```

    exit
;
exit
;
;
network voip3/1
; -- VoIP interface Configuration --
    base-interface
; -- Base Interface Configuration --
    base-interface bri3/1 255 link
    base-interface bri3/1 255 number-of-circuits all
;
    exit
;
exit
;

```

As you can see, to use the ISDN interfaces you also need to configure the base interface. In VoIP BRI boards, the base interface is wired and consequently cannot be changed. In VoIP-ISDN interfaces, an ISDN data interface is configured.

The VoIP ISDN interfaces follow the same philosophy as the dial interfaces (e.g., PPP) and additionally require a call profile configured through the “GLOBAL-PROFILE DIAL” configuration menu. For further information, please see manual bintec-Dm 732-I Dial Profile.

#### 4.2.4.1 Example

A voice gateway with two VoIP boards, one being POTS in slot 2 and the other 2BRI in slot 3, allow several analog telephones and a switchboard to access a remote environment through SIP. The PBX connects to the voice gateway through an ISDN access line with TEI equal to 0.

The analog telephones have extensions 3000 to 3002 and can access the PBX extensions by dialing 20xx. So the PBX admits the called numbers, these are changed to pattern 3841xx and the type and numeration plan in the ISDN calls are changed.

The PBX extensions have numbers 2000 to 2003 and must dial 0 to exit the basic access followed by the required number.

The remote numbers follow the 8xxx pattern and are accessed through SIP.

```

log-command-errors
no configuration
set hostname VoIP_BRI
set data-link x25 serial0/0
set data-link x25 serial0/1
set data-link x25 serial0/2
global-profiles dial
; -- Dial Profiles Configuration --
    description "voip dial"
;
    profile voip_dial default
    profile voip_dial inout
    profile voip_dial isdn-type audio
;
exit
;
telephony
; -- Telephony configuration --
    dial-peer 1 voice-port
        destination-pattern 3000
        target voice-port voip2/0 1
    exit
;
    dial-peer 2 voice-port
        destination-pattern 3001
        target voice-port voip2/0 2
    exit
;
    dial-peer 3 voice-port

```

```

    destination-pattern 3002
    target voice-port voip2/0 3
exit
;
dial-peer 4 voice-port
    destination-pattern 3002
    target voice-port voip2/0 4
exit
;
dial-peer 10 voice-port
    destination-pattern 20..
    outgoing prefix 3841
    outgoing strip-digits 2
    target voice-port voip3/0 1
exit
;
dial-peer 100 sip
    codec g729
    destination-pattern 8...
    target ipv4 192.168.1.100
    no vad
exit
;
exit
;
network voip2/0
; -- VoIP interface Configuration --
    line 1 telephone-number 3000
;
    line 2 telephone-number 3001
;
    line 3 telephone-number 3002
;
    line 4 telephone-number 3003
;
    dsp-firmware g729-g711
exit
;
network voip3/0
; -- VoIP interface Configuration --
    isdn called number-type national
    isdn called plan-type isdn/telephony
    line 1 telephone-number 2000
;
    dsp-firmware g729-g711
    base-interface
; -- Base Interface Configuration --
    base-interface bri3/0 255 link
    base-interface bri3/0 255 profile voip_dial
    base-interface bri3/0 255 number-of-circuits all
;
    exit
;
exit
;
network ethernet0/0
; -- Ethernet Interface User Configuration --
    ip address 192.168.1.10 255.255.255.0
;
;
;
    exit
;
network bri3/0
; -- BRI ISDN Configuration --
    set global emulate network
exit

```



```

;
network voip3/1
; -- VoIP interface Configuration --
  base-interface
; -- Base Interface Configuration --
  base-interface bri3/1 255 link
  base-interface bri3/1 255 number-of-circuits all
;
  exit
;
  shutdown
exit
;
network bri3/1
; -- BRI ISDN Configuration --
  set global emulate network
  shutdown
exit
;
event
; -- ELS Config --
  enable trace subsystem ISDN ALL
  enable trace subsystem VoIP ALL
  enable filter
  ev-buffer 2000 200
exit
;
;
;
;
protocol sip
  application address 192.168.1.10
  application gateway
  realm example.bintec.es
exit
;
dump-command-errors
end
; --- end ---

```

## 4.3 VoIP-ISDN Interface

### 4.3.1 Description

Strictly speaking, this is not associated with a board; it is a virtual interface which operates over a data ISDN interface in the device so audio calls can be executed over it. The behavior is the same as for a VoIP BRI or a VoIP PRI interface except it only admits G711 A-law and m-law codecs.

### 4.3.2 Configuration

This is added through the ADD DEVICE VoIP-ISDN configuration command. This command requires an interface number between 100 and 9999. By default no interface of this type has been defined.

```

*p 4

Config>ADD DEVICE VoIP-ISDN 100
Config>list device

Interface          Connector      Type of interface
ethernet0/0        LAN1           Fast Ethernet interface
serial0/0           SERIAL0/WAN1  X25
serial0/1           SERIAL1/WAN2  X25
serial0/2           SERIAL2/WAN3  X25
bri0/0             BRI/ISDN1     ISDN Basic Rate Int
x25-node           ---           Router->Node

```

```
voip100          ---          ISDN VoIP
VoIP-GK Config>
```

To configure this, access it through the interface name followed by the assigned number.

```
Config>network voip100
```

```
-- VoIP interface Configuration --
voip100 config>
```

The rest of the aspects in the said interface are configured in the same way as any other VoIP ISDN interface: you need to configure the base interface, indicating which one it is, what profile it will use to execute and receive calls and what is the maximum number of channels to use.

The number of calls that can be executed depend on the number of available channels: on the one hand depending on how the basic access is used by other dial interfaces, and on the other, depending on the number of channels available when configuring the base interface.

#### 4.3.2.1 Example

We want to use the motherboard basic access in an access router to execute audio and data calls. Two PPP interfaces carry out the data calls to number 384840. The BRI interface number is 384820.

The VoIP-ISDN interface number is 4000 and executes calls over the ISDN interface substituting the 4 for 384. The received calls are routed to 200x. The VoIP profile is used to control the calls and the maximum number of audio calls that can be executed is one.

```
log-command-errors
no configuration
set hostname VoIP-ISDN-SIP
add device ppp 1
add device ppp 2
add device voip-isdn 100
set data-link x25 serial0/0
set data-link x25 serial0/1
set data-link x25 serial0/2
global-profiles dial
; -- Dial Profiles Configuration --
  profile Datos1 default
  profile Datos1 inout
  profile Datos1 remote-address 384840
  profile Datos1 code sylvia
  profile Datos1 idle-time 300
;
  profile Datos2 default
  profile Datos2 inout
  profile Datos2 remote-address 384840
  profile Datos2 code alice
  profile Datos2 idle-time 300
;
  profile VoIP default
  profile VoIP inout
  profile VoIP isdn-type audio
;
exit
;
telephony
; -- Telephony configuration --
  dial-peer 100 voice-port
    destination-pattern 4...
    outgoing prefix 384
    outgoing strip-digits 1
    target voice-port voip100 1
  exit
;
  dial-peer 2000 voice-port
    incoming called number 3848..
    incoming prefix 200
    incoming strip-digits 5
```

```
exit
;
dial-peer 200 sip
  destination-pattern 2...
  target ipv4 192.168.1.100
exit
;
exit
;
network ethernet0/0
; -- Ethernet Interface User Configuration --
  ip address 192.168.1.10 255.255.0.0
;
;
;
  exit
;
network bri0/0
; -- BRI ISDN Configuration --
  set global local-address 384820
exit
;
network pppl
; -- Generic PPP User Configuration --
  ip address 192.168.10.1 255.255.255.0
;
;
  base-interface
; -- Base Interface Configuration --
  base-interface bri0/0 255 link
  base-interface bri0/0 255 profile Datos1
;
  exit
;
exit
;
network ppp2
; -- Generic PPP User Configuration --
  ip address 192.168.20.1 255.255.255.0
;
;
  base-interface
; -- Base Interface Configuration --
  base-interface bri0/0 255 link
  base-interface bri0/0 255 profile Datos2
;
  exit
;
exit
;
network voip100
; -- VoIP interface Configuration --
  line 1 telephone-number 4000
;
  base-interface
; -- Base Interface Configuration --
  base-interface bri0/0 255 link
  base-interface bri0/0 255 profile VoIP
  base-interface bri0/0 255 number-of-circuits 1
;
  exit
;
exit
;
event
; -- ELS Config --
  enable trace subsystem ISDN ALL
```

```

enable trace subsystem VoIP ALL
enable filter
ev-buffer 1000 200
exit
;
;
;
protocol sip
  application address 192.168.1.10
  application gateway
  realm example.bintec.es
exit
;
dump-command-errors
end

```

## 4.4 PRI ISDN Interface

### 4.4.1 Description

The PRI ISDN interface is a digital interface permitting you to establish multiple calls over a physical interface, at a rate of 64 or 56 Kbps. The number of calls depends partly on the type of interface (up to 30 calls in cases of an E1 interface and up to 23 for a T1 interface) and partly on the number of time-slots that are reserved for the calls. Calls can be audio or data. In cases of audio, the signal is digitalized complying with the G.711 A law norm in Europe and the G.711 m law in the USA.

In order to carry out calls, you need an establishment mechanism for them. These mechanisms can be based on different norms such as ITU-T Q.931, ECMA QSIG or R2 (ITU-T series Q.400).

Each of these norms establishes a series of protocols that must establish peer-to-peer as a necessary requirement to establish, maintain and release a call. Additionally, depending on the type of norm, the ends can have a symmetric behavior or have to distinguish between network mode (NT) or terminal mode (TE) behavior.

### 4.4.2 Connectors

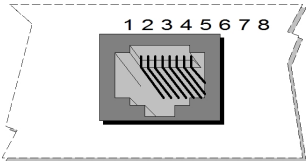
There are two connector formats: RJ45 and coaxial. Do not use both formats simultaneously or the link cannot establish properly. The impedances can be selected through configuration in the base g703port interface. For further information, please see manual bintec-Dm 746-I G703 Interface.

#### 4.4.2.1 RJ45 Connector

This is normally used in unbalanced connections with an impedance of 120 Ohms.

The pins are as follows:

##### Numeration contacts in the RJ45 connector

RJ45 connector	RJ45	Signal
	4	Tx+
	1	Rx+
	2	Rx-
	5	Tx-

From the device point of view, the TX signals are considered outgoing and the RX incoming.

#### 4.4.2.2 Coaxial connector

This is normally used in balanced connections with impedance of 75 Ohms

The connector format used is DIN BNC screw thread 5.6/1.6.

Looking from the front, the transmission connector is situated on the left and the transmission connector on the right.

## 4.4.3 Boards

### 4.4.3.1 VoIP PRI board

This contains a primary access. From the behavior point of view, this can operate as E1 with ITU-T Q.931 (both in user format as well as in network format), QSIG and R2 signaling. This only supports audio calls.

From the voice encoding point of view, this board has three signaling processors that support G.711, G.729 annex A and annex B and G.723 at 5.3 and 6.4 kbps.



By default, the interface is configured to operate as E1 without specifying any signaling or the number of channels assigned to calls.

## 4.4.4 Configuration

From the configuration point of view, the VoIP PRI boards are seen as a VoIP interface and a G703 interface. A VoIP PRI board situated in slot 3 is shown as follows:

```
Config>list device

Interface          Connector      Type of interface
-----
ethernet0/0        FE0/LAN1      Fast Ethernet interface
ethernet0/1        FE1/LAN2      Fast Ethernet interface
serial0/0          SERIAL0/WAN1  X25
bri0/0             BRI/ISDN1     ISDN Basic Rate Int
x25-node           ---           Router->Node
voip3/0            SLOT3         PRI VoIP PCI card
g703port3/0        SLOT3         G703 E1/T1 PCI cardConfig>
```

The voip3/0 interface is associated with the g703port3/0 interface. The VoIP interface supports everything related to VoIP audio calls from the ISDN access (in this case the primary access). The G703 interface supports everything related to the ISDN physical interface itself.

The VoIP interface configures all the lines with the same parameters. In this type of interface, the number of lines is determined by the base interface depending on the number of available B channels. For this reason from the configuration point of view, the ISDN VoIP interfaces only allow you to configure one-line parameters.

The G703 interface is similar to other types of G703 interfaces used for data. Here you configure things relating to the physical layer (clock sources, impedances, etc.), the channels available for the VoIP interface and parameters relating to the type of signaling used to execute the calls. For further information, please see manual bintec-Dm 746-I G703 Interface.

The default configuration for a VoIP PRI board is as follows:

```
network voip3/0
; -- VoIP interface Configuration --
  base-interface
; -- Base Interface Configuration --
  base-interface g703port3/0 255 link
  base-interface g703port3/0 255 number-of-circuits all
;
  exit
;
exit
```

Where there are no channels configured to make calls nor has any signaling been specified.

As you can see, the ISDN PRI interfaces also require the configuration of the base interface. In VoIP PRI boards, the base interface is wired and consequently cannot be changed. In VoIP-ISDN interfaces you configure a data ISDN interface.

The VoIP-ISDN interfaces follow the same philosophy as the dial interfaces (e.g., PPP) and additionally require a call profile, configured through the “GLOBAL-PROFILE DIAL” configuration menu. For further information, please see manual bintec-Dm 732-I Dial Profile.

*Example:*

A corporation has an IP telephony network based on SIP telephones and an IR-VOZ call manager. In order to route calls to public network fixed numbers a primary media-gateway is used with ITU-T Q.931 signaling. The configuration for this is as follows:

```
log-command-errors
no configuration
;
global-profiles dial
; -- Dial Profiles Configuration --
  profile VoIP default
  profile VoIP inout
  profile VoIP isdn-type audio
;
exit
;
telephony
; -- Telephony configuration --
  tones err-timeout 2m
  dial-peer 10 voice-port
    destination-pattern 9.....
    incoming prefix *
    target voice-port voip2/0 1
    no vad
  exit
;
  dial-peer 20 sip
    codec g729
    destination-pattern *T
    outgoing strip-digits 1
    target ipv4 172.1.1.5
    no vad
  exit
;
exit;
;
network ethernet0/0
; -- Ethernet Interface User Configuration --
  ip address 172.1.1.100 255.255.0.0
;
exit
;

network voip2/0
; -- VoIP interface Configuration --
;
  base-interface
; -- Base Interface Configuration --
  base-interface g703port2/0 255 link
  base-interface g703port2/0 255 profile VoIP
  base-interface g703port2/0 255 number-of-circuits all
;
  exit
;
exit
;
network g703port2/0
; -- T1E1 / G703 Configuration --
  pri-group timeslots 1-31
exit
;
;
protocol ip
```

```

; -- Internet protocol user configuration --
  route 0.0.0.0 0.0.0.0 172.1.1.1
;
exit
;
;
protocol sip
  application address 172.1.1.100
  application gateway
  realm madrid.bintec.es
exit
;
dump-command-errors
end
; --- end ---

```

With this configuration, everything beginning with 9 and containing 9 digits is sent over the primary interface. Calls received by the primary are processed with dial-peer 10, which adds an asterisk to the call so it is routed by dial-peer 20; once this dial-peer has been selected as outgoing, it removes the asterisk and sends the call to the IR-VOZ call-manager. If this process is not followed, all the received calls are routed over the primary interface again.

## 4.5 E&M Interface

### 4.5.1 Description

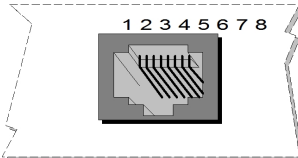
An E&M interface is an analog interface where you can distinguish between the link side and the signaling side. This is characterized by the fact that the audio goes through one side (in a 2 or 4 wire format) and the hook-on/hook-off signaling goes through the other.

Two signals are used to indicate the hook-on/hook-off state, E and M, which each end presents and detects. These signals are based on continuous currents. Depending on the number of cables needed to present these signals and where the batteries and the ground connections are located, we have different types of E&M interfaces: in total there are 5 different ones, identified by their cardinals.

To execute a call over an E&M interface, in addition to knowing whether the other end is hook-on or hook-off, you also need to know if it is available to receive called number dialing. Depending on how the call receiver indicates if this is available or not, we have three types of possible signaling: *immediate*, the receiver is always available, consequently the caller can dial the number as soon as he/she picks up the phone; *wink-start*, the receiver indicates it is available once a “wink” pulse has been given to the caller; *delay*, the receiver gives a variable delay pulse while it is not available. The *wink* and *delay* pulses are given by the receiver hooking-off and hooking-on; the difference between the *wink* and *delay* pulse is the length, i.e., *wink* has a fixed duration, while *delay* depends on the degree of caller occupation.

### 4.5.2 Connector

The connector used is an RJ45. The pins are as follows:

RJ45 connector	RJ45	2 wire signals	4 wire signals
	7	E	E
	2	M	M
	6	---	T
	3	---	R
	5	T	T1
	4	R	R1

### 4.5.3 Types of VoIP E & M boards

### 4.5.3.1 VoIP 2 E & M board

The VoIP E&M interface is an analog interface permitting the connection of telephone switchboards to IP networks. This board has two accesses which operate as units on the signaling side. These allow you to configure the type of E&M interface, the number of cables to be used for audio and the type of signaling.

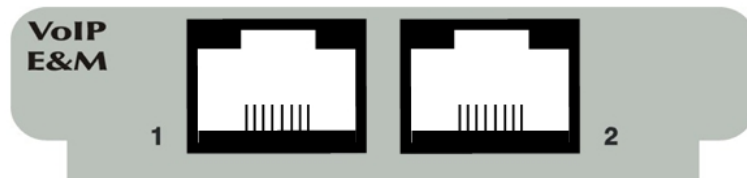
E&M types supported are I, II, III and V. Of these II and V permit unit interconnection on the signaling side.

In order to send audio, the 4 wire and 2 wire modalities are supported. Each transmission direction in the 4 wire modality uses 2 wires. The 2 wire modality requires systems which pass from 2 wire to 4 wire to be used, such as hybrid coils and echo cancellation.

With respect to signaling, the following types are supported: *immediate*, *wink-start*, *delay* and *always-on*. The first three are standard E&M interface signals. *Always-on* signaling is such that the access immediately hooks-off without taking the M pin (pin 2) state into account and maintains this state indefinitely. This is useful as it establishes permanent audio links in services such as Land Mobile Radio (LMR). With *always-on* signaling, you can remotely send and receive the M and E signals respectively. These signals are sent as RTP traffic complying with the RFC 2833 standard.

The lines are semi-independent so the type of E&M and the number of cables to be used in the audio should be the same. The rest of the parameters are independent.

From the voice codification point of view, this board has a signal processor which supports G.711, G.729 annex A and annex B and G.723 at 5.3 and 6.4 kbps. Codes G.723 and G.729 are not simultaneously supported due to performance reasons.



### 4.5.4 Configuration

From the configuration point of view, the VoIP E&M boards are presented as a single voip interface where you can configure each line independently.

```
Config>LIST DEVICE

Interface          Connector      Type of interface
ethernet0/0        LAN1           Fast Ethernet interface
serial0/0           SERIAL0/WAN1  X25
serial0/1           SERIAL1/WAN2  X25
serial0/2           SERIAL2/WAN3  X25
bri0/0             BRI/ISDN1     ISDN Basic Rate Int
x25-node           ---           Router->Node
voip2/0            SLOT2         E&M VoIP PCI card
Config>
```

By default, the lines are configured as type 1, wink-start signaling and 2 wires.

#### Example 1:

A voice *gateway* routes calls received from a central over IP using H323. Connection to the PBX is carried out through a 4-wire V-type E&M line and with *wink-start* signaling. All calls to telephone numbers beginning with 3 and 4 containing 5 digits are routed over IP to the configured *gatekeeper*.

The configuration is as follows:

```
log-command-errors
no configuration
set hostname EnM-GW
;
telephony
; -- Telephony configuration --
dial-peer 1 voice-port
    codec g729
    destination-pattern 200..
```



```

    target voice-port voip1/0 1
    no vad
  exit
;
dial-peer 100 h323
  codec g729
  destination-pattern 3....
  destination-pattern 4....
  target gatekeeper
  no vad
  exit
;
exit
;
;
network ethernet0/0
; -- Ethernet Interface User Configuration --
  ip address 172.1.1.100 255.255.0.0
;
exit
;
network voip1/0
; -- VoIP interface Configuration --
  line 1 e&m-operation 4-wires
  line 1 e&m-type type-v
;
  line 2 no active
  line 2 e&m-operation 4-wires
  line 2 e&m-type type-v
;
  dsp-firmware g729-g711
exit
;
protocol h323
  application address 172.1.1.100
  application gateway
  gatekeeper address 172.1.1.10
  gatekeeper zone CENTRAL_GK
  gw name bintec_GW
  gw tech-prefix 1#
exit
;
dump-command-errors
end
; --- end ---

```

**Example 2:**

Test in loop mode for the two VoIP E&M board accesses. The assembly consists of our router where there is a VoIP POTS card in slot 2 with two telephones and a VoIP E&M card in slot 3. This latter is configured as type 2 so both lines can be wired using an appropriate crossover cable.

The configuration is such that when picking up the telephone on line 1 and dialing 21, a call is made to the E&M board line 1. This changes the call to 32 and externally routes the call. As this an external loop, the call is received on line 2 in the same board which carries out a new internal call to the telephone on line 2. Consequently, two conversations and audio between the two telephones in the setup are established through a single call.

```

log-command-errors
no configuration
set hostname EnM-IMM-T2-4W
set data-link x25 serial0/0
set data-link x25 serial0/1
set data-link x25 serial0/2
telephony
; -- Telephony configuration --
  dial-peer 31 voice-port
    destination-pattern 31
    target voice-port voip2/0 1
  exit

```

```

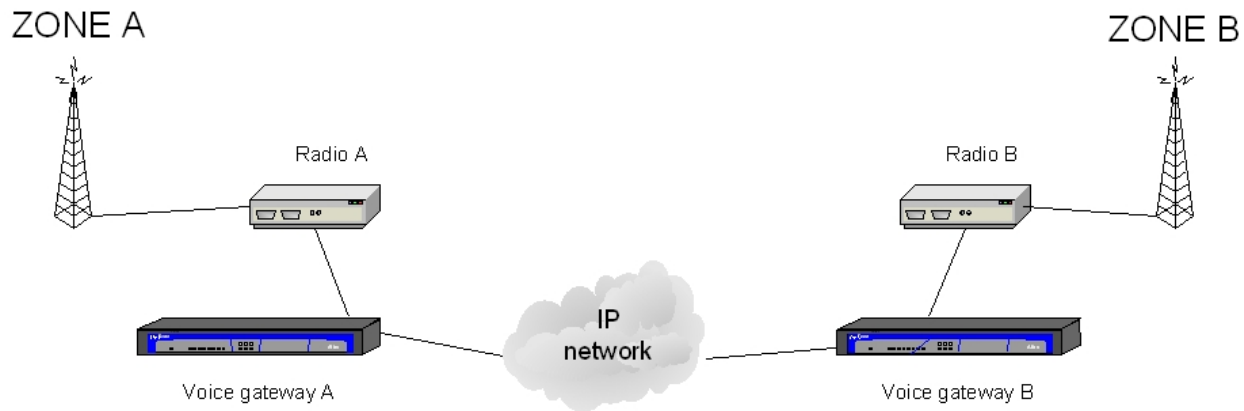
;
dial-peer 32 voice-port
  destination-pattern 32
  target voice-port voip2/0 2
exit
;
dial-peer 21 voice-port
  destination-pattern 21
  outgoing prefix 32
  outgoing strip-digits 2
  target voice-port voip3/0 1
exit
;
dial-peer 22 voice-port
  destination-pattern 22
  target voice-port voip3/0 2
exit
;
exit
;
;
network voip2/0
; -- VoIP interface Configuration --
  line 1 interface-type fxs
  line 1 telephone-number 31
;
  line 2 interface-type fxs
  line 2 telephone-number 32
;
  line 3 interface-type fxo
;
  line 4 interface-type fxo
;
exit
;
network voip3/0
; -- VoIP interface Configuration --
  line 1 telephone-number 21
  line 1 e&m-operation 4-wires
  line 1 e&m-type type-ii
  line 1 e&m-signal immediate
;
  line 2 e&m-operation 4-wires
  line 2 e&m-type type-ii
  line 2 telephone-number 22
  line 2 e&m-signal immediate
;
exit
;
event
; -- ELS Config --
  enable trace subsystem VoIP ALL
  enable filter
  filter 1 default
  filter 1 text "E&M"
  filter 1 action red
  filter 2 default
  filter 2 text "voip3"
  filter 2 action blue
exit
;
dump-command-errors
end
; --- end ---

```

**Example 3:**

You wish to extend two radiotelephony networks over IP so that the radio devices in zone A can access the radio

devices in zone B and vice versa. The setup is as shown in the following figure:



To connect the radio devices to the voice *gateway*, you use a 2-wire E&M interface so that the E and M signals connect to the radio devices: the E signal acts on the connection station's PTT and the base station's carrier detection circuit (CD) activates the M signal when there is audio.

As no call is required and the E and M signals must be remotely sent, *always-on* signaling is configured on the E&M line and the transmission of them is enabled. Type II is used due to the absence of DC connectivity between the radio device and the *gateway* in order to avoid ground loops being formed.

Audio is sent in G711 A-law to maximize its quality and the vad is enabled so audio is sent only when present. You need to have the call permanently established and execute signal *relay* for which a direct dial-peer is used, and in this case the RTP port to be used in the communication is set. For further information on this type of dial-peer, please see manual bintec-Dm 722-I Telephony over IP.

The configuration for the voice *gateway* associated with zone A is as follows:

```
log-command-errors
no configuration
set hostname ZONE_A
telephony
; -- Telephony configuration --
dial-peer 10 direct
  codec g711alaw
  destination-pattern 100
  local-ip 172.1.1.100
  rtp port-exclusive 20000
  target ipv4 172.10.1.100
exit
;
exit
;
network ethernet0/0
; -- Ethernet Interface User Configuration -
ip address 172.1.1.100 255.255.0.0
;
exit
;
network voip1/0
; -- VoIP interface Configuration --
line 1 direct-dialing 100 0s
line 1 e&m-signal always-on
line 1 e&m-type type-ii
line 1 e&m-lead-relay all
;
line 2 no active
line 2 e&m-operation 4-wires
line 2 e&m-type type-ii
;
exit
;
protocol ip
; -- Internet protocol user configuration --
route 0.0.0.0 0.0.0.0 172.1.1.1
;
```

```

exit
;
dump-command-errors
end
; --- end ---

```

Due to line 1's configuration, right after the device starts up line 1 hooks-off and immediately dials 100 which establishes a call through *dial-peer* 10. This *dial-peer* doesn't use any kind of signaling protocol so it immediately sends the audio it receives from the line. It also sends the M signal state through RTP, complying with the RFC 2833 standard, and updates the E signal state in the line depending on what is received from the remote end.

The configuration of the voice gateway associated with zone B is similar, with only the IP addresses changing.

## 4.6 VoIP-DUMMY Interface

### 4.6.1 Description

This interface is not associated with a board, but rather is a virtual interface. It allows you to make and receive calls so that once established, a voice message stored in the device is reproduced. The number of audio channels, i.e., simultaneous calls supported, are configured using the **channels** command.

You can configure a voice message for each of the supported codecs so that this is selected depending on the codec negotiated on establishing the call. If there isn't a voice message available for the selected codec, the call is established and nothing is reproduced.

The voip-dummy interface can be used to reproduce alert voice messages for certain numbers with the possibility of forwarding the call to another number when the message ends (e.g., number not permitted, number currently unavailable due to problems in the network, welcome voice message, etc.), to implement an automated answering system or IVR (see manual bintec-Dm 722-I Telephony over IP section 2.8), or to generate music on hold for an IP multicast (see section 2.2.1 in manual bintec-Dm 722-I Telephony over IP). It is also useful for management and maintenance tasks thereby permitting you to make and receive calls without needing one of the conversation's interlocutors.

### 4.6.2 Configuring

This is added through the ADD DEVICE VoIP-DUMMY configuration command. This command requires an interface number between 1000 and 1999. By default no interface of this type has been defined.

```

*p 4

Config>ADD DEVICE VoIP-DUMMY 1000
Config>list device

Interface          Connector      Type of interface
ethernet0/0        LAN1           Fast Ethernet interface
serial0/0           SERIAL0/WAN1  X25
serial0/1           SERIAL1/WAN2  X25
serial0/2           SERIAL2/WAN3  X25
bri0/0             BRI/ISDN1     ISDN Basic Rate Int
x25-node           ---           Router->Node
voip1/0            SLOT1         POTS VoIP PCI card
voip2/0            SLOT2         E&M VoIP PCI card
voip1000           ---           Dummy VoIP
Config>

```

To configure this, access through the interface name followed by the assigned number.

```

Config>network voip1000

-- VoIP interface Configuration --
voip1000 config>

```

Allows you to configure the behavior and the audio messages to be played depending on the codec selected in the call.

```

voip1000 config>line 1 ?
  active          Line can receive and make calls
  behavior        VoIP-dummy audio behavior

```

```

default          Default line configuration
direct-dialing   Call is automatically carried out to this number
no              Negates a command or sets its defaults
play            Play a voice-message in a dummy ifc
telephone-number Configure the telephone number associated to the line
timing          Configure timing parameters
voip1000 config>

```

The voip-dummy interface admits the following operating modes:

```

voip1000 config>line 1 behavior ?
play-loop        play an audio message in a loop
play-disc       play an audio message and disconnect
play-fwd        play an audio message and forward
play-ivr        play an audio message and wait for DTMF
voip1000 config>

```

**play-loop:** continuously reproduce the audio recording in a loop.

**play-disc:** reproduce the audio recording once only and end the call.

**play-fwd:** reproduce the audio recording once only and forward the call to the configured number.

**play-ivr:** plays the audio recording once and transfers the call according to the DTMF digits received.

You can configure the voice message to use depending on the codec negotiated in the call. If there isn't an available message, the call is established but nothing is reproduced.

```

voip1000 config>line 1 play ?
G711    Voice-message coded as G.711 samples
G723    Voice-message coded as G.723 frames
G729    Voice-message coded as G.729 frames
voip1000 config>line 1 play G729 ?
voice-message  Play a voice message
voip1000 config>line 1 play G729 voice-message 1
voip1000 config>

```

To delete this type of interface, use the NO DEVICE VoIP command together with the interface number.

See manual bintec-Dm 722-I Telephony Over IP for more examples and information on the functioning of IVR.

#### Example:

In a scenario where the router is acting as an H.323 Gateway with two FXS extensions, 100 and 101, configure a voip-dummy interface to make and receive test calls with extension 120 as well as to tell the caller that line 2 (telephone 101) either does not answer or is busy. If you call 101 and it is busy, the call jumps to dial-peer 15 and the voice message is produced. You can also call from the virtual extension 120 through the **call-make** monitoring command found in the telephony menu. Please see manual bintec-Dm 722-I Telephony over IP. The same message is recorded in two different codecs, G.711 and G.729, as both of them can negotiate through H.323.

```

log-command-errors
no configuration
set hostname H323-LOCAL
add device voip-dummy 1000
set data-link x25 serial0/0
set data-link x25 serial0/1
set data-link x25 serial0/2
telephony
; -- Telephony configuration --
dial-peer 10 voice-port
    destination-pattern 100
    target voice-port voip1/0 1
exit
;
dial-peer 11 voice-port
    destination-pattern 101
    target voice-port voip1/0 2
exit
;
dial-peer 100 h323
    destination-pattern 2..

```

```

        target ipv4 172.24.78.96
    exit
;
dial-peer 20 voice-port
    destination-pattern 120
    target voice-port voip1000 1
    exit
;
dial-peer 15 voice-port
    destination-pattern 101
    target voice-port voip1000 1
    exit
;
exit
;
network ethernet0/0
; -- Ethernet Interface User Configuration --
    ip address 172.24.78.94 255.255.0.0
;
;
;
    exit
;
network voip1/0
; -- VoIP interface Configuration --
    dsp-firmware g729-g711
    line 1 interface-type fxs
    line 1 telephone-number 100
;
    line 2 interface-type fxs
    line 2 telephone-number 101
    line 2 forward-no-answer 120
;
    line 3 interface-type fxo
;
    line 4 interface-type fxo
;
exit
;
network voip1000
; -- VoIP interface Configuration --
    line 1 telephone-number 120
    line 1 behavior play-disc
    line 1 play G711 voice-message 1
    line 1 play G729 voice-message 3
;
exit
;
event
; -- ELS Config --
    enable trace subsystem H323 ALL
    enable trace subsystem VoIP ALL
exit
;
;
;
protocol h323
    application address 172.24.78.94
    application gateway
exit
;
dump-command-errors
end
; --- end ---

```

## 4.7 VoIP-CELLULAR Interface

## 4.7.1 Description

This is a virtual interface that operates over a data CELLULAR interface in the device so that audio calls can be executed over it. A CELLULAR interface is a 3<sup>rd</sup> generation mobile terminal. For further information, please see manual bintec-Dm 781-I Cellular Interface.

The behavior of a VoIP-CELLULAR interface is similar to that of a VoIP-ISDN interface, in the sense that it makes audio calls over a data interface, it only supports the G711 A-law codec and only one audio call is allowed per interface.

To make audio calls, it's essential that the board supporting the data CELLULAR interfaces has a 3<sup>rd</sup> Generation telephony module that supports them. To identify the telephony module on your board, use the LIST command found in the CELLULAR interface monitoring menu.

```
+network cellular1/0
cellular1/0 AT+lis
  Daughter Board           = CELLULAR PCI card
  Module Manufacturer      = Sierra Wireless, Inc.
  Module Model             = MC8775V
  Module Firmware         = H1_1_8_3MCAP C:/WS/FW/H1_1_8_3MCAP/MSM6280/SRC 2007/03/08 18:17:
  IMEI                    = 987654321098765
  IMSI                    = 112233445566777
  SIM Card ID              = 9911223344
  Drop by ping failed      = 0
  Drop by tracert failed   = 0
  Drop by traffic failed   = 0
  Dialers registered       = VoIP
  Current dialer registered = none
  State                   = (1) DISCONNECT
  Call request             = 1
  Telephone number         = 918076565;
  Total connection time    = 17 seconds
  Current connection time  = 0 seconds
  Time to establish connection = 1 sec
cellular1/0 AT+
```

It's also necessary that the carrier SIM permits audio calls. If it doesn't, when you try and make calls an error message appears.

## 4.7.2 Configuration

This is added through the **ADD DEVICE VoIP-CELLULAR** configuration command. This command requires an interface number between 2000 and 2999. By default no interface of this type has been defined.

```
*p 4

Config>ADD DEVICE VoIP-CELLULAR 2000
Config>list device

Interface      Connector      Type of interface
ethernet0/0    LAN1           Fast Ethernet interface
serial0/0       SERIAL0/WAN1   X25
serial0/1       SERIAL1/WAN2   X25
serial0/2       SERIAL2/WAN3   X25
bri0/0         BRI/ISDN1      ISDN Basic Rate Int
x25-node        ---            Router->Node

cellular1/0    SLOT1          AT COM
cellular1/1    SLOT1          AT COM

voip2000      ---            Cellular VoIP

Config>
```

To configure this, access through the interface name followed by the assigned number.

```
Config>network voip2000
```

```
-- VoIP interface Configuration --
voip2000 config>
```

As regards all other aspects, this interface is configured as any VoIP ISDN interface is: you have to configure the base interface, indicating which one it is and which profile is used to make and receive the calls.

You can only use the cellularx/0 as a base interface to execute audio calls. The cellularx/1 interface will produce errors if you try and route calls over it.

With one VoIP-CELLULAR interface you can only make one audio call.

In order to process audio calls, it's also necessary to configure the PIN associated with the active SIM, the telephone number associated with the SIM and enable the circuit switch mode in the base interface. The telephone number configured here is that used in TELEPHONY as "called number" to route the calls. For further information, please see manuals bintec-Dm 781-I Cellular Interface and bintec-Dm 722-I Telephony over IP.

#### 4.7.2.1 Example

You wish to make simultaneous GSM calls and data calls in an access router over a carrier's mobile network. The audio calls are made from telephones connected to a POTS board. The incoming calls are only routed to line 1 on the POTS board.

The SIM telephone number is 9911223344 and the PIN 1234. The data calls are made by a PPP interface. The carrier connection data for the data calls is: APN, "internet-apn"; user, "myname"; password "mypwd". With this data, the device configuration is as follows:

```
log-command-errors
no configuration
set hostname HSDPA-ROUTER
add device ppp 1
add device voip-cellular 2000
set data-link at cellular1/0
set data-link at cellular1/1
global-profiles dial
; -- Dial Profiles Configuration --
  profile HSDPA default
  profile HSDPA dialout
  profile HSDPA 3gpp-apn internet-apn
  profile HSDPA idle-time 300
;
  profile VoIP default
  profile VoIP inout
;
exit
;
telephony
; -- Telephony configuration --
  dial-peer 10 voice-port
    destination-pattern 6.....
    target voice-port voip2000 1
  exit
;
  dial-peer 20 voice-port
    destination-pattern 9911223344
    target voice-port voip2/0 1
  exit
;
exit
;
network cellular1/0
; -- Interface AT. Configuration --
  always-attached
  local-address 9911223344
  pin plain 1234
  network domain cs+ps
exit
;
network cellular1/1
; -- Interface AT. Configuration -
```



```

lcp-options cellular1/1 default
lcp-options cellular1/1 acfc
lcp-options cellular1/1 pfc
lcp-options cellular1/1 accm a0000
exit
;
network voip2/0
; -- VoIP interface Configuration --
  line 1 interface-type fxs
;
  line 2 interface-type fxs
;
  line 3 interface-type fxs
;
  line 4 interface-type fxs
;
exit
;
network pppl
; -- Generic PPP User Configuration --
  ppp
; -- PPP Configuration --
  authentication sent-user myname password mypwd
  ipcp local address assigned
  no ipcp peer-route
  lcp echo-req off
  exit
;
  base-interface
; -- Base Interface Configuration --
  base-interface cellular1/1 link
  base-interface cellular1/1 profile HSDPA
;
  exit
;
exit
;
;
network voip2000
; -- VoIP interface Configuration --
  base-interface
; -- Base Interface Configuration --
  base-interface cellular1/0 link
  base-interface cellular1/0 profile VoIP
;
  exit
;
exit
;
event
; -- ELS Config --
  enable trace subsystem PPP ALL
  enable trace subsystem AT ALL
  enable trace subsystem VoIP ALL
  enable filter
  ev-buffer 3000 200
exit
;
;
protocol ip
; -- Internet protocol user configuration --
  address ethernet0/0 172.24.78.94 255.255.0.0
  address pppl unnumbered
;
;
  route 0.0.0.0 0.0.0.0 pppl
;

```

```

classless
;
;
exit
;
;
;
dump-command-errors
end

```

If the telephone on line 1 makes a call to 918076565, the traces are as follows:

```

04/18/07 16:50:50 VoIP.003 Ev DTMF RCV 9 ln voip2/0-1
04/18/07 16:50:50 VoIP.004 State DIALING ln voip2/0-1
04/18/07 16:50:50 VoIP.003 Ev DTMF RCV 1 ln voip2/0-1
04/18/07 16:50:52 VoIP.003 Ev DTMF RCV 8 ln voip2/0-1
04/18/07 16:50:52 VoIP.003 Ev DTMF RCV 0 ln voip2/0-1
04/18/07 16:50:52 VoIP.003 Ev DTMF RCV 7 ln voip2/0-1
04/18/07 16:50:53 VoIP.003 Ev DTMF RCV 6 ln voip2/0-1
04/18/07 16:50:53 VoIP.003 Ev DTMF RCV 5 ln voip2/0-1
04/18/07 16:50:53 VoIP.003 Ev DTMF RCV 6 ln voip2/0-1
04/18/07 16:50:53 VoIP.003 Ev DTMF RCV 5 ln voip2/0-1
04/18/07 16:50:53 VoIP.004 State CALLING ln voip2/0-1
04/18/07 16:50:53 VoIP.003 Ev INCOMMING CALL ln voip2000-1
04/18/07 16:50:53 VoIP.010 DIAL: cmd CALL MAKE ln voip2000-1
04/18/07 16:50:53 AT.020 Dialer VoIP command CIR_OPEN[0] local st 1 intf cellular1/0
04/18/07 16:50:53 AT.020 Connection request on intf cellular1/0
04/18/07 16:50:53 VoIP.012 DIAL: out start ln voip2000-1
04/18/07 16:50:53 VoIP.004 State WTG HOOK OFF ln voip2000-1
04/18/07 16:50:53 AT.020 ATCMD-->AT+CBST=0,0,1 intf cellular1/0
04/18/07 16:50:53 AT.020 ATCMD-->ATD918076565; intf cellular1/0
04/18/07 16:50:53 VoIP.011 DIAL: ev ALERTING ln voip2000-0
04/18/07 16:50:53 AT.020 Event up CIR_ALERTING[8] local st 28 intf cellular1/0
04/18/07 16:50:53 AT.006 Dial Req Call, addr = 918076565; intf cellular1/0
04/18/07 16:50:53 VoIP.003 Ev INFO ln voip2000-1
04/18/07 16:50:53 VoIP.004 State ALERTING ln voip2000-1
04/18/07 16:50:53 VoIP.012 DIAL: out AUDIO ESTABLISHED ln voip2000-1
04/18/07 16:50:53 VoIP.004 State CALL ESTABLISHED ln voip2000-1
04/18/07 16:50:53 VoIP.003 Ev ESTABLISH ln voip2/0-1
04/18/07 16:50:53 VoIP.004 State CALL ESTABLISHED ln voip2/0-1
04/18/07 16:51:04 AT.020 ATRES--> CONNECT intf cellular1/0
04/18/07 16:51:04 VoIP.011 DIAL: ev OPEN ln voip2000-0
04/18/07 16:51:04 AT.020 Event up CIR_OPEN[2] local st 28 intf cellular1/0
04/18/07 16:51:04 AT.007 Out Call 918076565; establ intf cellular1/0
04/18/07 16:51:04 VoIP.003 Ev HOOK OFF ln voip2000-1
04/18/07 16:51:04 VoIP.003 Ev ESTABLISH ln voip2/0-1
04/18/07 16:51:18 AT.020 ATRES--> DISCONNECT intf cellular1/0
04/18/07 16:51:18 AT.009 Clear Call 918076565; release intf cellular1/0
04/18/07 16:51:19 VoIP.003 Ev HOOK ON ln voip2/0-1
04/18/07 16:51:19 VoIP.004 State IDLE ln voip2/0-1
04/18/07 16:51:19 VoIP.003 Ev RELEASE ln voip2000-1
04/18/07 16:51:19 VoIP.010 DIAL: cmd CALL END ln voip2000-1
04/18/07 16:51:19 AT.020 Dialer VoIP command CIR_CLOSE[1] local st 7 intf
cellular1/0
04/18/07 16:51:19 AT.020 Disconnection request on intf cellular1/0
04/18/07 16:51:19 VoIP.004 State WTG IDLE ln voip2000-1
04/18/07 16:51:19 AT.020 ATCMD-->ATH intf cellular1/0
04/18/07 16:51:19 AT.020 Hangup successful intf cellular1/0
04/18/07 16:51:19 VoIP.011 DIAL: ev CLOSED ln voip2000-0
04/18/07 16:51:19 AT.020 Event up CIR_CLOSED[1] local st 19 intf cellular1/0
04/18/07 16:51:19 VoIP.003 Ev HOOK ON ln voip2000-1
04/18/07 16:51:19 VoIP.004 State IDLE ln voip2000-1
04/18/07 16:51:19 VoIP.010 DIAL: cmd CALL END ln voip2000-1
04/18/07 16:51:19 VoIP.012 DIAL: no dial available ln voip2000-1

```

## Chapter 5 Appendix

### 5.1 Echo Compensation

Some Voice over IP installations where analog Voice Kits are used (e.g., with FXS/FXO interfaces) can cause echoes. This is due to the fact a conversion from two to four wires is carried out through a hybrid coil where impedance adjustment is not perfect. The echo source is displayed in Figure 1.

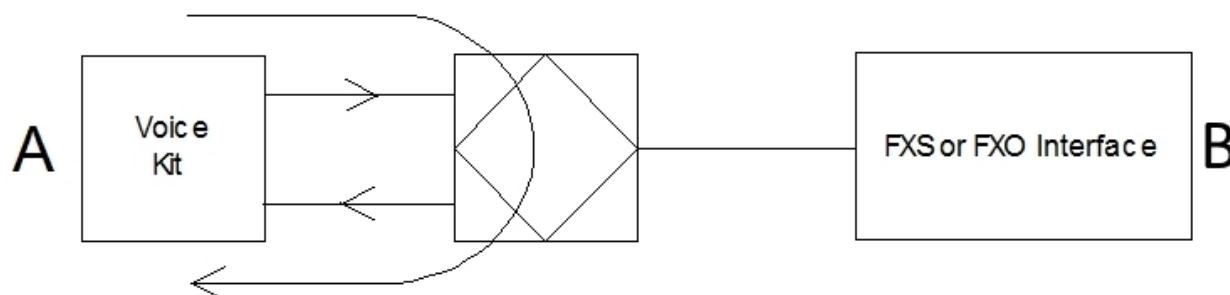


Fig. 10: Figure 1: Appearance of the echo when passing from two to four wires

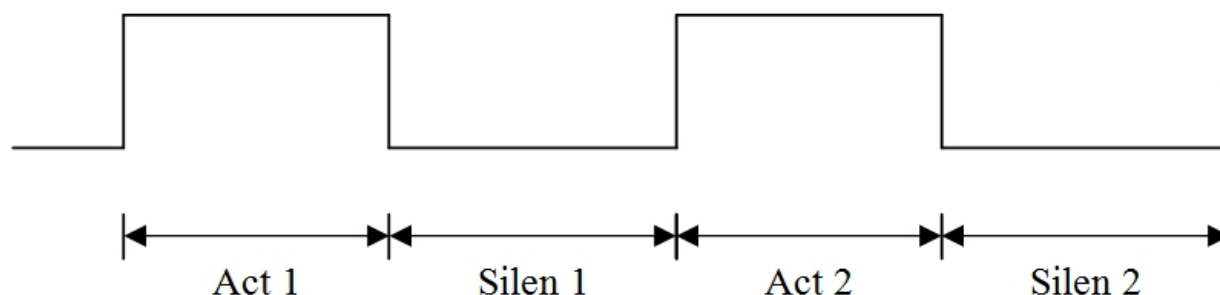
Our device incorporates an echo canceler which aims to reduce said echo. However this can only work effectively when the Signal to Echo ratio is greater or equal to 6 dB. In this case, this signal is the output signal from the kit towards the hybrid.

The only configuration parameter affecting the echo is the microphone gain, therefore less gain means better signal to echo ratio and consequently the canceler operates better. (The loudspeaker gain does not affect this as it is applied to both the signal as well as the echo).

If the echo is perceived on the A side of the conversation, the microphone gain parameter requires adjusting at the B end. This is because the echo produced on the A side as shown in Figure 1 is encoded and transmitted over IP to the other end and consequently arrives with an appreciable delay difference with respect to the original signal and manifests as a perceptible echo.

### 5.2 Detecting the busy tone in FXO

Hook-on detection in FXO lines is carried out through a detector contributed by the VoIP board DSP. This detector searches for a busy pattern, expressed in the following way:



where ActX is an interval when there is a signal and where SilenX is a silence interval. The duration of these intervals is configurable. The signal during the active intervals (Act) can be composed of one or two frequencies of different amplitudes.

#### 5.2.1 Configuration

The configuration of the busy pattern and several detection thresholds is done in the TELEPHONY menu through the "tone fxo-busy-pattern" command. Up to five busy patterns can be configured.

```
Config>TELEPHONY
-- Telephony configuration --
Telephony Config>tones fxo-busy-pattern ?
<1..5>          Value in the specified range
total-energy-threshold  Total energy threshold
low-energy-threshold   Low energy threshold
```

```

high-energy-threshold      High energy threshold
Telephony Config>tones fxo-busy-pattern 1 ?
  activity1              Time during which the tone is active
  activity2              Time during which the tone is active
  freq-1-level          Freq 1 level
  freq-2-level          Freq 2 level
  frequency1            Tone frequency
  frequency2            Tone frequency
  silence1              Time during which the tone is inactive
  silence2              Time during which the tone is inactive

```

Default values are shown through the “list parameters” command.

```

Telephony Config>list parameters
Dial tone frequency:      425          Ring tone activity: 15
Dial voice message:      Ring tone silence: 30

Alerting tone frequency: 425          Busy tone frequency: 425
Alerting tone activity:  15          Busy tone activity: 2
Alerting tone silence:   30          Busy tone silence: 2
Alerting voice message:  Busy voice message:

Error tone frequency:    425          Error tone silence 1: 2
Error tone activity:     2           Error tone silence 2: 6
Error voice message:

      --- FXO busy patterns ---

Pattern 1:

Frequency 1 (Hz):        425          Frequency2 (Hz):        425
Freq 1 level (dBm):     -5           Freq 2 level (dBm):    -32
Activity 1 (cs):         20           Activity 2 (cs):        20
Silence 1 (cs):         20           Silence 2 (cs):        20

Pattern 2:

Frequency 1 (Hz):        425          Frequency2 (Hz):        425
Freq 1 level (dBm):     -5           Freq 2 level (dBm):    -32
Activity 1 (cs):         20           Activity 2 (cs):        20
Silence 1 (cs):         20           Silence 2 (cs):        20

Pattern 3:

Frequency 1 (Hz):        425          Frequency2 (Hz):        425
Freq 1 level (dBm):     -5           Freq 2 level (dBm):    -32
Activity 1 (cs):         20           Activity 2 (cs):        20
Silence 1 (cs):         20           Silence 2 (cs):        20

Pattern 4:

Frequency 1 (Hz):        425          Frequency2 (Hz):        425
Freq 1 level (dBm):     -5           Freq 2 level (dBm):    -32
Activity 1 (cs):         20           Activity 2 (cs):        20
Silence 1 (cs):         20           Silence 2 (cs):        20

Pattern 5:

Frequency 1 (Hz):        425          Frequency2 (Hz):        425
Freq 1 level (dBm):     -5           Freq 2 level (dBm):    -32
Activity 1 (cs):         20           Activity 2 (cs):        20
Silence 1 (cs):         20           Silence 2 (cs):        20

Total Energy Threshold (dBm): -50
Low Energy Threshold (dBm):  -35
High Energy Threshold (dBm):   0

      --- FXO supervision pattern ---

```

```

Frequency 1 (Hz):      0           Frequency2 (Hz)           0
Freq 1 level (dBm):   -5           Freq 2 level (dBm)       -32

DTMF tones timeout:  10

```

If the pattern is that Act1 is equal to Act2 and Silen1 is equal to Silen2, we recommend setting Activity 2 and Silence 2 to 0, as shown in the above figure: The hook-on pattern is configured for the busy pattern used in Spain (425 Hz pure tone, cadence 200 ms/200ms). If the pattern is configured with small sequences of pulses separated by a long silence, the signal, for detection purposes, can be considered as a continuous sequence of pulses.

## 5.2.2 Practical Configuration

The busy pattern detector is highly dependent on the frequency, so the difference between the real value and the configured value must not exceed 2% of the real value. Other aspects that the detection is dependent on are: a) microphone and receiver gains and b) the duration of the signal and silence intervals.

### 5.2.2.1 What should I do if I don't know either the frequency or the duration of the intervals?

A simple way would be to consult your line supplier, however if this is not possible, they can be obtained by measuring them. To do this you need to record a voice message with the hook-on tone and process it.

#### 5.2.2.1.1 Recording a hook-on tone

Suppose that the VoIP board is in slot 2 and line 1 is going to be used in FXO mode, the following voice configuration is used:

```

;
telephony
; -- Telephony configuration --
dial-peer 1 voice-port
destination-pattern 100
target voice-port voip1/0 2
exit
;
exit
;
network voip2/0
; -- VoIP interface Configuration --
line 1 interface-type fxo
line 1 direct-dialing 100 0s
;
line 2 interface-type fxs
;
line 3 interface-type fxs
;
line 4 interface-type fxs
exit

```

To store the tone as a voice message, use the “trace rcd” command, located in the telephony monitoring menu, indicating the slot number, instance, voice message line and number. In our case the command is “trace rcd 2 0 1 1” to record voice message 1. The process is as follows:

- Enter “trace rcd 2 0 1 1” (without quotation marks) but do not press intro as yet.
- Call the telephone number associated with the line configured as FXO. When you do this the device will hook-off line 1 on detecting the rings and automatically call port 2 configured as FXS. If there is a telephone connected to this port, it will ring and you need to pick up the phone to establish the call.
- Hang up the caller telephone. From this point the FXO receives the busy tone. Press intro. From here the busy tone is recorded. The console displays the following text:

```

RECORD Telephony Mon+trace rcd 2 0 1 1
Rcd data...
RECORD Telephony Mon+

```

This process will terminate after some 12 seconds. You can check this through the following command:

```

RECORD Telephony Mon+display voice

These voice messages are loaded in memory:

```

```

Voice number   SIZE           CODEC           VERSION
-----
7              72400          G711alaw       2

```

```
RECORD Telephony Mont+
```

d) Verify the quality of the recording through the “voice” monitoring command.

You may not hear anything or very low pulses. This means the recording signal level is low. You can check the signal level to detect through the “list energy” voip monitoring command.

Execute this command while the busy tone is present and after a few seconds you can see the dynamic range of the signal as well as the signal energy. If the codec is G.711 or G.729, an energy value in dBm is given every 10 ms. For the values and the situation of these, check whether or not there is a tone.

```

RECORD+net voip2/0
-- VoIP network monitor --
RECORD voip2/0+list energy 1
Audio range (dBm): -60 / -20
                -22
                -21
                ...

```

From these values you can get an idea of the duration of the busy tone intervals.

There is no point in continuing to the next point until you have checked that the recorded signal has an acceptable dynamic margin.

To resolve this problem, increase the FXO line microphone gain.

```

*
RECORD *p 4
RECORD Config>net voip2/0

-- VoIP interface Configuration --
RECORD voip2/0 config>line 1 mi
RECORD voip2/0 config>line 1 mic-gain 15

```

Save, restart the device and repeat the process until you have an acceptable signal.

e) Get the voice message from FTP. This is named LOC1.loc. If voice message 2 has been recorded, the file is LOC2.loc.

### 5.2.2.1.2 Obtaining the durations and frequencies from the busy pattern

Use the KTH Wavesurfer program, available in various platforms, to obtain this information (<http://www.speech.kth.se/wavesurfer/download.html>). Currently we are using release 1.8.3. This program allows you to see the different signal parameters, reproduce the signal, etc.

Once installed, open the loc7.loc file (FILE->OPEN). Don't worry about the format, once you can see this in the file tree, click on it. A screen will appear giving information on how to process it in raw mode. By default a sample frequency is selected of 8000 HZ, A law and mono which are correct.

Press OK and subsequently select the pop-up window “Waveform” where the wave form for the stored signal will appear.

In the wave form you can measure the duration of the signal and the silence pulses. To do this, select the required pulse and drag with the left hand mouse button depressed. On doing this, the duration of the selected pulse is shown in ms. Measure the duration of the signal and silence pulses. To configure them in the device, divide by 10 as these are configured in cs.

To find out the signal frequency, position the mouse over the signal pulse and press the right hand button. Select “LTAS...” and the signal spectrum (frequency information) appears at this point. If you move the mouse over this curve in the lower part of the window, it gives the signal frequency and amplitude. Configure “FFT points” at 2048. The high amplitude point corresponds to the signal frequency (the higher the amplitude the lower the value is in dB in negative values or these can even take positive values). If there are various frequencies where the maximum value is given, take the central frequency. This is the value to configure. In cases where the signal is made up of two tones, two peaks will appear in these frequencies. Configure some adjusted frequencies, this is important so the detector can operate.

### 5.2.2.2 Initial configuration and test

Configure the frequency and duration parameters in the telephony menu. If the signal consists of a single 480 tone in a pulse train format of 300 ms of signal and 300 ms of silence, this needs to be configured as shown below (in this case, we have selected pattern number 1, but we can select which ever pattern number we want):

```
tones fxo-busy-pattern 1 activity1 30
tones fxo-busy-pattern 1 frequency1 480
tones fxo-busy-pattern 1 silence1 30
```



#### Warning

PLEASE REMEMBER, the durations must be CONFIGURED IN HUNDREDTHS OF A SECOND (divide the taken measurements by 10).

Save the configuration and restart the device. When the device prompt appears, display the events using the “view” command. Make a call to the device in the same way as when recording the busy tone, i.e., call the FXO line, let it hook-on and once you have done that, hang up the caller. If all goes well, the line should detect the busy tone and hang up (the slot 2 B LED switches off and the VoIP events show the line passes to IDLE).

Repeat this process various times to check that it hangs up correctly when the caller hangs up.

Repeat the call and on hearing the dialing tone, dial 31. If there is a telephone connected to line 2, it will ring. Pick it up and you can talk between both ends of the call. Hang up the caller and if everything goes smoothly the FXO line will detect the busy tone and hang up so the call goes to an error state with the corresponding tone. Repeat this various times to check the hook on is correct.

## 5.3 FXO line supervision

FXO line supervision lets you know if a line connected to an FXO port in a VoIP POTS board is able to route an outgoing call.

Supervision is carried out by detecting the dialing tone. This tone is characterized by being a continuous tone made up of one or two frequencies. By default supervision is deactivated and can be configured per line. If it's enabled, each time a call is made on an FXO line, the dialing tone is detected for a maximum configurable time: if the tone is detected, dialing is executed; if not an error appears and the call is processed with an alternative dial-peer which can route the call.

### 5.3.1 Configuring

To carry out supervision, you need to configure what the dialing tone is and enable supervision on the FXO line. Optionally you can configure the time interval to detect the dialing tone and the levels relating to the tone signal level.

FXO line supervision is enabled in the voice interface itself.

```
voip1/0 config>line 1 ?
  active           Line can receive and make calls
  alert           Substitute the alert tone with a voice message
  hold            Hold voice message
  busy            Substitute the busy tone with a voice message
  default         Default line configuration
  dial            Dial configuration options
  direct-dialing  Call is automatically carried out to this number
  echo-cancellation Echo cancellation is done in the line
  error           Substitute the error tone with a voice message
  framer          Framer configuration menu
  mic-gain        Indicate the line input gain
  interface-type  Indicate the line type fxs/fxo
  ip-calls        Permit making calls over IP
  forward-no-answer Telephone to forward the call if the callee doesnt
                  answer
  no              Negates a command or sets its defaults
  speaker-gain    Indicate the line output gain

  supervised      Supervise FXO line availability

  telephone-number Configures the telephone number associated to the line
```

```

timing                Configure timing parameters
tone-level           Volume gain with which you generate the DTMF digits
track                Track this line
voip1/0 config>line 1 supervised
voip1/0 config>show config

  line 1 interface-type fxo
  line 1 supervised
;
  line 2 interface-type fxo
;
voip1/0 config>
voip1/0 config>

```

From this same menu you can also configure the time interval to carry out detection. Once this has timed out, the line is assumed to be unavailable and another dial-peer can be searched for to route the call. By default, the detection interval is 2 seconds. In the following example it has been set to 3 seconds.

```

voip1/0 config>line 1 timing supervision-interval 3
voip1/0 config>show config

  line 1 interface-type fxo
  line 1 supervised
  line 1 timing supervision-interval 3
;
  line 2 interface-type fxo
;
  line 3 interface-type fxo
;
  line 4 interface-type fxo
;
voip1/0 config>

```

The tone to detect is configured from the TELEPHONY menu.

```

Config>TELEPHONY

-- Telephony configuration --
Telephony Config>tone fxo-supervision-pattern ?
  freq-1-level    Freq 1 level
  freq-2-level    Freq 1 level
  frequency1      Tone frequency
  frequency2      Tone frequency
Telephony Config>

```

The default values are displayed using the “list parameters” command.

```

Telephony Config>list parameters
Dial tone frequency:      425      Ring tone activity: 15
Dial voice message:      Ring tone silence: 30

Alerting tone frequency:  425      Busy tone frequency: 425
Alerting tone activity:   15      Busy tone activity:  2
Alerting tone silence:   30      Busy tone silence:   2
Alerting voice message:  Busy voice message:

Error tone frequency:     425      Error tone silence 1: 2
Error tone activity:      2      Error tone silence 2: 6
Error voice message:

      --- FXO busy pattern ---

Pattern 1:

Frequency 1 (Hz):        425      Frequency2 (Hz):      425
Freq 1 level (dBm):     -5      Freq 2 level (dBm):  -32
Activity 1 (cs):        20      Activity 2 (cs):      20
Silence 1 (cs):         20      Silence 2 (cs):       20

Pattern 2:

```



```

Frequency 1 (Hz):    425           Frequency2 (Hz):    425
Freq 1 level (dBm): -5           Freq 2 level (dBm): -32
Activity 1 (cs):    20           Activity 2 (cs):    20
Silence 1 (cs):    20           Silence 2 (cs):    20

Pattern 3:

Frequency 1 (Hz):    425           Frequency2 (Hz):    425
Freq 1 level (dBm): -5           Freq 2 level (dBm): -32
Activity 1 (cs):    20           Activity 2 (cs):    20
Silence 1 (cs):    20           Silence 2 (cs):    20

Pattern 4:

Frequency 1 (Hz):    425           Frequency2 (Hz):    425
Freq 1 level (dBm): -5           Freq 2 level (dBm): -32
Activity 1 (cs):    20           Activity 2 (cs):    20
Silence 1 (cs):    20           Silence 2 (cs):    20

Pattern 5:

Frequency 1 (Hz):    425           Frequency2 (Hz):    425
Freq 1 level (dBm): -5           Freq 2 level (dBm): -32
Activity 1 (cs):    20           Activity 2 (cs):    20
Silence 1 (cs):    20           Silence 2 (cs):    20

Total Energy Threshold (dBm): -50
Low Energy Threshold (dBm):  -35
High Energy Threshold (dBm):   0

        --- FXO supervision pattern ---
Frequency 1 (Hz):    0           Frequency2 (Hz)           0
Freq 1 level (dBm): -5           Freq 2 level (dBm)       -32

DTMF tones timeout: 10           Error Timeout: 30

```

You need to at least configure frequency 1 for the dialing tone. By default, no frequency is configured. In Spain we use a 425 Hz pure tone.

The dialing tone detector is highly dependent on the frequency. Consequently, the difference between the real value and the configured value must not surpass 2% of the real value. Another aspect that detection depends on is the microphone and receiver gains. To determine the dialing tone parameters, follow the guidelines given in the section on "Detecting the busy signal in FXO" in this appendix using the dial tone part.

#### Example:

In the following configuration, all three figure numbers beginning with 9 are sent by various FXO lines. The dialing tone is a 435 Hz pure tone. Should one of these be busy or unavailable, a search is made for one that is free and available.

```

log-command-errors
no configuration
set hostname FXO-SUPERVISED
telephony
; -- Telephony configuration --
  tones fxo-supervision-pattern frequency1 425
  dial-peer 10 voice-port
    destination-pattern 9..
    target voice-port voip1/0 1
  exit
;
  dial-peer 11 voice-port
    destination-pattern 9..
    target voice-port voip1/0 2
  exit
;
  dial-peer 12 voice-port
    destination-pattern 9..

```

```

    target voice-port voip1/0 3
  exit
;
dial-peer 13 voice-port
  destination-pattern 9..
  target voice-port voip1/0 4
  exit
;
dial-peer 100 sip
  destination-pattern 1..
  target ipv4 172.24.78.95
  exit
;
exit
;
;
network ethernet0/0
; -- Ethernet Interface User Configuration --
  ip address 172.24.78.94 255.255.0.0
;
;
;
  exit
;
network voip1/0
; -- VoIP interface Configuration --
  line 1 interface-type fxo
  line 1 supervised
;
  line 2 interface-type fxo
  line 2 supervised
;
  line 3 interface-type fxo
  line 3 supervised
;
  line 4 interface-type fxo
  line 4 supervised
;
exit
;
;
protocol sip
  application address 172.24.78.94
  application gateway
exit
;
dump-command-errors
end

```

If for example, line 1 has nothing connected or doesn't work, the following events appear:

```

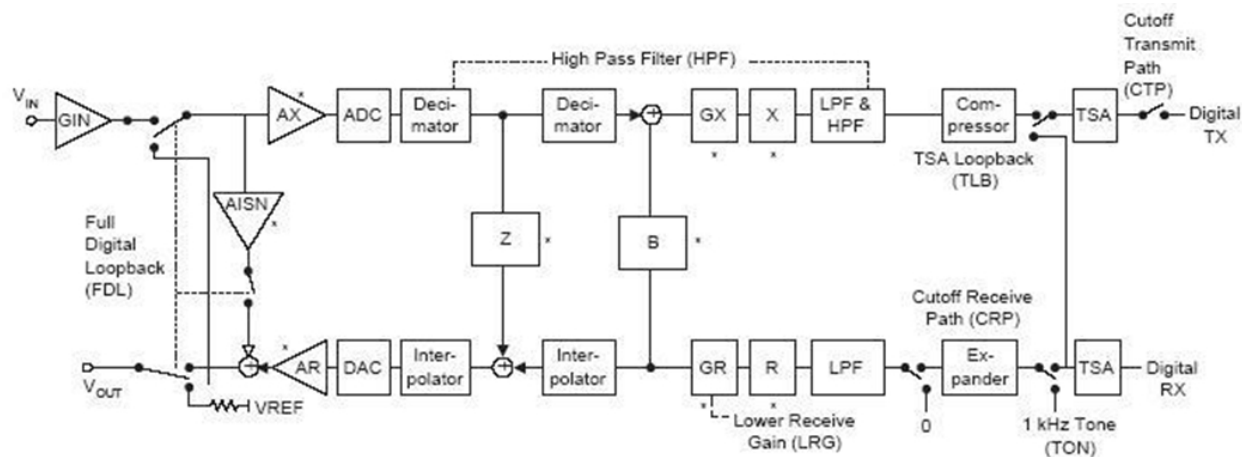
06/15/06 16:43:40 VoIP.003 Ev INCOMMING CALL ln voip1/0-1
06/15/06 16:43:40 VoIP.004 State WTG HOOK OFF ln voip1/0-1
06/15/06 16:43:42 VoIP.006 TmOut CONTROL TIMER ln voip1/0-1
06/15/06 16:43:42 VoIP.013 Error: FXO line not available ln voip1/0-1
06/15/06 16:43:42 VoIP.004 State IDLE ln voip1/0-1
06/15/06 16:43:42 VoIP.003 Ev INCOMMING CALL ln voip1/0-2
06/15/06 16:43:42 VoIP.004 State WTG HOOK OFF ln voip1/0-2
06/15/06 16:43:44 VoIP.003 Ev INFO ln voip1/0-2
06/15/06 16:43:44 VoIP.004 State ALERTING ln voip1/0-2
06/15/06 16:43:44 VoIP.003 Ev HOOK OFF ln voip1/0-2
06/15/06 16:43:44 VoIP.004 State CALL ESTABLISHED ln voip1/0-2
06/15/06 16:43:44 VoIP.003 Ev CHANGE-CODEC ln voip1/0-2
06/15/06 16:43:44 VoIP.005 TmOut TONE TIMER ln voip1/0-2

```

So the call exits through line 2. Supervision is carried out on both lines. The INFO event indicates that line 2 has detected the dialing tone.

## 5.4 Analog Board Framer

The analog boards (e.g., POTS and E&M) use a *framer* which carries out the analog-digital and digital-analog conversion. The *framer* internal diagram used is (per channel):



The DSP receives audio through the *Digital TX* and in turns sends the audio through the *Digital RX*. The framer receives the analog through the  $V_{in}$  terminal and transmits it through the  $V_{out}$  terminal.

In the figure you can identify the GIN, AX, AR and LRG amplifiers and attenuators which are configurable by line. On modifying the INPUT-GAIN parameter, you're modifying the GX block. When modifying the OUTPUT-LOSS parameter, the GR block is being modified.

In this diagram you can also identify the loops available at the framer layer. On enabling the LOOP FRAMER EXTERNAL loop, you modify the Z block so the received signal is not altered and the lower CRP switch and the upper CTP switch are opened. Consequently the signal received by the  $V_{in}$  is transmitted to the  $V_{out}$  output terminal. On enabling the LOOP FRAMER INTERNAL loop, the full digital loop is enabled. On enabling the LOOP FRAMER DSP-INTERNAL loop, the TLB loop is enabled so everything received by the *Digital RX* is sent by the *Digital TX* without altering anything.

Finally, you can also identify in the diagram the 1 KHz (TON) tone generator, available through the TEST TONE FRAMER-1KHZ-EXTERNAL monitoring command.