



## Telephony over IP

bintec Dm722-I

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## I Related Documents

bintec Dm715-I BRS

bintec Dm730-I DHCP Protocol

bintec Dm749-I NSM

bintec Dm754-I NSLA

bintec Dm766-I SIP

bintec Dm767-I H323 Protocol

bintec Dm770-I VoIP Interfaces

bintec Dm776-I Power Over Ethernet

bintec Dm777-I NOE

bintec Dm782-I SCCP

bintec Dm791-I MGCP Protocol

# Chapter 1 Introduction

## 1.1 Introduction to IP Telephony

The concept behind IP telephony (also known as Voice over IP) is simple: transmission of voice messages using IP networks. This technology allows you to transmit voice messages, video, fax, modem, etc., over UDP/IP packets, which can then be transmitted over the Internet using a private IP network or simply between two devices in the same LAN. The media sent by IP is encoded using codecs, which allow various levels of quality and compression (bandwidth). Audio and video are sent using real time RTP protocol (over UDP) while fax is usually sent over UDP using T.38 protocol.

Signaling protocols are used to establish and terminate calls, locate the called end, and negotiate the codecs used in the call. These protocols can be open standards drawn up by international groups (IETF (Internet Engineering Task Force) and ITU (International Telecommunications Union)), or proprietary protocols of a manufacturer's own design, consisting of complete and closed IP telephony solutions.

Currently, the most widely used IP telephony protocols are H.323, drafted by ITU, and SIP, drafted by IETF. Both are signaling protocols used to establish and release calls. Once a call is established, both use RTP to send voice or video through RTP/UDP/IP packets.

## 1.2 Functionalities available in the device

IP enabled devices require a wide range of technology:

- *Quality of Service (QoS)* mechanisms (*bintec Dm715-I BRS*) enabling traffic classification, traffic marking, bandwidth allocation, prioritization and limiting. Said function is essential to ensure the quality of real time streams in a Voice and Data integrated network.
- *DHCP Server and Relay (bintec-Dm730-I DHCP Protocol)*: needed to deploy IP telephony terminals through autoinstallation and automatic supply.
- Layer 2 detection and configuration protocols such as *LLDP (bintec Dm813-I LLDP Protocol)* or *CDP (bintec Dm785-I CDP Protocol)* to automatically deploy IP telephony terminals.
- *Power Over Ethernet (PoE) Interfaces (bintec Dm776-I Power Over Ethernet)* in some devices, which let power be supplied through IP telephone terminals network cable (among other things), simplifying wiring, installation and increasing the reliability of IP telephony deployments.
- Numerous advanced routing features: *header compression* to optimize bandwidth (*CRTP, IPHC*, etc.), *routing protocols (RIP, OSPF, BGP*, etc.), a large quantity of data interfaces, *policy routing, IPSec encryption, backup* mechanisms for the IP network, *TVRP, VLAN, VRF*, etc.
- Remote management through *SNMP, Syslog, CLI* and *Telnet events*, etc.
- *STUN (bintec Dm769 STUN Protocol)* to support architectures with NAT.
- *Active monitoring* mechanisms (*bintec Dm749-I NSM* and *bintec Dm754-I NSLA*) for connectivity and network quality parameters (*packet loss, delay, jitter*), which allow routing and IP telephony reconfiguration, providing *backup* and *supervision* for both data and voice.
- IP telephony *signaling protocols*: *SIP (bintec Dm766-I SIP)*, signaling protocol developed by IETF; *H.323 (bintec Dm767-I H323\_Protocol)*, set of protocols (H.245, H.225, Q.931, RAS) defined by ITU; *UA/NOE (bintec Dm777-I NOE)* Alcatel's proprietary protocol for IP telephony, *MGCP (bintec Dm791-I MGCP Protocol)*, protocol developed by IETF to control Media Gateways; and *SCCP (bintec Dm782-I SCCP)*, a Cisco proprietary protocol for IP telephony.
- Conventional telephony interfaces (*bintec Dm770-I VoIP Interfaces*): *FXS, FXO, ISDN-BRI, ISDN-PRI, R2-E1, analog and digital E&M*.
- Dial plan, numerical transformations, codec configuration, RTP/RTCP, fax, DTMF digits, switchboard functions, call admission control (CAC), call detail record (CDRs), etc. These generic features, independent of the signaling protocols and conventional telephony interfaces used, are described in this manual.

The latter three points encompass the core components of IP telephony: IP signaling protocols, conventional telephony interfaces and dial plan logic, and media configuration and call management.

The available features are different for each IP signaling protocol. For further information on this, please see the corresponding manual. On a special note, the MGCP protocol can be disassociated from the rest, as its purpose is for a remote server to control conventional telephony interfaces available in the device, executing calls through them; currently, only ISDN interfaces can be handled by MGCP, both BRIs (from the PCI card and motherboard) as well as PRIs. For this specific protocol, intelligence and the available services reside in the remote server.

As for other protocols, in general terms and without going into detail, we can distinguish three types of functions

when it comes to call establishment:

- Media Gateway functionality: possibility of establishing calls between conventional interfaces and devices with IP signaling. This feature is available in all conventional telephony interfaces and all IP signaling protocols.
- Conventional Analog Switchboard functionality: establishment of calls between conventional telephony interfaces. This feature is available for all existing interfaces.
- IP Switchboard or IP Telephony Server functionality: establishment of calls between IP devices. The following calls are supported: SIP-SIP, NOE-NOE, SIP-NOE, SCCP-SCCP and SIP-SCCP.

The type of call established depends on the dial plan, which associates dial patterns with protocols or interfaces and whose basic unit is the dial-peer. (This is described in the following chapter). In addition, numerous call parameters can be configured in the dial-plan itself (through the dial-peer) depending on the call source and/or destination. The dial-plan can be dynamically reconfigured to adapt to different IP connectivity loss situations, bad quality on the network, in order to limit calls, call balance, etc. This is configured through dial-peers **track**, which makes the dial-peer active or inactive depending on the associated parameter.

Certain switchboard features can also be configured through the dial-plan, including call capture groups, directed pickup, hunt groups, simultaneous ringing, call transfer, etc. These features are explained in Chapter 2 Section 2 Switchboard Features.

Hence, the configuration of the telephone menu, described in this manual, is an essential part of the configuration of the equipment for telephony functions.

## Chapter 2 Generic Telephony Configuration

### 2.1 Dial Peers

#### 2.1.1 Dial Peers Definition

A dial-peer is an abstract concept to represent the calling or called entity in a telephone call. The calling entity is represented through a dial-peer, known as incoming, and the caller entity through a dial-peer known as outgoing.

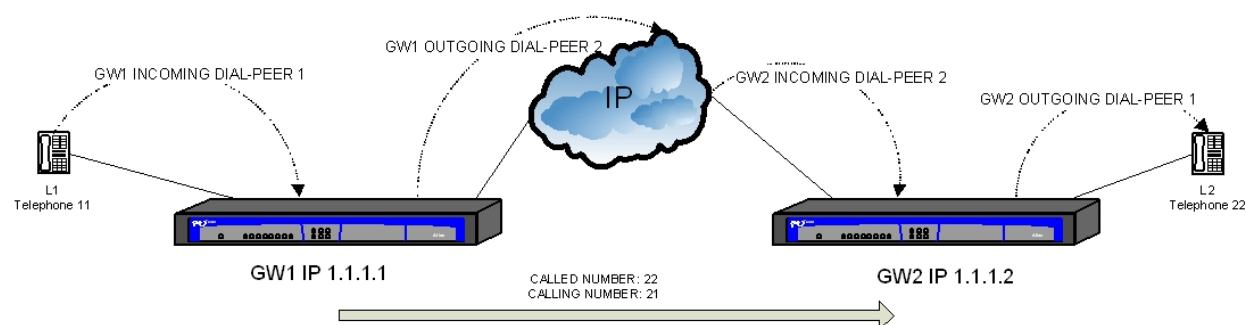
For example, let's suppose a telephone connected to GW1 router line 1 with telephone number 11 calls telephone number 22, which corresponds to GW2 router line 2, using SIP. The following shows the telephony menu configuration for both devices and a diagram.

GW1:

```
dial-peer 1 voice-port
  destination-pattern 11
  target voice-port voip1/0 1
exit
;
dial-peer 2 sip
  destination-pattern 2.
  target ipv4 1.1.1.2
exit
;
```

GW2:

```
dial-peer 1 voice-port
  destination-pattern 22
  target voice-port voip1/0 2
exit
;
dial-peer 2 sip
  destination-pattern 1.
  target ipv4 1.1.1.1
exit
;
```



As shown in the diagram, for GW1, the call enters through dial-peer 1 as this dial-peer has router line 1 as its target. Once connected, the incoming dial-peer searches for the outgoing dial-peer and finds 2, as this has pattern 2., which matches called number 22, and forwards the call to IP 1.1.1.2 through SIP.

The GW2 device receives a call matching dial-peer 2, as calling number is 11, and dial-peer 2 destination-pattern is 11. Once connected, the incoming dial-peer searches for the outgoing dial-peer and finds dial-peer 1, which has 22 as destination-pattern, matching called number 22. The call therefore is as follows:

GW1 incoming dial-peer 1 corresponding to a telephone connected to line 1.

Outgoing dial-peer 2 corresponding to a SIP device with IP 1.1.1.2.

GW2 incoming dial-peer 2 corresponding to a SIP device with IP 1.1.1.1.

Outgoing dial-peer 1 corresponding to line 2 in the device.



For further information on how the device associates a call with a dial-peer, please see section 2.2 Dial-Peers Search.

There are several types of dial-peers:

- **voice-port**: can match both in incoming and outgoing with calls over device interfaces. See *bintec Dm 70-I VoIP Interfaces* manual.
- **sip**: can match both in incoming and outgoing with IP calls signaled through SIP. See *bintec Dm770-I SIP* manual.
- **H.323**: can match both in incoming and outgoing with IP calls signaled through H.323. See *bintec Dm767-I H323* manual.
- **group**: only matches as outgoing dial-peer and allows the call to be simultaneously forwarded through various dial-peers. See section 2 – Switchboard Features – in this chapter.
- **direct**: only matches as outgoing dial-peer and establishes RTP/RTCP audio to a fixed IP address/port configured in said dial-peer, without executing any signaling. Calls of this type can only be established when the source is one of the device ports.
- **noe**: this can match both in incoming and outgoing for calls with Alcatel IP Touch terminals, which use their own signaling, known as UA/NOE. Please see manual *bintec Dm777-I NOE*.
- **sccp**: this can match both in incoming and outgoing for IP calls, which signal using Cisco's SCCP proprietary protocol. Please see manual *bintec Dm782-I SCCP*.
- **facility**: this only matches as an outgoing dial-peer and allows you to configure certain switchboard features such as call capture and transfers. Real calls are not established through said dial-peer. See Section 2 – Switchboard Features – in this chapter.

Not all calls have an incoming dial-peer; if the device cannot find an incoming dial-peer for a call, it assigns the default dial-peer (zero). All calls have an outgoing dial-peer as the device would not know where to send the call without one. If the device cannot find an outgoing dial-peer, said call is released. Through **call application incoming-match force**, this behavior is modified so the call is rejected if it cannot find an incoming dial-peer. This gives more flexibility and control, especially when modifying dial-plan using dial-peers **track** command.

## 2.1.2 Dial Peers Search

When configuring a device, the manner in which the incoming and outgoing dial-peer for a new call is decided upon is crucial.

Basically, there are two influencing elements when searching for an incoming and outgoing dial-peer. These are called number and calling number. If the call originates from the device's physical interface (VoIP interface), the interface and the originating line for the call also influence the search to locate the incoming dial-peer.

**destination-pattern**, **incoming called number** and **incoming calling number** commands admit a dial-plan pattern where the following syntax is used:

- 0-9,+,\*,# Represent literal characters.
- . Represents any one of the above digits.
- , Is used to include dialing pauses in outgoing calls by analog interfaces (FXS/FXO/E&M). Each ',' introduces a pause of 1.1 seconds that is only useful in **prefix** commands and translations.
- T Represents a variable length string.
- P Waits for dialing to finish in overlap dialing.

Besides, the command **destination-pattern** also admits a range of digits with the following syntax:

- [digit1-digit2] Represents any digit between *digit1* and *digit2*, both included.

When more than one dial-peer matches using the same command, the chosen dial-peer is the one with a specific pattern. If the specification level is the same, the order of configuration is followed.

These literal characters are more specific than the '.' wildcard, which, in turn, is more specific than 'T'. Both 'T' and 'P' only make sense at the end of a pattern and, in both cases, prompt interfaces with overlap dialing (FXS, ISDN, etc.) to behave in a particular manner: although the pattern matches, the call is not immediately executed and only happens when explicit dialing ends or the timer involving digits configured in these interfaces times out (see command **line<id> timing interdigit-timeout** in manual *bintec Dm770-I VoIP Interfaces*). The difference between these two options is that 'T' considers any number the previous pattern had a prefix match, while 'P' forces the number to specifically match the pattern. 'P' simply waits for the dialing to end in order to call over different dial-peers whose patterns begin with 8P or 88 (in interfaces with overlap dialing). Without 'P', the call is always made by dialing the first 8. 'P' does not affect the pattern's specification level. Therefore, number 2222 is as specific as 222.P and more specific than 22..(which, in turn, is more specific than 22T and this more specific than T).

The following paragraphs detail the search for the incoming and outgoing dial-peers, depending on these three parameters.

### 2.1.2.1 Incoming

When a call comes in, a search is made through the list of dial-peers to find an incoming dial-peer associated with this call. Only dial-peers of the same type as the incoming call are checked, i.e. a call coming in via SIP can only have a SIP dial-peer as an incoming dial-peer. A call coming in via a device's VoIP interface can only have a VoIP dial-peer as an incoming dial-peer, etc.

Once the dial-peer type has been verified, a list containing all dial-peers of this type is searched and the following criteria are applied:

- If **incoming access-list <id>** has been configured, the device checks if the call signaling source is permitted in the corresponding access list. If it isn't, call source cannot match said dial-peer. If **incoming acc-list <id>** has not been configured, then any call source is allowed.
- The called number is compared with the incoming called number configured in the dial-peer. If one matches, the search ends.
- If, after comparing all dial-peers, none matches the above criteria, the calling number is compared with the incoming calling number configured in the dial-peer. If one matches, the search ends.
- If, after comparing all the dial-peers, none of them matches the above criteria, the calling number is compared with the destination pattern and the destination-alias in each dial-peer. If one matches, the search ends.
- If, after comparing all the dial-peers, none of them matches the above criteria and the call comes from a line pertaining to a device's VoIP interface, target voice-port compared with the interface and the line the call comes in through. For virtual ISDN VoIP interfaces, the number for all the lines is 1.
- If any match, the search ends.
- If none match, incoming dial-peer 0 is assigned.

### 2.1.2.2 Outgoing

- The called number is compared with the destination-pattern and the destination-alias in each dial-peer. If it matches the search ends.
- If, after comparing all the configured dial-peers, none of them matches, a search is carried out over the dynamic dial-peers, which are created when the SIP, H.323, NOE or SCCP terminals register in the device.
- If no dial-peer has been found for the call and if the incoming call is SIP and there is an active proxy configured, the device behaves as a SIP server. The call is then routed to the SIP proxy with the device behaving as Back-To-Back.
- If none match, the call ends.

Once an outgoing dial-peer has been found, the call is routed to it. If that dial-peer releases the call because it is busy (or because some error has occurred preventing the call from being forwarded), the search continues with the next dial-peer until either there are no further dial-peers to match or, one of the matching dial-peers does not release the call due to error or because it is busy. If **huntstop <cause>** in the outgoing dial-peer is configured, the search details detained if there are errors.

Through the **incoming dial-plan peer-group <id>** command for dial-peer matching as incoming, you can configure a group of dial-peers to search for the outgoing dial-peer. To do this indicate a peer-group, which lists the dial-peers to be searched and allows you to include dynamic dial-peers created through external telephone registers ( **dynamic-peers** command), so you can configure a different dial-plan depending on where a call comes in. **call application dial-plan default peer-group <id>** command allows you to configure a peer-group to be used as a dial-plan in calls where the incoming dial-peer does not have the dial-plan configured. The **call application dial-plan sip-dynamic peer-group <id>** lets you configure a common dial-plan when the incoming dial-plan is dynamic. If you don't configure a dial-plan command, the device searches for an outgoing dial-peer among those configured in the telephony menu.

Please note modifications over the calling and called number, which may have been configured in the incoming dial-peer, are applied BEFORE proceeding with the search for an outgoing dial-peer.

To check dial-peers associated with an established call, go to the telephony monitoring menu and execute **list calls active**.

## 2.1.3 Importance of incoming Dial-Peers

The importance of outgoing dial-peers is clear. If a call cannot find an outgoing dial-peer, it can't be forwarded and consequently is immediately terminated. You may think incoming call matching an incoming dial-peer as unimportant, given the device will assign it to a dial-peer 0 (as said dial-peer has all the default values and can suitably route a call). This behavior, however, can be modified through call application **incoming-match force** (command) so if an incoming dial-peer can't be found, the call is rejected.

Dial-peer is very important when configuring parameters for incoming calls. Let's assume, for example, a device ex-

ecutes calls from numbers with pattern 2. to telephones with pattern 1. over H323, through a dial-peer with a specific codec configured, which we want to be used for calls between these two devices.

```
dial-peer 1 h323
  codec g7236k4
  destination-pattern 1.
  target ipv4 1.1.1.2
exit
```

If we only configure the above, dial-peer 1 will match all outgoing calls but not incoming. This means the codec used in incoming calls may not be the one needed. However if we configure an incoming called number with pattern 2. (assuming the device calling will use numbers with this pattern) dial-peer 1 also matches incoming calls: codec g7236k4 can be used for both incoming and outgoing calls.

```
dial-peer 1 h323
  codec g7236k4
  destination-pattern 1.
  incoming called number 2.
  target ipv4 1.1.1.2
exit
```

Furthermore, incoming dial-peers are important because they allow you to configure a subgroup of all existing dial-peers to use when searching for the outgoing dial-peer. I.e., they can have different dial-plans, depending on where the call comes in.

## 2.1.4 Examples

The following shows four examples of different calls over the same configuration. The configuration is as follows:

```
; Configuración del menú feature access-lists
access-list 10
  entry 1 default
  entry 1 permit
  entry 1 source address 172.24.27.133 255.255.255.255
;
exit

; Configuración del menú TLPHY
dial-peer 1 voice-port
  destination-pattern 11
  target voice-port voip1/0 1
exit
;
dial-peer 2 voice-port
  destination-pattern 12
  target voice-port voip1/0 2
exit
;
dial-peer 3 sip
  destination-pattern 2.
  incoming called number 3.
  incoming prefix 12
  incoming strip-digits 2
  target ipv4 172.24.100.130
exit
;
dial-peer 4 sip
  incoming access-list 10
  incoming called number T
  incoming dial-plan peer-group 1
exit

peer-group 1
  dial-peer 2
exit
```

### 2.1.4.1 Example 1

Call coming in on line 1 with called number 23 and without calling number. A search is made for incoming dial-peer; none matches until the criteria compares the source interface for the call with dial-peer 1 target voice-port. Consequently the selected incoming dial-peer is 1. As destination pattern is configured and the call does not have a calling number, 11 is selected as said calling number.

To search for the outgoing dial-peer, the whole dial-peers list is checked and 3 with destination-pattern 2. are found to match as outgoing dial-peer. As no modification has been configured either over the calling or called number when acting as the outgoing dial-peer, the call is established with incoming dial-peer 1, outgoing 3, called number 23 and calling 11.

### 2.1.4.2 Example 2

Call coming in on line 3 with called number 12 and without calling number. The incoming dial-peer is searched for but none matches, therefore dial-peer 0 is selected as incoming.

To search for the outgoing dial-peer, the whole dial-peer list is checked for one to match called number 12. Dial-peer 2 is found and this is chosen as outgoing dial-peer. The call is sent to line 2 in the VoIP1/0 interface with called number 12 and without the calling number.

### 2.1.4.3 Example 3

SIP incoming call with called number 35 and calling number 23. A search is carried out for the incoming peer, in this case incoming called number 3. coincides with called number 35, so incoming peer is 3. Although the call has come from IP 172.24.27.133, it won't match dial-peer 4 as 3. is more specific than T. The modifications are applied over the called number configured in the dial-peer so two digits are eliminated (incoming strip-digits 2) and prefix 12 (incoming prefix 12) is applied. Called number is 12 after having applied the configured modifications.

To find the outgoing dial-peer, the dial-peers list is checked to find a dial-peer that matches called number 12. Dial-peer 2 does and is selected as the outgoing dial-peer. The call is sent to line 2 over the VoIP1/0 interface with called number 12 and calling number 23.

### 2.1.4.4 Example 4

SIP incoming call from 172.24.27.133 with called number 12 and calling number 55. The device looks for the incoming peer, with this called and calling number it cannot match 3 but can match 4, as the access list permits this and is configured to match any called number (T). As dial-peer 4 has peer-group 1 configured as dial-plan, it only searches for an outgoing dial-peer (among the configured in this group), which is dial-peer 2. In this case the called number matches the destination-pattern, so the call is established on the voip1/0 interface line 2 with called number 12 and calling number 55.

Bear in mind that any other called number would mean the call would not establish. The dial-plan indicates dial-peer 2 can only be used as outgoing.

## 2.2 Switchboard Features

The device is capable of providing switchboard features for extensions in conventional telephony interfaces as well as IP telephones.

The supported features are as follows:

- Call transfer.
- Music on hold.
- Simultaneous ringing of various phones.
- Group pickup.
- Directed call pickup.
- Call forwarding.
- Hunt groups.
- IVR.

The following sections explain each feature in detail, the protocols and the interfaces these support, together with configuration examples

## 2.2.1 Call Transfer

A call can be transferred from both an IP terminal (SIP, SCCP, NOE), and an analog telephone connected to an FXS port. In both cases, blind transfer, semi-attended and attended are supported. A SIP terminal requests a standard transfer through a REFER message. The operation a user has to execute to transfer a call depends on the terminal being used (please see the corresponding manual). You can transfer calls from an FXS Terminal. User operation in this case is as follows:

With the call you want to transfer established, press the R key located on the telephone. You should hear a dialing tone. Dial the number you wish to transfer the call to and if you want to execute a blind transfer, simply hang up. A semi-attended transfer is one where you wait until the call begins to ring before hanging up so executing the transfer. To execute an attended transfer, wait until the new query call establishes (called picks up). In this situation, where the original call is on hold and the query established, complete transfer by hanging up the phone or, abort the query call and return to the original by pressing '2'. To cancel transfer and return to the original call (any time before the query call has established), press 'R' again.

Call transfer does not require any configuration.

## 2.2.2 Music on hold and voice messages

The device can provide music on hold or voice messages in various situations. Two types of recorded messages are supported: voice messages recorded from an FXS interface in proprietor format and wav. files. You can use up to nine different messages identified by numbers 1 to 9.

Proprietor files are recorded configuring a facility dial-peer with **target loc-record**. On dialing the configured number in said dial-peer destination-pattern from an FXS interface, two confirmation tones are heard and the audio begins to be recorded. To finalize, press a number on the telephone, which will be the number for the recorded message. The voice message is stored in the device flash with the name LOC <num>.loc and in the codec configured in the dial-peer facility. Maximum recording time is 30 seconds. Subsequently, you hear two confirmation tones and the recorded voice message is reproduced. Please remember a voice message is only reproduced in calls using the same codec used on making the recording, i.e. a call in G.729 cannot reproduce a voice message recorded in G.723.1 and vice versa. G.723.1 at 5.3 Kbps and G.723.1 at 6.4 Kbps formats are fully compatible with each other, so a call in G.723.1 at 5.3 Kbps can reproduce a voice message recorded with G.723.1 at 6.4 Kbps and vice versa.

The format supported for wav. files is G711 a law or u law, with sample frequency 8 KHz and a single channel (mono). Through voice-message <loc> filename <file> (see chapter 4), you can associate voice message identifier (1-9) with a file name in wav. format. The size of these files is only limited by the space in the flash.

You can view device audio recordings data through **display voice-message** monitoring (see chapter 3). Voice messages can be used as dialing tone, error tone, etc., in FXS/FXO lines; such as music on hold in any conventional telephony interface, or voice messages to be played in virtual interfaces (please see manual *bintec Dm770-I VoIP Interfaces*).

Music on hold is generated in interfaces when the call is retained at the other end of the conversation. This is configured through voice port using **line <id> hold voice-message <loc>**. (Please see manual *bintec Dm770-I VoIP Interfaces*). Calls can be put on hold from a SIP, NOE or FXS terminal.

The following point describes the possibility of sending a permanent stream of IP audio from a flash file. Please see section 2.8 Basic IVR for more examples related to audio recordings.

### 2.2.2.1 Continuous music on hold stream over IP

You can configure the device to send continuous RTP streams over IP reproducing a looped audio recording in the device flash. To do this, create a voip-dummy interface ( **add device voip-dummy <id>**) to reproduce a voice message through **line 1 behavior play-message** and **line 1 play <codec> voice-message <num>** command. Through **line <id> direct-dialing <tlph> <tim>** you can make a call to the configured number when the device has started up.

You create a direct dial-peer in the telephony menu, which allows you to establish the signalless call to the IP address and ports configured through **target**. The source IP used when sending is the internal ip, although you can configure this through the dial-peer using **local-ip**.

Destination may also be a multicast IP. So the router can send packets, proxy-igmp must be enabled and the forwarding device must be configured as an upstream interface.

The following configuration example shows how to send an audio stream in G711 alaw codec from an 8kalaw.wav file to the multicast ip 239.1.1.1 and port 2000 and another stream in g729 from a LOC2.LOC file to the same IP and to port 2010 via the ethernet0/0 interface

```
add device voip-dummy 1000
add device voip-dummy 1001
```

```

telephony
; -- Telephony configuration --
voice-message 1 filename 8kalaw.wav
;
dial-peer 10 direct
codec g711ulaw
destination-pattern 1
target ipv4 239.1.1.1 port 2000
exit
;
dial-peer 11 direct
codec g729
destination-pattern 2
target ipv4 239.1.1.1 port 2010
exit
;
exit
;
network voip1000
; -- VoIP interface Configuration --
line 1 play G711 voice-message 1
line 1 behavior play-message
line 1 direct-dialing 1 0s
;
exit
;
network voip1001
; -- VoIP interface Configuration --
line 1 play G729 voice-message 2
line 1 behavior play-message
line 1 direct-dialing 2 0s
;
exit
;
protocol ip
; -- Internet protocol user configuration --
internal-ip-address 172.24.78.156
;
address ethernet0/0 172.24.78.156 255.255.0.0
;
;
;
proxy-igmp
; -- IGMP proxy user configuration --
enable
upstream ethernet0/0 default
exit
;
exit

```

### 2.2.3 Ringing various telephones simultaneously

You can configure multiple ring extensions so various telephones ring at the same time. This is configured through a group dial-peer. The target for this type of dial-peer is a peer-group, which allows you to define a set of dial-peers and telephones to call through said dial-peers (see **peer-group** command).

#### Example 1:

When dialing 2000, you want SIP telephones 1001 and 1002 and the FXS 800 (on the viop1/0 line 1), to ring.

```

dial-peer 1 voice-port
destination-pattern 800
target voice-port voip1/0 1
exit
;
dial-peer 2 sip
destination-pattern 100.
target dynamic

```

```

exit
;
dial-peer 3 group
  destination-pattern 2000
  target group 1
exit
;
peer-group 1
  dial-peer 1 800
  dial-peer 2 1001
  dial-peer 2 1002
exit

```

On dialing 2000, this matches dial-peer 3 for the outgoing call and calls using dial-peer 1, telephone number 800, dial-peer 2, number 1001 and dial-peer 2 once more, number 1002.

#### Example 2:

When dialing 2000, you want all the dynamically registered SIP telephones and the FXS 800, (on the voip1/0 line 1), to ring.

```

dial-peer 1 voice-port
  destination-pattern 800
  target voice-port voip1/0 1
exit
;
dial-peer 2 group
  destination-pattern 2000
  target group 1
exit
;
peer-group 1
  dial-peer 1 800
  dynamic-peers
exit

```

On dialing 2000, this matches dial-peer 2 in the outgoing call and calls using dial-peer 1, telephone number 800 and all the dynamic dial-peers.

## 2.2.4 Group Pickup

You can configure capture groups so a dial-pattern is associated with a group of dial-peers where they can capture outgoing calls. Configure a facility dial-peer, where **target group-pickup peer-group <id>** is configured. On making a call to said number, if there is a call coming through the outgoing dial-peer pertaining to the peer-group, the call is captured. Please see the description on **target group-pickup** for further information.

#### Example 1:

We have the following telephones: three SIP telephones with numbers 1001, 1002 and 1003, two FXS telephones in the voip1/0 interface lines 1 and 2 with numbers 800 and 801. On dialing \*8, you can capture a call ringing on the FXS phone 800 or the SIP phones 1001 and 1002.

```

dial-peer 1 voice-port
  destination-pattern 800
  target voice-port voip1/0 1
exit
;
dial-peer 2 voice-port
  destination-pattern 801
  target voice-port voip1/0 2
exit
;
dial-peer 3 sip
  destination-pattern 100.
  target dynamic
exit
;
dial-peer 4 facility
  destination-pattern *8

```

```

    target group-pickup peer-group 1
  exit
;
peer-group 1
  dial-peer 1
  dial-peer 3 1001
  dial-peer 3 1002
exit

```

On dialing \*8, this matches dial-peer 4 in outgoing and checks to see if there are any calls ringing with this outgoing dial-peer 1 or outgoing dial-peer 3 and called numbers 1001 or 1002. If this occurs the call is captured, if not error is returned.

#### Example 2:

You want to add the possibility of capturing calls ringing on SIP phones with code \*9 to the above example. Add the following to the previous configuration:

```

dial-peer 5 facility
  destination-pattern *9
  target group-pickup peer-group 2
exit
;
peer-group 2
  dynamic-peers
exit

```

On dialing \*9, this matches outgoing dial-peer 5 and a check is carried out to see if there is a call ringing with the dynamic outgoing dial-peer.

#### Example 3:

In the above example you want to add that the FXS 800 and the SIP 1003 phones can only capture using code \*8 and not \*9. To do this, configure a specific dial-plan for calls coming into these extensions using **incoming dial-plan**.

```

dial-peer 1 voice-port
  destination-pattern 800
  incoming dial-plan peer-group 3
  target voice-port voip1/0 1
exit
;
dial-peer 2 voice-port
  destination-pattern 801
  target voice-port voip1/0 2
exit
;
dial-peer 3 sip
  destination-pattern 100.
  target dynamic
exit
;
dial-peer 4 facility
  destination-pattern *8
  target group-pickup peer-group 1
exit
;
dial-peer 5 facility
  destination-pattern *9
  target group-pickup peer-group 2
exit
;
dial-peer 6 sip
  incoming calling number 1003
  incoming dial-plan peer-group 3
exit
;
peer-group 1
  dial-peer 1
  dial-peer 3 1002
  dial-peer 3 1003

```



```

exit
;
peer-group 2
    dynamic-peers
exit
;
peer-group 3
    dial-peer 1
    dial-peer 2
    dial-peer 3
    dial-peer 4
    dynamic-peers
exit

```

Calls made by the voip1/0 interface line 1, have incoming dial-peer 1 and the dial-plan is restricted to the dial-peers in peer-group 3 dial plan. Calls from the SIP 1003 phone have incoming dial-peer 6, which is also associates the dial-plan with peer-group 3, where dial-peer 5 is not present thus preventing the SIP phones from capturing using code \*9.

## 2.2.5 Directed Pickup

You can execute directed pickup by configuring a facility dial-peer whose target is **target directed-pickup prefix <length>. destination-pattern** must match the code a user needs to dial to request pickup. Normal dialing changes for the **strip-digits**, **prefix** and **translation** are applied to the called number. Finally, the directed pickup prefix containing the length indicated in the <length> parameter found in the **directed-pickup prefix** command, is removed from the resulting number. Said number is the extension you want to pickup. Please see the description on the **target directed-pickup** for further details.

*Example 1:*

To example1 (from the previous section), you want to add the possibility of capturing calls through pickup, through capture \*8. In addition, you want to be able to pickup SIP phones by simply dialing the last digit for the extension (e.g. \*81 to pickup 1001).

The capture group (\*81) must coexist with the directed pickup both with the full extension (\*8801 or \*81001) as well as the abbreviation for the SIP phones (\*81).

```

dial-peer 1 voice-port
    destination-pattern 800
    target voice-port voip1/0 1
exit
;
dial-peer 2 voice-port
    destination-pattern 801
    target voice-port voip1/0 2
exit
;
dial-peer 3 sip
    destination-pattern 100.
    target dynamic
exit
;
dial-peer 4 facility
    destination-pattern *8P

    target group-pickup peer-group 1
exit
;
dial-peer 5 facility
    destination-pattern *8T
    target directed-pickup prefix 2
exit
;
dial-peer 6 facility
    destination-pattern *8.P
    outgoing prefix *8100
    outgoing strip-digits 2
    target directed-pickup prefix 2
exit

```

```

;
peer-group 1
dial-peer 1
dial-peer 3 1001
dial-peer 3 1002
exit

```

Dial-peer 4 destination-pattern needs to be modified by adding 'P'. On dialing \*8 from an FXS, directed pickup is not executed and waits until the user indicates end of dialing by pressing '#' or, until time between digits times out. The user can, therefore, continue dialing if they want to carry out a directed pickup.

Dial-peer 5 allows you to execute a directed pickup over any extension, while dial-peer 6 allows you to carry out abbreviated captures using the \*8 pattern transforming the dialed number into \*8100 to capture abbreviated SIP extensions. For \*8 abbreviated capture, you can also place a 'P', which allows you to dial more digits from an FXS to pickup (with the complete extension) using dial-peer 5.

## 2.2.6 Call Forwarding

You can forward a call from a SIP terminal, as well as analog phone, connected to an FXS port. For SIP terminals, configure forwarding in the terminal itself. Please see this operation in the corresponding manual. Given an incoming call, the SIP phone responds with a forwarding 3xx code provoking a new call to the new destination.

From an FXS terminal, you can activate and deactivate unconditional forwarding and no answer. To do this, configure facility dial-peers with a destination-pattern that matches the code the user must dial, together with one of the following targets: **forward-always**, **forward-always-clear**, **forward-noanswer**, **forward-noanswer-clear** and **forward-clear**; so this respectively activates unconditional forwarding, deactivates unconditional forwarding, activates forwarding if unanswered, deactivate forwarding if unanswered and deactivate all forwarding. Unconditional forwarding has priority over forwarding if unanswered when they are both activated. Please see the corresponding commands for more details.

*Example 1:*

To the above section example, you want to add the possibility of carrying out the following forwarding from FXS terminals: activate unconditional forwarding to phones 100. dialing \*21., forwarding if unanswered dialing \*20. and deactivating any forwarding by dialing \*0

```

dial-peer 10 facility
destination-pattern *21.
outgoing prefix *21100
outgoing strip-digits 3
target forward-always prefix 3
exit
;
dial-peer 11 facility
destination-pattern *20.
outgoing prefix *20100
outgoing strip-digits 3
target forward-noanswer prefix 3
exit
;
dial-peer 12 facility
destination-pattern *0
target forward-clear
exit

```

## 2.2.7 Call Hunt Groups

You can configure a call forward sequence when the called phones are busy or if they don't respond in a determined period of time. This is configured, respectively, through the **forward busy peer-group** and **forward no-answer peer-group** commands in the call outgoing dial-peer. Depending on which one you select, the following feature is available.

Case one: if the call destination is busy the call is tried through the first configured **forward busy peer-group** dial-peer that isn't busy; once the other end is ringing, the rest of the dial-peers in the peer-group are ignored.

In the second case, if the destination is busy and the call is not picked up in a configured period of time (**forward no-answer timeout**), the call is tried through the first free **forward no-answer peer-group** dial-peer.

The following is an advanced feature for both hunt groups. Starting point: an outgoing dial-peer is ringing in a hunt group. This has a **forward no-answer peer-group** configured, and doesn't pick up the call within the configured

time. The call is forwarded to the next dial-peer, which is calculated depending on the type of original hunt group: if the original group was a forward no-answer, then the call continues to be forwarded within the original hunt group; if the original group was a forward busy, then the hunt group configured in the dial-peer, which didn't pick up the call, takes the call and begins to forward it.

#### Example 1:

We have the following telephones: three SIP telephones with numbers 1001, 1002 and 1003, two FXS phones from the voip1/0 interface lines 1 and 2 with numbers 800 and 801. If telephones 1001 or 800 are busy, the call is forwarded to the first free telephone between numbers 1001, 1002 and 801.

In the following configuration, if the call is 800 and matches dial-peer1 and 1001 in 4. both have **forward busy peer-group 1** configured.

```
dial-peer 1 voice-port
  destination-pattern 800
  forward busy peer-group 1
  target voice-port voip1/0 1
exit
;
dial-peer 2 voice-port
  destination-pattern 801
  target voice-port voip1/0 2
exit
;
dial-peer 3 sip
  destination-pattern 100.
  target dynamic
exit
;
dial-peer 4 sip
  destination-pattern 1001
  forward busy peer-group 1
  target dynamic
exit
;
dial-peer 4 facility
  destination-pattern *8
  target group-pickup peer-group 1
exit
;
peer-group 1
  dial-peer 3 1001
  dial-peer 3 1002
  dial-peer 2
exit
```

#### Example 2:

We have four FXS telephones on lines 1 to 4 on the voip1/0 interface with numbers 101 to 104. What we want is if telephone 101 is busy, the call forwards to number 102. If 102 is busy or doesn't respond within 10 seconds (you don't need to configure **forward busy**), then the call is forwarded to the first free telephone between 103 and 104; when both of these are busy, the call is forwarded one last time to 102.

```
dial-peer 1 voice-port
  destination-pattern 101
  forward busy peer-group 1
  target voice-port voip1/0 1
exit
;
dial-peer 2 voice-port
  destination-pattern 102
  forward no-answer peer-group 2
  forward no-answer timeout 10s
  target voice-port voip1/0 2
exit
;
dial-peer 3 voice-port
  destination-pattern 103
  target voice-port voip1/0 3
```

```

exit
;
dial-peer 4 voice-port
    destination-pattern 104
    target voice-port voip1/0 4
exit
;
peer-group 1
    dial-peer 2
exit
;
peer-group 2
    dial-peer 3
    dial-peer 4
    dial-peer 2
exit

```

## 2.2.8 IVR

IVR stands for Interactive Voice Response. This is a system capable of establishing calls and interacting with the user via audio recordings.

You can create virtual voip interfaces (please see voip-dummy in manual **bintec Dm770-I VoIP Interfaces** ), which only receive calls, reproduce recorded messages, detect DTMF digits and transfer a call to another number. By configuring message reproduction, you can create telephone numbers, which, when called, provide recorded information. This can be applied to provide a help number or automated information, which can be updated by simply changing the audio file in the device through FTP. This can also be used to give a warning message when a service is out of order, through a dial-peer activated by an NSLA poll. Another possibility is a welcome message in incoming calls before transferring the call to the destination. More complex configurations allow for a number of different voice messages to be configured. A user can navigate interactively through the menus, using DTMF tones, and establish the call with the intended recipient.

A dummy virtual voice interface can be configured in the following operation modes (command **line <id> behavior**):

- **play-loop**: plays a looped recording indefinitely.
- **play-disc**: plays a recording and terminates the call.
- **play-fwd**: plays a recording and forwards to the called number or to a configured fixed number.
- **Play-ivr**: plays a recording and waits to receive DTMF digits from the calling terminal. The device numbering plan is checked to see if the received digits fit a set destination, in which case the call is forwarded to that extension. A translation can be applied to the received extension before the numbering plan is checked. You can also configure an extension to which the call may be diverted if there are errors and, another if the timer expires without receiving any digits.

You can execute advanced IVRs by combining various dummy interfaces.

### Example 1:

In a company office, under normal conditions, you can call four digit corporate number using SIP. These calls are directed to the SIP proxy at address 10.10.0.1. If there are connectivity problems the user will hear a message. This recording tells the user there are temporary problems and he should dial the destination office public number to establish the call over PSTN.

```

add device voip-dummy 1000
;
telephony
; -- Telephony configuration --
    voice-message 1 filename warning.wav
;
dial-peer 1 voice-port
    destination-pattern ....
    target voice-port voip1000 1
    track ns-la-advisor 1
exit
;
dial-peer 2 sip
    destination-pattern ....
    target sip-proxy
exit
;

```

```

dial-peer 3 voice-port
  destination-pattern T
  target voice-port voip1/0 1
  exit
exit
;
network voip1000
; -- VoIP interface Configuration --
  line 1 behavior play-disc
  line 1 play G711 voice-message 1
;
  channels 2
exit

```

This only shows the part of the configuration relative to voice message reproduction. To configure a SIP proxy or an NSLA poll, please see the examples in manual **bintec Dm766-I SIP**. Create a voip-dummy interface with line 1 configured to reproduce message 1, which is a wav in G711 alaw mono format with 8 bits and 8khz sample frequency. The recording is loaded as message 1 through **voice-message 1 filename warning.wav**. Two channels are configured in the dummy interface so it can attend to two calls at once. Under normal circumstances, neither the ns-la-advisor 1 nor the dial-peer 1 would be active and consequently calls with 4 digit patterns are sent to the proxy. When the poll indicates problems with the connection, dial-peer 1 activates and calls to short numbers are established with the dummy interface, which plays the recorded voice message. Dial-peer 3 establishes calls to numbers that are not 4 digits through the voip1/0 voice interface connected to the RTC.

#### Example 2:

You want to modify the previous example so once the voice message warning has been played (call to be sent to a switchboard due to network problems), the number is transferred to 921112233, where the call is taken and established with the required extension

```
line 1 behavior play-fwd 921112233
```

Configure the **play-fwd** behavior in the dummy interface so the call can be transferred to another number once the voice message has finished.

#### Example 3:

Taking the above example, you want to automatically forward to the dialed internal extension number, after the recorded message has been played, adding the prefix 92111. This allows the call to be connected through the public network.

```

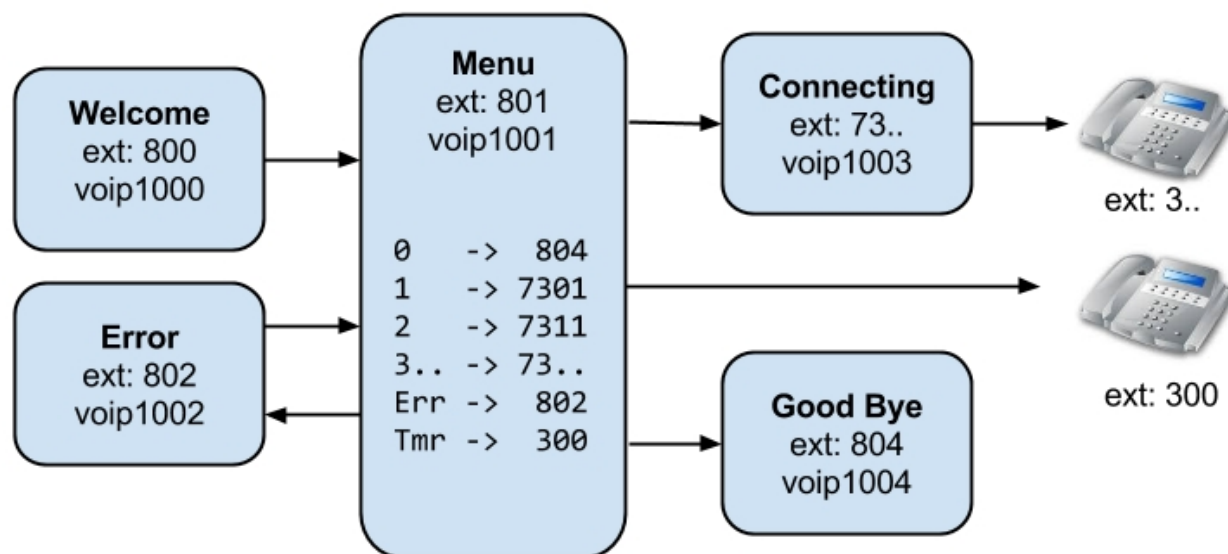
telephony
; -- Telephony configuration --
...
  dial-peer 1 voice-port
    destination-pattern ...
    outgoing prefix 92111
    target voice-port voip1000 1
    track ns-la-advisor 1
  exit
...
network voip1000
; -- VoIP interface Configuration --
  line 1 behavior play-fwd
  line 1 play G711 voice-message 1
;
  channels 2
exit

```

First dial-peer 1 must add the prefix to the called number when sending the call to the dummy interface. The **play-fwd** setting must be configured in the dummy interface without any preconfigured extension number so the call is forwarded to the called number after the recorded message has been played and is connected through dial-peer 3 and the voip1/0 voice interface.

#### Example 4:

You want to configure an automated attendant system or IVR with the settings shown in the following figure.



Incoming calls received on extension number 800 will hear a greeting followed by a recorded message describing the following options:

- Press 1 for sales department (internal extension 301).
- Press 2 for technical support (internal extension 311).
- Enter the internal extension number, if known, starting with 3.
- Press 0 to exit and end the call.
- Wait to connect to the operator (internal extension 300).

If the extension number is incorrect, a message is played (indicating error) and the caller returned to the main menu. In addition, before a call is connected to the internal extension numbers, a message is played, just prior to connection, warning the call may be recorded.

The following is the part of the configuration corresponding to IVR. Configuration relative to the rest of the telephone numbering plan, which support a number of different protocols, terminals and interfaces, is not included here (beyond the intent of this example). Only SIP dial-peer 10 has been included to represent the 3.. internal extensions as registered SIP terminals.

```

add device voip-dummy 1000
add device voip-dummy 1001
add device voip-dummy 1002
add device voip-dummy 1003
add device voip-dummy 1004
;
telephony
; -- Telephony configuration --
voice-message 1 filename welcome.wav
voice-message 2 filename menu.wav
voice-message 3 filename error.wav
voice-message 4 filename connecting.wav
voice-message 5 filename goodbye.wav
;
translation 1
rule 1 match ^0$ 804
;
rule 2 match ^1$ 7301
;
rule 3 match ^2$ 7311
;
rule 4 match ^3..$ 7
rule 4 behaviour prefix
;
exit
;
dial-peer 10 sip
destination-pattern 3..
target dynamic
exit
;

```

```
dial-peer 100 voice-port
  description "Welcome"
  destination-pattern 800
  target voice-port voip1000 1
exit
;
dial-peer 101 voice-port
  description "Menu"
  destination-pattern 801
  target voice-port voip1001 1
exit
;
dial-peer 102 voice-port
  description "Error"
  destination-pattern 802
  target voice-port voip1002 1
exit
;
dial-peer 103 voice-port
  description "Connecting"
  destination-pattern 73..
  outgoing strip-digits 1
  target voice-port voip1003 1
exit
;
dial-peer 104 voice-port
  description "Goodbye"
  destination-pattern 804
  target voice-port voip1004 1
exit
;
exit
;
network voip1000
; -- VoIP interface Configuration --
  line 1 play G711 voice-message 1
  line 1 behavior play-fwd 801

  channels 10
exit
;
network voip1001
; -- VoIP interface Configuration --
  line 1 play G711 voice-message 2
  line 1 behavior play-ivr
  line 1 behavior play-ivr translation 1
  line 1 behavior play-ivr error-fwd 802
  line 1 behavior play-ivr timeout-fwd 300
  line 1 timing initial 7s
;
  channels 10
exit
;
network voip1002
; -- VoIP interface Configuration --
  line 1 play G711 voice-message 3
  line 1 behavior play-fwd 801
;
  channels 10
exit
;
network voip1003
; -- VoIP interface Configuration --
  line 1 play G711 voice-message 4
  line 1 behavior play-fwd
;
  channels 10
```

```

exit
;
network voip1004
; -- VoIP interface Configuration --
    line 1 play G711 voice-message 5
    line 1 behavior play-disc
;
    channels 10
exit

```

A dummy virtual interface is created for each different message. These are all equipped with ten channels to enable ten calls to be processed simultaneously. Each one is assigned a corresponding message and desired behavior.

Dial-peer 100 is the IVR entry point associated with extension 800. The call is connected to the voip1000 virtual interface configured to play the greeting and transfer the call to extension 801.

Dial-peer 101 then connects the call to voip1001 interface. This plays the various message options. The **play-ivr** setting is configured so the call is transferred according to the DTMF digits received. Translation 1 is configured to transform the number received.

When 0 is received, it becomes 804 so the call is forwarded to dial-peer 104 and the voip1004 interface (which plays the end message recording) and hangs up.

When 1 is received, it becomes 7301; 2 becomes 7311 and prefix 7 is added to three digit numbers starting with 3. Thus the call is diverted for connection through dial-peer 103, which removes the 7 (strip-digits 1) and connects the call to voip1003 interface, which plays the connecting message, and forwards the call to the final number.

If digits, which do not exist in the numbering plan after applying the translation, are received in the menu (voip1001), the call is diverted to extension 802 (**line 1 behavior play-ivr error-fwd 802**). This connects the call to voip1002 interface (plays the error message) and diverts back to the main menu (801).

**line 1 timing initial 7s** timer is configured in voip1001 interface so after the recording is played, it waits 7 seconds for the user to enter an option. If no digits are received when this time is up, the call is forwarded to extension 300 (**line 1 behavior play-ivr timeout-fwd 300**), which is the number for the operator who will answer the call manually.

With the proposed configuration, there is nothing to prevent a user from entering one of the 8.. extension numbers used for the IVR system. This can be avoided by adding a translation rule to block such numbers:

```

rule 5 match ^8.. ""
rule 5 behaviour blacklist

```

## 2.2.9 Basic IVR in FXS/FXO interfaces

It's also possible to substitute the signaling tones (dial, ring, busy and error tones) in the device analog interfaces for audio recordings, which allows you to implement IVR basic configurations. Please see **tones** command in the **telephony** menu for global configuration for all ports, or commands for analog line **line <id> dial**, **line <id> alert**, **line <id> busy** y **line <id> error**, for all the dial, ring, busy and error tones respectively.

By using recorded messages, the device can provide IVR basic functional in FXO interfaces: reproduce a recording in incoming calls and wait until the caller dials a number through DTMF.

Behavior for incoming calls in FXO interfaces is configurable. Default is on hook-off the dial tone is heard. The caller must dial a number through DTMF, which is dealt with as a called number and establishes the call based on the device dial-plan. In this case, the user perceives double dialing.

Substituting the dial tone for a recording, which explains what the caller can dial if they have a simple IVR. Additionally, it's possible to configure a number, which is used as called if the user does not dial anything within a determined time. If you configure 0 seconds, the FXO hooks-off and establishes the call directly to the configured number.

*Example:*

An office has two FXS extensions: numbers 800 and 801; three SIP telephones, 1001, 1002 and 1003; and two FXO lines (voip1/0 lines 3 and 4) to receive calls. The aim is when a call is received on line 3, a voice message is played giving the user the option to dial 1 to speak to a salesperson (SIP Telephone 1001), 2 to speak to the technical service (SIP Telephone 1002) and 3 to contact customer services (SIP Telephone 1003 and FXS 801). If, within 20 seconds, the user does not press any key, a call is established with customer services. If a call comes in on line 4, it will immediately be passed to the technical service. Calls to any number, which is not the office, will go through both FXOs.

The following configuration has a 4 FXS/FXO card in spot 1. You configure the first two ports as FXS and 3 and 4 as



FXO. Message 1 is configured in interface 3, substituting the dial tone through **line 3 dial voice-message 1**. This message is an **ivr.wav** file, in alaw 8 bits, mono and 8khz sample frequency, which is in the device flash and contains a recording describing the choices for the user. This file is loaded as message 1 through **voice-message 1 filename ivr.wav** command. **direct-dialing 3 20s** indicates that 20 seconds after hook-off, number 3 is called if the caller has not pressed any key. Line 4 immediately calls line 2 on hook-off (**direct-dialing 2 0s**).

As the dial-plan overlaps (you can call 1 to speak to sales and all SIP telephones that begin with 1), the dial-plan has been divided into two parts. The default dial-plan does not include dial-peers 5, 6, 7 and 8, which are used for IRV options; therefore office extensions cannot dial 1, 2 or 3. A different dial-plan is assigned to FXO lines (**incoming dial-plan peer-group 4**), which only includes said dial-peers so only numbers 1, 2, and 3 can be dialed from an FXO.

```
telephony
; -- Telephony configuration --
  call application dial-plan default peer-group 3
  voice-message 1 filename ivr.wav
;
  dial-peer 1 voice-port
    destination-pattern 800
    target voice-port voip1/0 1
  exit
;
  dial-peer 2 voice-port
    destination-pattern 801
    target voice-port voip1/0 2
  exit
;
  dial-peer 3 voice-port
    destination-pattern T
    incoming dial-plan peer-group 4
    target voice-port voip1/0 3
  exit
;
  dial-peer 4 voice-port
    destination-pattern T
    incoming dial-plan peer-group 4
    target voice-port voip1/0 4
  exit
;
  dial-peer 5 sip
    destination-pattern 100.
    target dynamic
  exit
;
  dial-peer 6 sip
    destination-pattern 1
    outgoing prefix 100
    target dynamic
  exit
;
  dial-peer 7 group
    destination-pattern 2
    target group 1
  exit
;
  dial-peer 8 group
    destination-pattern 3
    target group 2
  exit
;
  peer-group 1
    dial-peer 1
    dial-peer 5 1002
  exit
;
  peer-group 2
    dial-peer 2
    dial-peer 5 1003
```

```

exit
;
peer-group 3
    dial-peer 1
    dial-peer 2
    dial-peer 3
    dial-peer 4
    dynamic-peers
exit
;
peer-group 4
    dial-peer 5
    dial-peer 6
    dial-peer 7
    dial-peer 8
    dynamic-peers
exit
;
exit
;
network voip1/0
; -- VoIP interface Configuration --
    line 1 interface-type fxs
;
    line 2 interface-type fxs
;
    line 3 interface-type fxo
    line 3 dial voice-message 1
    line 3 direct-dialing 3 20s
;
    line 4 interface-type fxo
    line 4 direct-dialing 2 0s
;
exit

```

## 2.3 Call Control

### 2.3.1 Call Control Description

It's often useful if the device is able to reject or vary call routing, depending on the calls already established in the device, or depending on the state of various interfaces. The device is capable of executing call control through two mechanisms:

#### 2.3.1.1 Global call control

Through **call threshold global total-calls** and **call threshold global ip-calls**, you can limit the total number of calls from the device or calls established through IP, respectively. In both cases, a limit (after which calls are rejected) and a lower threshold (after which calls are permitted once more) are configured.

#### 2.3.1.2 Advanced call control

You can configure a very flexible call control so a dial-plan varies depending on the number of calls established from certain sources or destinations, and depending on the device interface state.

This feature is based on the configuration of level indicator filters in the device's **NSLA** Feature (please see *bintec Dm754-I NSLA*). Level indicators are variables, which update depending on the state of various modules in the device. The value of said variables is filtered each time the value is updated (through the NSLA feature filter system and Boolean logic). This process generates a logical value that allows you to activate and deactivate dial-peers, proxy, device ports and even packet routing parameters.

The configuration elements needed to control calls are as follows:

- **update level-indicator <id> value <val> <state>**

Indicates the level indicator <id> with value <val> in a state <state> is updated. This command is available in each dial-peer with the **per-call** and **per-external-call** states, which increases the level indicator corresponding to the given value; for each established call matching this dial-peer (both for incoming and outgoing); and also in SIP dial-

peers with the **when-down** state, to update the level indicator when the SIP dial-peer is in down state due to the **options-keepalive** command.

This configuration element is also available in the network menu for each interface with **when-up**, **when-down**, **when-dorm** states, so the configured level indicator increases with the indicated value if the interface is in the appropriate state.

Finally, in the BRS menu (bandwidth-reservation), this can be configured in each interface and for a class, or at a global level, with **when-rate-exceeds** state; you must also specify a traffic rate and, optionally, a burst size. When this command is run, the sum of the value configured is added to the level indicator if the traffic rate surpasses the established threshold. An advantage of this is that, when you need to monitor several interfaces, you can configure a different bandwidth for each one.

- **track ns-la-advisor <id>**

Level indicators are processed through the NSLA feature with **filter**, **alarm** and **advisor** commands generating a logical value, which can be used in numerous device modules to dynamically modify behavior. As for telephony, it's possible to activate/deactivate dial-peers, the proxy server or device analog ports.

## 2.3.2 Examples

### 2.3.2.1 Local calls not sent to the proxy

You want to configure the device as a local switchboard in an office, with SIP and analog telephones, and a PSTN connection via ISDN. The device will establish local calls between extensions and those destined to the telephone network to fixed numbers. The rest of the calls are signaled to an external SIP server/proxy where, in turn, local extensions are registered so they can be accessed from other locations. The link with PSTN is accessible to establish calls from the SIP server. Therefore calls must be sent with an \* prefix, which is eliminated by the device.

The connection with the SIP server is monitored so if link quality is poor, all calls establish through PSTN. Additionally calls established with the server will be limited (incoming and outgoing), to four if the main WAN interface is active, and to 2 if the backup interface is active. If neither interface is active, all calls are established in local.

The device has a 4FXS/FXO card and an ISDN VOIP 2 BRI card. The configuration for both voip interfaces is as follows:

```
global-profiles dial
; -- Dial Profiles Configuration --
  profile voip default
  profile voip inout
  profile voip isdn-type audio
exit
;
network voip1/0
; -- 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 voip2/0
; -- VoIP interface Configuration --
  base-interface
; -- Base Interface Configuration --
  base-interface bri2/0 255 link
  base-interface bri2/0 255 profile voip
  base-interface bri2/0 255 number-of-circuits all
;
  exit
;
exit
```

**SIP configuration** enables gateway behavior for FXS extensions and ISDN link, and server behavior for local SIP telephones. The external proxy is configured with NSLA polling track, which monitors the quality of connections to ensure the service is active.

```
protocol sip
  application address 172.24.78.152
  application gateway
  application server default
```

```

application server local-ip-registrations
password client bintec proxy 172.24.75.210 default
proxy 172.24.75.210 track nsla-advisor 1
;
exit

```

**local-ip-registrations** mode is configured as server so SIP telephone registers are re-sent towards the proxy with the contact address of the router itself (instead of that corresponding to each telephone). This is done so calls from the server to local SIP telephones are signaled through the router and incoming call control can be applied. In normal operating mode, telephone registers are re-sent to the telephone contact address so the proxy signals calls them directly without passing through the device.

**configuration relative to NSLA** generates two logical values:

```

feature nsm
; -- Network Service Monitor configuration --
  operation 1
; -- NSM Operation configuration --
  type echo ipicmp 172.24.75.210
  frequency 10
  exit
;
  schedule 1 life forever
  schedule 1 start-time now
exit
;
feature nsla
; -- Feature Network Service Level Advisor --
  enable
;
  filter 1 nsm-op 1 rtt
  filter 1 significant-samples 3
  filter 1 activation threshold 200
  filter 1 activation sensibility 50
  filter 1 activation stabilization-time 1
  filter 1 deactivation threshold 150
  filter 1 deactivation sensibility 100
  filter 1 deactivation stabilization-time 1
;
  filter 2 level-indicator 1
  filter 2 activation threshold 5
  filter 2 activation stabilization-time 5
  filter 2 deactivation threshold 5
  filter 2 deactivation stabilization-time 5
;
  alarm 1 filter-id 1
;
  alarm 2 filter-id 2
;
  advisor 1 not alarm-id 1
;
  advisor 2 alarm-id 2
;
exit

```

- **advisor 1:** indicates if the proxy is active based on connectivity polling where it is considered down if the RTT for an icmp poll exceeds 200 ms in 2 out of 3 tests. A test is carried out every 10 seconds; therefore detection time is 20 seconds. To re-establish the server, the result of 3 consecutive tests must be below 150 ms.

```

network ppp1
  description "WAN"
  ...
  update level-indicator 1 value 40 when-up
;
exit
;
network ppp2
  description "WAN backup"
  ...

```

```

update level-indicator 1 value 20 when-up
exit
;
feature wrr-backup-wan
; -- WAN Reroute Backup user configuration --
pair 1 primary interface pppl secondary interface ppp2
;
enable
exit

```

- **advisor 2:** indicates if you can establish calls from/to the SIP Proxy. Four calls are permitted with the main WAN interface active, while if the backup interface is active, only two calls are allowed. Using **update level-indicator** for the corresponding nets, the level-indicator increases 1 in 40 and 20 respectively, when the interfaces are active. Each counted call subtracts 10 so the threshold of 5 is surpassed after 4 and 2 calls respectively, the advisor 2 deactivating when this limit is reached. Said advisor activates/deactivates the dial-peer that sends/receives external calls.

This dial-plan is configured in the **TLPHY** menu:

```

feature access-lists
; -- Access Lists user configuration -
access-list 1
entry 1 default
entry 1 permit
entry 1 source address 172.24.75.210 255.255.255.255
;
exit
;
exit
;
telephony
; -- Telephony configuration --
call application incoming-match force
call application outgoing-match force
call application dial-plan default peer-group 1
dial-peer 1 voice-port
description "Extension 2001"
destination-pattern 2001
target voice-port voip1/0 1
exit
;
dial-peer 2 voice-port
description "Extension 2002"
destination-pattern 2002
target voice-port voip1/0 2
exit
;
dial-peer 3 voice-port
description "Extension 2003"
destination-pattern 2003
target voice-port voip1/0 3
exit
;
dial-peer 4 voice-port
description "Extension 2004"
destination-pattern 2004
target voice-port voip1/0 4
exit
;
dial-peer 5 group
destination-pattern 2000
target group 3
exit
;
dial-peer 10 sip
description "Local sip phones"
destination-pattern 20..
target dynamic

```

```
exit
;
dial-peer 20 voice-port
  description "PST calls"
  destination-pattern 9.....
  destination-pattern 8.....
  destination-pattern 7.....
  incoming prefix 2000
  incoming strip-digits 9
  target voip2/0 1
exit
;
dial-peer 30 sip
  description "External proxy"
  destination-pattern T
  incoming access-list 1
  incoming called number T
  incoming dial-plan peer-group 2
  target sip-proxy
  track nsla-advisor 2
  update level-indicator 1 value -10 per-call
exit
;
dial-peer 31 voice-port
  description "PSTN backup"
  destination-pattern T
  target voip2/0 1
exit
;
dial-peer 40 voice-port
  description "PSTN gateway from proxy with prefix *"
  destination-pattern *T
  outgoing strip-digits 1
  target voip2/0 1
exit
;
peer-group 1
  dial-peer 1
  dial-peer 2
  dial-peer 3
  dial-peer 4
  dial-peer 5
  dial-peer 10
  dial-peer 20
  dial-peer 30
  dial-peer 31
  dynamic-peers
exit
;
peer-group 2
  dial-peer 1
  dial-peer 2
  dial-peer 3
  dial-peer 4
  dial-peer 5
  dial-peer 10
  dial-peer 40
  dynamic-peers
exit
;
peer-group 3
  dial-peer 1
  dial-peer 2
  dial-peer 3
  dial-peer 4
exit
;
```

```
exit
```

- Local extensions take pattern **20..** and FXS analog extensions are configured in dial-peers 1-4. In dial-peer 5 simultaneous ringing for the four analog extensions is configured and in dial-peer 10, the SIP telephones that dynamically register.
- Dial-peer 20 sends calls destined to fixed number via ISDN interface. Calls entering from ISDN match as incoming, substituting the called number for 2000, which makes the four FXS extensions ring simultaneously.
- Dial-peer 30 sends calls with any pattern to the configured SIP server. As the destination pattern is less specific than previous ones, this matches outgoing for numbers that do not match the previous patterns ones. Each call this dial-peer has established, updates the level-indicator in -10, so the calls can be counted. Said dial-peer passes to inactive when the proxy is considered down, **proxy 172.24.75.210 track nsia-advisor 1** in sip protocol, or when the limit for the calls used in this dial-peer is reached, track nsia-advisor 2. So incoming calls are counted as well, the device forces all incoming calls (from the proxy) to match as incoming for this dial-peer. This is achieved through **incoming access-list 1**, which only allows matching from proxy ip (172.24.75.210), and **incoming called number T** command, which matches incoming for any called number.
- Dial-peer 31 is a backup route permitting calls (which should go through the SIP server in normal conditions) through the ISDN interface. As this has the same destination-pattern as dial-peer 30, when dial-peer 30 is not active, all calls exit through 31.
- Dial-peer 40 is used so the server sends calls over ISDN interface and places an asterisk in front of the called number. This asterisk is eliminated by the device before signaling the call over ISDN.

At a global level, **call application incoming-match force** is configured so if there aren't any incoming matching dial-peers, the call is rejected. This allows you to reject incoming calls from the server when dial-peer 30 is inactive (maximum number of calls reached).

Two dial-plans can be configured with the defined dial-peers:

- The default dial-plan **call application dial-plan default peer-group 1** includes all dial-peers except for *dial-peer 40*, so local users cannot exit through ISDN by dialing \* in front of the number.
- Calls from the SIP proxy have a different dial-plan, **incoming dial-plan peer-group 2**, which includes local extensions, and dial-peer 40 for calls exiting through ISDN. This does not include dial-peers 20, 30 and 31, which are for locally sourced calls.

### 2.3.2.2 Local calls not sent to the proxy



#### Note

This example is based on the previous one. We recommend reading example a).

You wish to configure the device as a local switchboard in an office with SIP telephones, analog telephones and connection to PSTN through ISDN. The difference between this and the previous example is you want all calls (both local and external), to be sent to the proxy to be processed. If the call is local, the proxy will return it to the device.

The local calls should not be taken into account when limiting the total number of calls, as audio packets are sent over LAN (and not WAN). Supposing the local network has addresses 172.24.100.0, an access list is added for classification.

```
feature access-lists
; -- Access Lists user configuration --
access-list 1
    entry 1 default
    entry 1 permit
    entry 1 source address 172.24.75.210 255.255.255.255
;
exit
;
access-list 2
    entry 2 default
    entry 2 permit
    entry 2 source address 172.24.100.0 255.255.255.0
;
exit
;
exit
```

In the telephony menu, associate said access list with **local calls through call application local-networks access-list 2**.

Telephony configuration is as follows:

```
telephony
; -- Telephony configuration --
call application incoming-match force
call application outgoing-match force
call application local-networks access-list 2
dial-peer 30 sip
  description "External proxy, por defecto todo se envia al proxy"
  destination-pattern 2000
  destination-pattern 2001
  destination-pattern 2002
  destination-pattern 2003
  destination-pattern 2004
  destination-pattern 20..
  destination-pattern T
  incoming access-list 1
  incoming called number T
  incoming dial-plan peer-group 2
  target sip-proxy
  track nsla-advisor 2
  update level-indicator 1 value -10 per-external-call
exit
;
dial-peer 1 voice-port
  description "Extension 2001"
  destination-pattern 2001
  target voice-port voip1/0 1
exit
;
dial-peer 2 voice-port
  description "Extension 2002"
  destination-pattern 2002
  target voice-port voip1/0 2
exit
;
dial-peer 3 voice-port
  description "Extension 2003"
  destination-pattern 2003
  target voice-port voip1/0 3
exit
;
dial-peer 4 voice-port
  description "Extension 2004"
  destination-pattern 2004
  target voice-port voip1/0 4
exit
;
dial-peer 5 group
  destination-pattern 2000
  target group 3
exit
;
dial-peer 10 sip
  description "Local sip phones"
  destination-pattern 20..
  target dynamic
exit
;
dial-peer 20 voice-port
  description "PSTN calls"
  destination-pattern 9.....
  destination-pattern 6.....
  incoming prefix 2000
  incoming strip-digits 9
  target voice-port voip2/0 1
exit
```



```

;
peer-group 2
  dial-peer 1
  dial-peer 2
  dial-peer 3
  dial-peer 4
  dial-peer 5
  dial-peer 10
  dial-peer 20
  dynamic-peers
exit
;
peer-group 3
  dial-peer 1
  dial-peer 2
  dial-peer 3
  dial-peer 4
exit
;
exit

```

As already seen in the dial plan, all calls are sent to the proxy (if this is active). Subsequently, if call destination is a telephone from the device itself, the proxy returns said call to the device with the same number.

As calls coming from the proxy cannot be resent to said proxy as (this is prevented through **incoming dial-plan**), the call is then sent to the corresponding dial-peer.

The proxy isn't accessible either because it is out of reach (track 1) or because the maximum number of external calls has been surpassed (track 2): this is normally four and two when the device is in backup. Under these circumstances, calls to local extensions are processed by the device and calls to external telephones are sent via ISDN.

### 2.3.2.3 Call control depending on bandwidth occupied

In this example, the device behaves as a local switchboard in an office with SIP telephones with a PSTN connection through ISDN. Calls destined to local extensions or to telephone network numbers are established by the device: the rest of them are signaled through an external SIP server/proxy, where local extensions are also registered so they can be called from other offices.

Through bandwidth reservation system (BRS), bandwidth used in the connection with SIP proxy is monitored so, if the traffic threshold (200 Kbps) is surpassed, the link is not further overloaded and calls are established through PSTN. To classify voice traffic, all packets must be marked with a TOS 184 (in hexadecimal, 0xB8).

The device has an ISDN VOIP 2 BRI card for PSTN connection. The **voip** interface configuration is as follows:

```

global-profiles dial
; -- Dial Profiles Configuration --
  profile voip default
  profile voip inout
  profile voip isdn-type audio
exit
;
network voip1/0
; -- VoIP interface Configuration --
  base-interface
; -- Base Interface Configuration --
  base-interface bril/0 255 link
  base-interface bril/0 255 profile voip
  base-interface bril/0 255 number-of-circuits all
;
exit

```

SIP configuration enables this to act as a gateway for the ISDN link and, as server for local SIP telephones: the external proxy can also be configured.

```

protocol sip
  application address 172.24.78.152
  application gateway
  application server default
  application server local-ip-registrations
  password client bintec

```

```

proxy 172.24.75.210 default
;
exit

```

Configure **local-ip-registrations** mode as server, so SIP telephone registrations are re-sent to the proxy with the contact address of the router itself (instead of that corresponding to each telephone). This is so calls from the server to local SIP telephones are signaled through the router and incoming calls can be dropped if the link becomes congested.

For bandwidth reservation, first create an extended access list to define the class traffic. Given that all voice traffic is marked with TOS 184, simply set this limitation. Within the BRS menu, class traffic is set as real time to provide it with greater priority and indicator level updating is configured.

```

feature access-lists
; -- Access Lists user configuration -
access-list 100
entry 1 default
entry 1 permit
entry 1 tos-octet 184
exit
;
exit
;
feature bandwidth-reservation
; -- Bandwidth Reservation user configuration --
network ppp1
enable
class local 10
;
class default 40
;
class teleph 100 real-time
class teleph update level-indicator 1 value 10 when-rate-exceeds 200
;
access-list 100 teleph
;
exit
;
exit

```

NSLA configuration obtains a logical value with advisor 1, which indicates if ppp1 link, used to communicate with the proxy, is available to route new calls. If, in said link, the traffic rate has surpassed 200 Kbps, and this condition continues for longer than 5 seconds, advisor 1 gives a false logical value. The advisor activates/deactivates the dial-peer receiving/sending external calls.

```

feature nsla
; -- Feature Network Service Level Advisor --
enable
;
filter 1 level-indicator 1
filter 1 activation threshold 5
filter 1 activation stabilization-time 5
filter 1 deactivation threshold 5
filter 1 deactivation stabilization-time 5
;
alarm 1 filter-id 1
;
advisor 1 not alarm-id 1
;
exit

```

Finally, the dial-plan is configured in the TLPHY menu. Prior to that, create another access list to classify traffic from the proxy.

```

feature access-lists
; -- Access Lists user configuration -
access-list 1
entry 1 default
entry 1 permit
entry 1 source address 172.24.75.210 255.255.255.255

```

```

    exit
;
exit
;
telephony
; -- Telephony configuration --
    call application incoming-match force
    call application outgoing-match force
    rtp ip-tos 0xb8
;
    dial-peer 1 group
        description "All local sip phones ring"
        destination-pattern 2000
        target group 1
    exit
;
    dial-peer 10 sip
        description "Local sip phones"
        destination-pattern 20..
        target dynamic
    exit
;
    dial-peer 20 voice-port
        description "PSTN calls"
        destination-pattern 9.....
        destination-pattern 8.....
        destination-pattern 7.....
        incoming prefix 2000
        incoming strip-digits 9
        target voice-port voip1/0 1
    exit
;
    dial-peer 30 sip
        description "External proxy"
        destination-pattern T
        incoming access-list 1
        incoming called number T
        incoming dial-plan peer-group 2
        target sip-proxy
        track nsla-advisor 1
    exit
;
    dial-peer 31 voice-port
        description "PSTN backup"
        destination-pattern T
        target voice-port voip1/0 1
    exit
;
    peer-group 1
        dynamic-peers
    exit
;
    peer-group 2
        dial-peer 1
        dial-peer 10
        dynamic-peers
    exit
;
exit

```

- Dynamically registered local SIP telephones have pattern 20.. and are configured in dial-peer 10.
- Dial-peer 20 sends calls destined to fixed numbers through the ISDN interface. Calls coming in from ISDN match at input, substituting the called number for 2000, which makes all local SIP telephones ring.
- Dial-peer 30 sends calls with any pattern to the configured SIP proxy. As this destination-pattern is less specific than previous ones, it matches outgoing calls for numbers that don't match the previous ones. This dial-peer becomes inactive when the outgoing traffic rate in the link surpasses 200 Kbps. To reject calls, this dial-peer makes all calls coming in from the proxy match at input. This is achieved through **incoming access-list 1**, which only per-

mits matching from proxy IP (172.24.75.210), and **incoming called number T**. Calls from the SIP proxy have **incoming dial-plan peer-group 2** as their dial-plan, which includes local extensions: dial-peers 20, 30 and 31, used for locally originated calls, are not added.

- Dial-peer 31 is the backup route that routes calls (that normally go through the SIP server) through the ISDN interface. As this has the same destination-pattern as dial-peer 30, when dial-peer 30 isn't active, all calls exit through dial-peer 31.

**call application incoming-match force** is configured at the global level, so that if none of the incoming dial-peers match the call is dropped. This lets you reject incoming calls from the server when dial-peer 30 isn't active.

## 2.4 Telephony Configuration Commands

Access telephony configuration (Voice over IP), through the main menu in the following way:

- (1) At the (\*) prompt, enter **config** or **running-config** if you want the configuration changes to take immediate effect.
- (2) At the configuration prompt (Config>), enter **telephony**.
- (3) At the telephony configuration prompt (Telephony Config>), use the configuration commands.

This chapter enumerates and describes the generic telephony configuration commands. All configuration commands must be entered at the Telephony Config> prompt.

The **no telephony** command (in the telephony menu) deletes all the configuration from the menu.

### 2.4.1 [NO] CALL

*Syntax:*

```
Telephony Config>CALL ?
  admission-control  Admission control configuration
  application        Application configuration parameters
  threshold         Threshold values
```

*where:*

```
admission-control  Admission control configuration
  event            Send call admission event
  false-limit      FALSE call limit
  id1              ID1 in CAC event
  id2              ID2 in CAC event
  track            CAC event tracking configuration
  true-limit       TRUE call limit
application        Application configuration parameters
  dial-plan        Dial plan configuration
  default          Configure default dial-plan
  sip-dynamic      Configure dynamic peers dial-plan
  history          Call history track configuration parameters
  file             Specify a file to dump the cdr
  max-records      Maximum number of history records to store
  incoming-match   Incoming dial-peer matching configuration
  force            Call must find an incoming dial-peer
  dynamic-search   Search incoming dial-peer among dynamic peers
  transfer-keep    Keep incoming dial-peer in a transfer when possible
  local-networks   Access list to define the local network
  access-list      Access-list used to classify
  outgoing-match   Outgoing dial-peer matching configuration
  force            Call must find an outgoing dial-peer
threshold         Threshold values
  global           Global limitation parameters
  ip-calls         Limit the number of ip calls
  total-calls      Limit the number of total calls
```

### 2.4.1.1 [NO] CALL ADMISSION-CONTROL EVENT

Configures the appearance of a TLPHY.026 event when the NSLA subsystem advisor state changes (see manual *bintec Dm754-I NSLA*) from the **track nsal-advisor <id>** sub-option. Two text identifiers appear in this event (configured through **id1 <text>** and **id2 <text>** sub-options respectively) and a numeric value that depends if the advisor output is true or false. These values are configured through **true-limit <num>** and **false-limit <num>** sub-options respectively. You can also configure the number of repetitions for each event and the time between them through **repeat <num>** and **timer <tim>** suboptions. Default is no repetition. If there are repetitions, time between them is 10 seconds. This command is useful as the telephony central server can receive this event and adjust its Call Admission Control (CAC) to the limit given in said event. The NSLA advisor indicates if the device is in normal operating mode or backup, so the event indicates the limit of calls corresponding to each situation.

#### Syntax:

```
Telephony Config>CALL ADMISSION-CONTROL EVENT TRACK NSLA-ADVISOR <ID>
Telephony Config>CALL ADMISSION-CONTROL EVENT ID1 <TEXT>
Telephony Config>CALL ADMISSION-CONTROL EVENT ID2 <TEXT>
Telephony Config>CALL ADMISSION-CONTROL EVENT TRUE-LIMIT <NUM>
Telephony Config>CALL ADMISSION-CONTROL EVENT FALSE-LIMIT <NUM>
Telephony Config>CALL ADMISSION-CONTROL EVENT REPEAT <NUM>
Telephony Config>CALL ADMISSION-CONTROL EVENT TIMER <TIM>
Telephony Config>
```

### 2.4.1.2 [NO] CALL APPLICATION DIAL-PLAN DEFAULT

Configures the peer-group to be used as a list of permitted outgoing dial-peers by default. Calls can only be established when exiting through said dial-peers. If **call application dial-plan sip-dynamic** is configured and the call has a dynamic dial-peer as incoming, the peer-group configured with this command is used rather than the default peer-group. The dial-plan can also be configured in the incoming dial-peer through **incoming dial-plan**. In this case, said peer-group is used instead of the default group. If the default dial-plan is not configured and a call does not have a particular dial-plan, an attempt is made to establish the call by searching through all existing dial-peers.

#### Syntax:

```
Telephony Config>CALL APPLICATION DIAL-PLAN DEFAULT PEER-GROUP <id>
Telephony Config>
```

### 2.4.1.3 [NO] CALL APPLICATION DIAL-PLAN SIP-DYNAMIC

Configures the **dial-peer** to be used as a list of permitted outgoing dial-peers in calls whose incoming dial-peer is a dynamic dial-peer. If this is not configured, an attempt is made to establish the call by searching through all existing dial-peers.

#### Syntax:

```
Telephony Config>CALL APPLICATION DIAL-PLAN SIP-DYNAMIC PEER-GROUP <id>
Telephony Config>
```

### 2.4.1.4 [NO] CALL APPLICATION HISTORY MAX-RECORDS

Configures the maximum number of call registers (CDRs) to save in RAM memory. These registers can be consulted in the monitoring telephony menu through **list cdr released**. **no** restores the default value (100 registers).

#### Syntax:

```
Telephony Config>CALL APPLICATION HISTORY MAX-RECORDS <max-records>
Telephony Config>
```

### 2.4.1.5 [NO] CALL APPLICATION HISTORY FILE

Configures a file to save call registers (CDRs) in. Specifies a file name, time interval and how often to save. Optionally, you can specify a maximum length of time a register can remain in the file and maximum size of said file. By default, maximum time is 7 weeks and maximum size is 100000 Bytes. **no** eliminates the configuration of a file to save registers.

It's very important the time period between file updates does not surpass the maximum number of calls permitted in RAM. If it does, the latest calls overwrite the oldest and call data is lost when saving to disk. Use **call application history max-records** to configure the maximum number of registers to be saved in RAM.

An important aspect to bear in mind while recording is that voice quality may be affected. Given this, we suggest pro-

gramming a recording time when the device isn't in use.

*Syntax:*

```
Telephony Config>CALL APPLICATION HISTORY FILE <name> rec-interval <start-save> <end-save> each <save-period> max-age <age> max
```

#### 2.4.1.6 [NO] CALL APPLICATION INCOMING-MATCH FORCE

Configures telephony call function in the device so, if there are no dial-peers matching as incoming dial-peers, the call is rejected. **no** restores the default value for this parameter so an incoming dial-peer does not have to match to forward a call. If no dial-peer matches, dial-peer 0 is used (it has all the default values). Chapter 2, section 2.a) explains the mechanism to search for the incoming dial-peer for a call.

*Syntax:*

```
Telephony Config>CALL APPLICATION INCOMING-MATCH FORCE
```

#### 2.4.1.7 [NO] CALL APPLICATION INCOMING-MATCH DYNAMIC-SEARCH

This searches dynamic dial-peers registered in the device to see if the incoming call matches one of them. Default is not configured (so the process of establishing a call is not slowed down). This command is useful when **call application incoming-match force** is configured and there is no static dial-peer that matches calls coming from registered telephones (otherwise calls would be rejected). With this command configured, the corresponding incoming dynamic dial-peer is found and the call progressed.

*Syntax:*

```
Telephony Config>CALL APPLICATION INCOMING-MATCH DYNAMIC-SEARCH
```

#### 2.4.1.8 [NO] CALL APPLICATION INCOMING-MATCH TRANSFER-KEEP

This command makes a call transfer, which requires a new call to be established, consulting the dial-plan to do this. The incoming dial-peer for the original call is retained instead of searching for one to match the new transferred call. This is applied to all blind transfers when the transfer is a device port, and when attended, only when the device port is transferred from SIP.

*Syntax:*

```
Telephony Config>CALL APPLICATION INCOMING-MATCH TRANSFER-KEEP
```

#### 2.4.1.9 [NO] CALL APPLICATION LOCAL-NETWORKS ACCESS-LIST

Configures an access list so IP addresses, which match said list, are considered as pertaining to the local network. IP addresses used in this check are call RTP addresses. If a call has local addresses both in the incoming as well as in the outgoing, it's considered to be a local call. If one of the parts, either the incoming or the outgoing, has a local address and the other an external RTP address, it's considered an external call.

Through **update level-indicator**, you can only modify an indicator depending on the external calls, which use a dial-peer.

*Syntax:*

```
Telephony Config>CALL APPLICATION LOCAL-NETWORKS ACCESS-LIST <access-list>
```

#### 2.4.1.10 [NO] CALL APPLICATION MAX-REDIR

Configures the number of times a call can be redirected due to call forwarding. Default is 5.

*Syntax:*

```
Telephony Config>CALL APPLICATION MAX-REDIR <redir-num>
```

#### 2.4.1.11 [NO] CALL APPLICATION OUTGOING-MATCH FORCE

Configures the telephony calls function in the device so, if none of the dial-peers matches as an outgoing dial-peer, the call is not progressed. By using **no**, you restore the default value for this parameter (if an outgoing dial-peer does not match an incoming SIP call and a SIP proxy has been configured, the call is sent to the proxy).

*Syntax:*

```
Telephony Config>CALL APPLICATION OUTGOING-MATCH FORCE
```

### 2.4.1.12 [NO] CALL THRESHOLD GLOBAL IP-CALLS

Establishes a maximum limit (number) for calls over VoIP the device supports. *All calls are counted, except local calls between conventional telephony ports.* Said maximum limit is important so no further calls can be made once this is reached. A lower value must also be specified so, once reached, calls are reestablished.

#### Syntax:

```
Telephony Config>CALL THRESHOLD GLOBAL IP-CALLS LOW lower-limit HIGH higher-limit
Telephony Config>
```

#### Example:

You want to limit the number of calls over VoIP to three.

```
Telephony Config>CALL THRESHOLD GLOBAL IP-CALLS LOW 3 HIGH 3
Telephony Config>
```

### 2.4.1.13 [NO] CALL THRESHOLD GLOBAL TOTAL-CALLS

Establishes a maximum limit for the total number of calls the device supports. Specify a maximum limit for calls, so no further calls can be made once this is reached. A lower value must also be set so, once reached, calls are reestablished.

#### Syntax:

```
Telephony Config>CALL THRESHOLD GLOBAL TOTAL-CALLS LOW lower-limit HIGH higher-limit
Telephony Config>
```

#### Example:

You want to limit the total number of calls to three, and allow calls to be made again when this drops to two.

```
Telephony Config>CALL THRESHOLD GLOBAL TOTAL-CALLS LOW 2 HIGH 3
Telephony Config>
```

## 2.4.2 [NO] CALLED-NUMBER KEEP-INT

When this command is configured, special character '+' is treated as a regular one when entered at the beginning of the called number.

#### Syntax:

```
Telephony Config>called-number keep-int
```

#### Command history:

Release	Modification
10.09.29	This command was introduced as of version 10.09.29.
11.00.07	This command was introduced as of version 11.00.07.
11.01.03	This command was introduced as of version 11.01.03.

## 2.4.3 [NO] DIAL-PEER

To define a dial-peer and configure the parameters for this, use **dial-peer** followed by the dial-peer identifier you wish to configure. For further information on this command, please see the section 1 on Dial-Peers in this chapter.

At this point, enter the dial-peers configuration menu. You can eliminate a dial-peer from the configuration through **no dial-peer <id>**.

#### Syntax:

```
Telephony Config>DIAL-PEER <id> <type>
```

Id: dial-peer identifier, number between 1 and 10000. Type: type of dial-peer, currently voice-port, sip, h323, noe, sccp, facility, direct and group are supported. The configurable parameters are described below.

### 2.4.3.1 [NO] 183 EARLY-ANSWER

Calls coming in through this dial-peer are answered before the other end hooks-off, as they have the **early-answer** option configured. E.g. when a configured voice message is played, these are signaled to the caller with a partial 183 Session Progress response instead of a 200 OK. This option is only available in SIP dial-peers.

*Syntax:*

```
dial-peer 1 config>183-EARLY-ANSWER
```

### 2.4.3.2 [NO] 183-LOCAL-RING

If you configure this option, on receiving a “183 Session Progress” response to an outgoing call INVITE, such as SIP Gateway, a local ring tone is generated in the same way as if it were receiving a “180 Remote Altered” response. By default, this behavior is disabled. The 183 response is processed normally, establishing audio before hook-off if this has SDP.

*Syntax:*

```
dial-peer 1 config>183-LOCAL-RING
```

### 2.4.3.3 [NO] AUTH-USER

Configures a user name for authentication in an external SIP server. Both registers and calls may be authenticated:

For registers, Voice-port and group dial-peers, which have **destination-alias** or **destination-pattern** configured, without wildcards, register said extension in the configured external SIP server. This command specifies a user for a different authentication from the extension itself, which is registered. If not configured, the authentication user and the registered extension coincide.

For calls, if asked to authenticate an INVITE or Re-INVITE, the device uses the first of the following for user authentication: a user configured through **auth-user** in the dial-peer corresponding to the endpoint asking for authentication (usually outgoing), a user configured through **auth-user** in the dial-peer opposite the endpoint asking for authentication (usually incoming), the telephone number/user ID of the party opposite the endpoint asking for authentication (usually the calling phone).

See also the **password** command, which specifies the password used in the authentication.

*Syntax:*

```
dial-peer 1 config>AUTH-USER <text>
```

### 2.4.3.4 [NO] BULK-REGISTER

This command is available in voice-port and group dial-peers. If the extension configured in this dial-peer is going to be registered in the SIP Proxy, bulk number contact ( **bnc**) will be used to register multiple phone numbers.

*Syntax:*

```
dial-peer 1 config>BULK-REGISTER
```

#### Command history:

Release	Modification
11.01.03	This command was introduced as of version 11.01.03

### 2.4.3.5 [NO] CODEC

Specifies a specific codec to use with a dial-peer. If none is specified (and **voice-class** isn't either), all codecs are permitted by default in the following order: g7235k3, g7236k4, g729, g711alaw, g711ulaw.

*Syntax:*

```
dial-peer 1 config>CODEC <codec> FRAMES-PER-PACKET <frames>
```

*Codec:* codec to use: the following codecs are supported:

g711alaw	G711 A law at 64 kbps.
g711ulaw	G711 U law at 64 kbps.
g7235k3	G723 at 5.3 kbps.
g7236k4	G723 at 6.4 kbps.



g729

G729 at 8 kbps.

*Frames:* number of voice frames sent in each rtp packet. If none is specified, the number of frames per rtp packet will depend on the codec. Please see the **voice-class** command for further information.

### 2.4.3.6 [NO] DESCRIPTION

Character string to configure a description of a dial-peer. This is for informative purposes.

*Syntax:*

```
dial-peer 1 config>DESCRIPTION <description>
```

### 2.4.3.7 [NO] DESTINATION-ALIAS

Specifies a character string to match a dial-peer. This is useful if the dial-peer destination is a name instead of a number pattern, or if you want to register this dial-peer in an SIP register with a character string (if you configure destination-alias, this value is used instead of the destination-pattern for SIP registers). Please note the comparisons between a telephone number with a configured destination-alias are literal, i.e. the **.** does not represent any digit and the **T** does not represent an undetermined number of characters. For an incoming dial-peer voice-port for a line without a calling number configured, the destination-alias is used as calling.

*Syntax:*

```
dial-peer 1 config>DESTINATION-ALIAS <alias>
```

### 2.4.3.8 [NO] DESTINATION-PATTERN

Specifies the number pattern matching a dial-peer. To represent a variable digit, insert a **.**; to indicate a variable length string, end it with a **T**; and to represent a digit range, use the syntax **[digit1-digit2]**. A maximum of 20 characters are accepted (0 to 9 **.** **T** **[ - and ]**). In the case of an incoming dial-peer voice-port for a line that has no calling number or destination-alias configured in the dial-peer, a destination-pattern is used as calling, provided this does not contain variable symbols (**.** and **T**).

*Syntax:*

```
dial-peer 1 config>DESTINATION-PATTERN <pattern>
```

#### Command history:

Release	Modification
11.01.09	Digit range support was added as of version 11.01.09.

### 2.4.3.9 [NO] DISPLAY-NAME

In calls between a dial-peer voice-port and a SIP, the dial-peer voice-port display-name is used as “name to display” in SIP signaling.

*Syntax:*

```
dial-peer 1 config>DISPLAY-NAME <name>
```

### 2.4.3.10 [NO] DTMF-RELAY H245-ALPHANUMERIC

For this H323 dial-peer, dtmf digits are sent through the H245 data channel.

*Syntax:*

```
dial-peer 1 config>DTMF-RELAY H245-ALPHANUMERIC
```

### 2.4.3.11 [NO] DTMF-RELAY H245-SIGNAL

For this H323 dial-peer, dtmf digits are sent through the H245 data channel in signal mode.

*Syntax:*

```
dial-peer 1 config>DTMF-RELAY H245-SIGNAL
```

### 2.4.3.12 [NO] DTMF-RELAY IN-BAND

Specifies the digits to be sent in band as normal audio. This may mean for codecs with very strong compressions (G723, G729), the dtmf digits won't be heard correctly.

*Syntax:*

```
dial-peer 1 config>DTMF-RELAY IN-BAND
```

### 2.4.3.13 [NO] DTMF-RELAY RFC2833

Specifies the digits sent comply with the RFC 2833, i.e. as special packets within voice rtp flow. This is default in SIP dial-peers.

*Syntax:*

```
dial-peer 1 config>DTMF-RELAY RFC2833
```

### 2.4.3.14 [NO] EARLY-ANSWER VOICE-MESSAGE

If you configure this option, calls entering through this dial peer are answered by the router immediately before the other end hooks-off. While the other end does not hook-off, the configured voice message is played. When the other end hooks-off, the conversation is heard normally. This avoids a situation of uncomfortable silence (during negotiation time) at the beginning of the conversation. This option is only available for SIP and H323 dial peers.

*Syntax:*

```
dial-peer 1 config>EARLY-ANSWER VOICE-MESSAGE <loc>
```

### 2.4.3.15 [NO] FAX MODE T38-FORCE

Indicates a dial-peer is associated with a fax device that only uses T38 protocol. At the time the call is established, only T38 capacity is sent, not audio. You can also initiate the call with audio codecs and switch to T38, after a defined period of time, using the delay option.

This fax mode takes precedence over all others, i.e., if any dial-peer involved in the call, whether incoming or outgoing, has **t38-force** configured, then the call is made in t38-force mode.

*Syntax:*

```
dial-peer 1 config>FAX MODE T38-FORCE
```

### 2.4.3.16 [NO] FAX MODE T38-DETECT

Indicates a dial-peer may be associated with a fax device using T38 protocol. The call begins with normal audio opening an RTP channel. However, if a fax device is detected, renegotiation occurs to establish the T38 channel and close the RTP channel.

Said fax mode takes precedence over the transparent mode, i.e., if any dial-peer involved in the call, incoming or outgoing, has **t38-detect** mode configured, then the call is made in t38-detect, even if the rest are in transparent mode.

*Syntax:*

```
dial-peer 1 config>FAX MODE T38-DETECT
```

### 2.4.3.17 [NO] FAX MODE TRANSPARENT

Indicates T38 protocol must not be used even if the call is detected as a fax call. Negotiation is carried out as for a normal audio call and the fax is encoded/decoded as if it is dealing with voice. This is default and the least preferential, i.e., if, for example, a call is configured in **transparent** mode in the outgoing dial-peer but **t38-detect** in the incoming, the call is executed using t38-detect.

*Syntax:*

```
dial-peer 1 config>FAX MODE TRANSPARENT
```

### 2.4.3.18 [NO] FAX T38 ECM

Enables or disables error correction mode for a fax call, which uses T38. Default is enabled. However, if it's disabled in any dial-peer forming part of the call, this initiates with ECM disabled.

*Syntax:*

```
dial-peer 1 config>FAX T38 ECM
```

**2.4.3.19 [NO] FAX T38 MAX-SPEED**

Configures the maximum fax speed for a call using T38. The possible values are (in bits per second) 2400, 4800, 7200, 9600, 12000 and 14400. Default is 14400. Maximum call speed is the lowest maximum value configured in the dial-peers forming part of the call.

*Syntax:*

```
dial-peer 1 config>FAX T38 MAX-SPEED <speed>
```

**2.4.3.20 [NO] FAX T38 REDUNDANCY**

Configures the number of packets, which are duplicated in a call using T38. With greater redundancy, greater robustness regarding errors in the network, however this also means more bandwidth is used. By default, redundancy is disabled (configured to zero). If various dial-peers, forming part of the call, have a different value configured, the highest value is selected.

*Syntax:*

```
dial-peer 1 config>FAX T38 REDUNDANCY <redundancy>
```

**2.4.3.21 [NO] FORWARD BUSY PEER-GROUP**

Specifies a peer group to forward a call to if the peer is busy. This creates jump groups. Please see section 2.7 Hunt Groups in this chapter and peer-groups for further information.

On configuring a **forward busy peer** group, a call leaving through this dial-peer whose destination is busy, tries to hop to the first unoccupied dial-peer, within the peer-group indicated in this parameter.

*Syntax:*

```
dial-peer 1 config>FORWARD BUSY PEER-GROUP <groupid>
```

*Example:*

```
dial-peer 1 voice-port
  destination-pattern 201
  forward busy peer-group 1
  target voice-port voip1/0 1
exit
;
dial-peer 2 voice-port
  destination-pattern 202
  forward busy peer-group 1
  target voice-port voip1/0 2
exit
;
dial-peer 3 voice-port
  destination-pattern 203
  forward busy peer-group 1
  target voice-port voip1/0 3
exit
;
peer-group 1
  dial-peer 2
  dial-peer 3
  dial-peer 1
exit
;
```

Using this configuration, if you call one of the three lines and it is busy, the call is diverted to the rest of the lines in the order indicated by the peer group (first number 2, then 3 and then 1). If all three lines are busy, the call is released.

### 2.4.3.22 [NO] FORWARD NO-ANSWER PEER-GROUP

A timer starts when the peer is called; where this is busy, or does not respond before the timer times out, this command specifies a peer group the call is forwarded to. To configure wait time, use **forward no-answer timeout**. Please see section 2.7 in this chapter on Hunt Groups and **peer-group** command for further information.

On configuring **forward no-answer** peer group, a call leaving this dial-peer, whose destination is busy or the call is not picked up within the configured time, is forwarded to the first available dial-peer from among the other dial-peers within the peer-group indicated in this parameter.

**Syntax:**

```
dial-peer 1 config>FORWARD NO-ANSWER PEER-GROUP <groupid>
```

**Example:**

```
dial-peer 1 voice-port
  destination-pattern 201
  forward no-answer peer-group 1
  forward no-answer timeout 15s
  target voice-port voip1/0 1
exit
;
dial-peer 2 voice-port
  destination-pattern 202
  forward no-answer peer-group 1
  forward no-answer timeout 15s
  target voice-port voip1/0 2
exit
;
dial-peer 3 voice-port
  destination-pattern 203
  forward no-answer peer-group 1
  forward no-answer timeout 15s
  target voice-port voip1/0 3
exit
;
peer-group 1
  dial-peer 2
  dial-peer 3
  dial-peer 1
exit
;
```

This configuration means if any of the three lines are called and they take more than 15 seconds to pick up, the call is forwarded to the rest of them in the order indicated by the peer group (first 2, then 3 and then 1). If all lines are busy, or the call is not picked up in the configured time, the call is released.

### 2.4.3.23 [NO] FORWARD NO-ANSWER TIMEOUT

Configures the time to be waited, from the moment the ring tone begins in the destination peer, before considering the call as unanswered. For further information, please see **forward no-answer** peer-group. Default is 20 seconds.

**Syntax:**

```
dial-peer 1 config>FORWARD NO-ANSWER TIMEOUT <interval>
```

### 2.4.3.24 [NO] FROM-REALM

Configures the realm used in SIP message FROM field, generated in calls, and registers in this dial-peer. If this is not configured, global realm is used configured in SIP Protocol. (Please see manual *bintec Dm766-I SIP*).

**Syntax:**

```
dial-peer 1 config>FROM-REALM <realm>
```

### 2.4.3.25 [NO] G729-ANNEXA

When negotiating capacities through H242, if the call has codec g729 permitted, g729annexa is sent as the first option, instead of sending g729, and g729annexa as the second. This command is only available for H323 dial-peers.

*Syntax:*

```
dial-peer 1 config>G729-ANNEXA
```

### 2.4.3.26 [NO] H245-TUNNELING ENABLE

Sends H.323 protocol H.245 messages, tunneled in Q931-H.225 messages, in calls using this dial-peer. This command is only available for H323 dial-peers.

*Syntax:*

```
dial-peer 1 config> H245-TUNNELING ENABLE
```

### 2.4.3.27 [NO] H245-TUNNELING DISABLE

Disables sending of H.323 protocol H.245 messages, tunneled in Q931-H.225 messages, in calls using this dial-peer. This command is only available for H323 dial-peers.

*Syntax:*

```
dial-peer 1 config> H245-TUNNELING DISABLE
```

### 2.4.3.28 [NO] H245-TUNNELING SYSTEM

Uses H.323 global configuration, referring to the tunneling of H.245 messages in Q931-H.225 messages, in calls using this dial-peer. This is default. This command is only available for H323 dial-peers.

*Syntax:*

```
dial-peer 1 config> H245-TUNNELING SYSTEM
```

### 2.4.3.29 [NO] H323-START FAST

Uses the fast-start method when executing a call through this dial-peer. This command is only available for H323 dial-peers.

*Syntax:*

```
dial-peer 1 config>H323-START FAST
```

### 2.4.3.30 [NO] H323-START SLOW

Does not use the fast-start method when executing a call through this dial-peer. This command is only available for H323 dial-peers.

*Syntax:*

```
dial-peer 1 config>H323-START SLOW
```

### 2.4.3.31 [NO] H323-START SYSTEM

Decides whether to use the fast-start method or not, depending on what has been configured in the H323 global configuration. This is default. This command is only available for H323 dial-peers.

*Syntax:*

```
dial-peer 1 config>H323-START SYSTEM
```

### 2.4.3.32 [NO] HUNTSTOP ALL

By default, if a call is routed by a dial-peer and an error is produced, this continues to process the list of dial-peers to see if one matches. If this occurs, the call is retried by the next matching dial-peer until one of them is successful.

If you want to avoid a failed call being routed by the next dial-peers, configure **huntstop all** in the dial-peer where you want the search to end.

*Syntax:*

```
dial-peer 1 config>HUNTSTOP ALL
```

Where you want to be more selective when detaining the search in the dial-peers, and stipulate the search stops due to a determined cause, execute one of the commands indicated below.

### 2.4.3.33 [NO] HUNTSTOP BUSY

Stops the search where the release cause returned is busy.

*Syntax:*

```
dial-peer 1 config>HUNTSTOP BUSY
```

### 2.4.3.34 [NO] HUNTSTOP INCOMPATIBLE

Stops the search where the release cause returned is incompatible destination.

*Syntax:*

```
dial-peer 1 config>HUNTSTOP INCOMPATIBLE
```

### 2.4.3.35 [NO] HUNTSTOP INCOMPLETE

Stops the search where the release cause returned is incomplete.

*Syntax:*

```
dial-peer 1 config>HUNTSTOP INCOMPLETE
```

### 2.4.3.36 [NO] HUNTSTOP NORMAL

Stops the search where the release cause returned is normal.

*Syntax:*

```
dial-peer 1 config>HUNTSTOP NORMAL
```

### 2.4.3.37 [NO] HUNTSTOP NO-ANSWER

Stops the search where the release cause returned is the user does not answer, but has been advised, i.e. the remote end has rung.

*Syntax:*

```
dial-peer 1 config>HUNTSTOP NO-ANSWER
```

### 2.4.3.38 [NO] HUNTSTOP NO-CIRCUIT

Stops the search where the release cause returned is there is no available channel circuit.

*Syntax:*

```
dial-peer 1 config>HUNTSTOP NO-CIRCUIT
```

### 2.4.3.39 [NO] HUNTSTOP NO-RESPONSE

Stops the search where the release cause returned is that the remote end does not respond and has not been advised.

*Syntax:*

```
dial-peer 1 config>HUNTSTOP NO-RESPONSE
```

### 2.4.3.40 [NO] HUNTSTOP OUT-OF-SERVICE

Stops the search where the release cause returned is network is out of service.

*Syntax:*

```
dial-peer 1 config>HUNTSTOP OUT-OF-SERVICE
```

### 2.4.3.41 [NO] HUNTSTOP REJECT

Stops the search where the release cause returned is call rejected.

*Syntax:*

```
dial-peer 1 config>HUNTSTOP REJECT
```

#### 2.4.3.42 [NO] HUNTSTOP TIMEOUT

Stops the search where call rejection is due to a timer timing out in the connection process.

*Syntax:*

```
dial-peer 1 config>HUNTSTOP TIMEOUT
```

#### 2.4.3.43 [NO] HUNTSTOP UNASSIGNED

Stops the search where the release cause returned is the requested number is not assigned.

*Syntax:*

```
dial-peer 1 config>HUNTSTOP UNASSIGNED
```

#### 2.4.3.44 [NO] HUNTSTOP UNSPECIFIED

Stops the search where the release cause returned is not specified (i.e. none of the above).

*Syntax:*

```
dial-peer 1 config>HUNTSTOP UNSPECIFIED
```

#### 2.4.3.45 [NO] INCOMING ACCESS-LIST

Configures an access list as filter so the dial-peer can match as incoming. If source IP for the call is permitted, the device checks if this dial-peer matches. If the IP is not permitted, the dial-peer cannot match as incoming. **no** eliminates the filter, so any call can match this dial-peer by following the normal search process. For further information, please see section 1 in this chapter, Dial-Peers.

*Syntax:*

```
dial-peer 1 config>INCOMING ACCESS-LIST <id>
```

#### 2.4.3.46 [NO] INCOMING CALLED NUMBER

Specifies a number pattern for called telephone numbers that match the dial-peer for incoming calls. For further information, please see section 1 in this chapter, Dial Peers.

*Syntax:*

```
dial-peer 1 config>INCOMING CALLED NUMBER <pattern>
```

#### 2.4.3.47 [NO] INCOMING CALLED TRANSLATION

Specifies a translation over the called number when the dial-peer is used as an incoming dial-peer.

*Syntax:*

```
dial-peer 1 config>INCOMING CALLED TRANSLATION <translation>
```

#### 2.4.3.48 [NO] INCOMING CALLED TYPE

Specifies the type of called number when the dial-peer is used as an incoming dial-peer.

*Syntax:*

```
dial-peer 1 config>INCOMING CALLED TYPE <1-7>
```

#### 2.4.3.49 [NO] INCOMING CALLING ID-ALLOW

This shows calling ID when the dial-peer with this matches incoming calls. This displays the calling ID even if the incoming call has the caller restricted.

*Syntax:*

```
dial-peer 1 config>INCOMING CALLING ID-ALLOW
```

### 2.4.3.50 [NO] INCOMING CALLING ID-RESTRICT

Allows calling identification to be hidden when the dial-peer with this matches in incoming calls.

*Syntax:*

```
dial-peer 1 config>INCOMING CALLING ID-RESTRICT
```

### 2.4.3.51 [NO] INCOMING CALLING NUMBER

Specifies a numbers pattern for calling telephone numbers, which match the dial-peer for incoming calls. For further information, please see section 1 in this chapter, Dial Peers.

*Syntax:*

```
dial-peer 1 config>INCOMING CALLING NUMBER <pattern>
```

### 2.4.3.52 [NO] INCOMING CALLING TRANSLATION

Specifies a translation over a calling number when the dial-peer is used as an incoming dial-peer.

*Syntax:*

```
dial-peer 1 config>INCOMING CALLING TRANSLATION <translation>
```

### 2.4.3.53 [NO] INCOMING CALLING TYPE

Type of dial-plan, which by default follows the calling and called number within the UIT-T Q.931 norm. This information appears in the Calling Address and Called Address information elements in Q.931 SETUP and INFO messages. When the call originates from a voice ISDN line, the dial-plan type is taken from voice ISDN Q.931 messages. Admits values between 0 and 7, (excepting 5), and takes effect when the dial-peer is used as an incoming dial-peer.

*Syntax:*

```
dial-peer 1 config>INCOMING CALLING TYPE <1-7>
```

### 2.4.3.54 [NO] INCOMING DIAL PLAN

Configures a peer-group to use as outgoing dial-peer list permitted in calls where the dial-peer matches as incoming. If this is not configured, the device tries to establish the call by searching among all existing dial-peers. For further information, please see section 1 in this chapter, Dial Peers.

*Syntax:*

```
dial-peer 1 config>INCOMING DIAL-PLAN PEER-GROUP <id>
```

### 2.4.3.55 [NO] INCOMING PREFIX

Specifies a prefix to add to the number called when the dial-peer is used as an incoming dial-peer. The prefix may include some dialing pauses when using the ',' digit.

*Syntax:*

```
dial-peer 1 config>INCOMING PREFIX <prefix>
```

### 2.4.3.56 [NO] INCOMING STRIP-DIGITS

Specifies a number of digits to be eliminated from the called number when the dial-peer is used as an incoming dial-peer

*Syntax:*

```
dial-peer 1 config>INCOMING STRIP-DIGITS <digits>
```

### 2.4.3.57 [NO] LOCAL-IP

Specifies a local IP address to be used in calls with this dial-peer. In an outgoing SIP dial-peer, the address configured in this command is used in the *Via Header* field of an SIP packet. Therefore, responses to said SIP packet are addressed to that address, regardless of the source address for the SIP packet being responded to.



**Syntax:**

```
dial-peer 1 config>LOCAL-IP <ip>
```

**Command history:**

Release	Modification
10.09.27	In SIP dial-peers, configures the address used in <i>Via Header</i> field.
11.00.05	In SIP dial-peers, configures the address used in <i>Via Header</i> field.
11.01.01	In SIP dial-peers, configures the address used in <i>Via Header</i> field.

**2.4.3.58 [NO] MULTICAST-IP**

Configures a multicast IP address as traffic source. This is only available in direct dial-peers.

**Syntax:**

```
dial-peer 1 config>MULTICAST <ip>
```

**2.4.3.59 [NO] OPTIONS-KEEPALIVE**

Activates the periodic dial-peer poll to determine availability at any point. The poll is executed by sending **options** SIP petitions. Optionally, you can specify the period between polls. This command is only available in SIP dial peers.

**Syntax**

```
dial-peer 1 config>OPTIONS-KEEPALIVE [PERIOD <period>]
```

**2.4.3.60 [NO] OUTGOING CALLED TRANSLATION**

Specifies a translation over the called number when the dial-peer is used as an outgoing dial-peer.

**Syntax:**

```
dial-peer 1 config>OUTGOING CALLED TRANSLATION <translation>
```

**2.4.3.61 [NO] OUTGOING CALLING ID-ALLOW**

This shows calling ID when the dial-peer with this matches the outgoing calls. This displays the calling ID, even if the received call has the caller restricted.

**Syntax:**

```
dial-peer 1 config>OUTGOING CALLING ID-ALLOW
```

**2.4.3.62 [NO] OUTGOING CALLING ID-RESTRICT**

Allows calling identification to be hidden when the dial-peer with this is used in outgoing calls.

**Syntax:**

```
dial-peer 1 config>OUTGOING CALLING ID_RESTRICT
```

**2.4.3.63 [NO] OUTGOING CALLING TRANSLATION**

Specifies a translation over the calling number when the dial-peer is used as an outgoing dial-peer.

**Syntax:**

```
dial-peer 1 config>OUTGOING CALLING TRANSLATION <translation>
```

**2.4.3.64 [NO] OUTGOING PREFIX**

Specifies a prefix to be added to the number called when the dial-peer is used as an outgoing dial-peer. The prefix may include some dialing pauses, when using the ',' digit.

**Syntax:**

```
dial-peer 1 config>OUTGOING PREFIX <prefix>
```

### 2.4.3.65 [NO] OUTGOING STRIP-DIGITS

Specifies a number of digits to eliminate from the called number when the dial-peer is used as an outgoing dial-peer.

*Syntax:*

```
dial-peer 1 config>OUTGOING STRIP-DIGITS <digits>
```

### 2.4.3.66 [NO] PASSWORD

The configuration of this command affects various situations:

*External server authentication*

Voice-port and group dial-peers, which have **destination-alias** or **destination-pattern** configured, without wildcards, register said extension in the configured external SIP server.

If the server requests authentication, the password configured through this command in the dial-peer, being registered, is used. If this command is not configured, the global password configured in the SIP protocol through **password client** from the protocol sip menu is used (Please see manual *bintec Dm766-I SIP*). If this is not configured either, it is not possible to respond correctly to an authentication request from the SIP server.

*Authentication of calls to the external server*

For calls, when asked to authenticate an INVITE or Re-INVITE, the device uses the first of the following as the password: the one configured with **password** in the dial-peer corresponding to the endpoint asking for authentication (usually outgoing), the one configured with the **password** command in the dial-peer opposite the endpoint that is asking for authentication (usually incoming), the global password configured in the SIP protocol using **password client** from the **sip protocol** menu (See manual *bintec Dm766-I SIP*).

*Request for credentials from SIP terminals such as the registrar server:*

If **local-registrar user-check** is configured in the sip protocol menu (see manual *bintec Dm766-I SIP*), upon receipt of registers in the device's registrar server, the extension being registered is checked to make sure that it is configured in one of the device's dial-peers. If that dial-peer has password configured, authentication is requested. If password isn't configured, no authentication is requested when receiving a SIP register for this dial-peer.

See also **auth-user**, which lets you specify a user for authentication.

*Syntax:*

```
dial-peer 1 config>PASSWORD <password>
```

### 2.4.3.67 [NO] PCM-LAW

This is used when the dial-peer is associated with a virtual ISDN VoIP interface connected to an ISDN line. Specifies the type of PCM codification used by the dial-peer, which uses the device located at the other end of the line.

*Syntax:*

```
dial-peer 1 config>pcm-law
a-law      G711 a-law
u-law      G711 u-law
```

### 2.4.3.68 [NO] REGISTER H323

This command is available in voice-port and group dial-peers, and indicates if the extension configured in this dial-peer should register (or not) in the H323 Gatekeeper if this is configured. The extension registered is that configured through **destination-alias** or, if this doesn't exist, through **destination-pattern**, provided that pattern does not contain '.' or 'T' wildcards. By default, the register is active and can be disabled through **no register h323**. You can globally enable and disable the register in H323 protocol.

*Syntax:*

```
dial-peer 1 config>REGISTER H323
```

### 2.4.3.69 [NO] REGISTER SIP

This command is available in voice-port and group dial-peers. This indicates if the extension, configured in this dial-peer, should register in the Proxy SIP (if this is configured). The extension registered is the one configured through **destination-alias**, or if this doesn't exist, through the **destination-pattern**, provided the pattern does not have either '.' or 'T' wildcards. The register is active by default. Said register can be deactivated through **no register sip**. You can globally enable and disable the register in the device in SIP protocol.

*Syntax:*

```
dial-peer 1 config>REGISTER SIP
```

### 2.4.3.70 [NO] REGISTER TIMER

Through this command, you can individually configure the extension register time for this dial-peer in the SIP Proxy. This command is available in voice-port and group dial-peers. By default, the time globally configured in the SIP protocol, through **timers register**, is used.

*Syntax:*

```
dial-peer 1 config>REGISTER TIMER <time>
```

### 2.4.3.71 [NO] REGISTRAR

This command is available in voice-port dial-peers and indicates the IP address and UDP port where the dial-peer must register when registration is carried out through the SIP protocol. This is used when you want to indicate a specific dial-peer registers in a different IP from the one configured in the SIP protocol. The port is optional, if nothing is indicated, the default value from the SIP protocol is used, 5060.

*Syntax:*

```
dial-peer 1 config>REGISTRAR <ip> [<port>]
```

### 2.4.3.72 [NO] RING-SIGNAL

Defines the ring pattern for an FXS line depending on the caller. For example, this can make the telephone connected to the FXS port ring in one way for internal calls and in a different way for calls from the public network.

Configures the ring pattern to use when the dial-peer with this is used as an incoming call dial-peer. The configured values are only significant when the outgoing call dial-peer is associated with an FXS interface.

If this is not configured, values globally configured through **ring-signal activity** and **ring-signal silence** in the telephony menu are used. **ring-signal activity** configures the time in tenths of a second during which the call signal (ring) is maintained active. Admits values between 1 and 25. Default is 15 tenths of a second. **ring-signal silence** configures the time in tenths of a second during which the call signal (ring) is maintained inactive. Admits values between 0 and 30. Default is 30 tenths of a second.

*Syntax:*

```
dial-peer 10 config>RING-SIGNAL ACTIVITY <time>
dial-peer 10 config>RING-SIGNAL SILENCE <time>
```

### 2.4.3.73 [NO] RTP AROUND

RTP packets for SIP-SIP calls associated with this dial-peer do not pass through the device. RTP flow is directly exchanged between call source and destination SIP terminals.

*Syntax:*

```
dial-peer 10 config$
dial-peer 10 config>RTP AROUND
```

### 2.4.3.74 [NO] RTP FLOW-MODE

Configures how the conversation audio is established with this dial-peer. This could be uni-directional or bi-directional. When declared uni-directional, the voice gateway does not send audio in the conversation, but does process that received. Default is bi-directional audio. This command is only available in direct dial-peers.

*Syntax:*

```
dial-peer 1config>RTP FLOW-MODE ?
receive-transmit RTP receive and transmit audio/events
```

```
receive-only      RTP only receives audio/events
dial-peer 10 config$
```

### 2.4.3.75 [NO] RTP KEEPALIVE

If this is configured, the router sends an RTP (keepalive packet) once the time configured without an RTP packet (in a call using this dial-peer) times out. This action indicates the call is still active. This command is useful when vad is active and the other end has a timeout configured through **rtp timeout**. Possible errors and disconnections in the call can be detected if no audio has been received for more than the determined period of time. By configuring **keepalive**, even if there is silence, a packet will always be sent to prevent timeout at the other end. Timeout will only activate if there is a real problem in the call. Default is keepalive deactivated.

*Syntax:*

```
dial-peer 1 config>RTP KEEPALIVE ?
<1s..1h>    Time value
dial-peer 1 config>RTP KEEPALIVE 5s
dial-peer 1 config>
```

### 2.4.3.76 [NO] RTP MIX-RX-AUDIO-SOURCES

Direct dial-peers allow you to mix various RTP audio sources provided audio is encoded in G.711 A-law. The resulting audio (with the addition of the various sources) is reproduced in the physical interface associated with the call. Said mixture process is also known as audio collection.

Command used to indicate the maximum number of simultaneous sources to mix. If the number of sources present is greater than that configured, they're ignored. Default is no audio collection. This admits values between 2 and 60 sources. This command is only available in direct dial-peers

*Syntax:*

```
dial-peer 1 config>RTP MIX-RX-AUDIO-SOURCES ?
<2..60>    Value in the specified range
dial-peer 1 config$
```

### 2.4.3.77 [NO] RTP PAYLOAD-TYPE CN

If this command is configured, the router tries to negotiate the payload for comfort noise packets (CN) from the silence detection (VAD) in SIP calls. This only makes sense for the G.711 codec as in G.723 and G.729, CN packets use the same payload as audio. If negotiation is successful and G.711 codec is used, VAD is activated and CN packets are sent with payload negotiated. If the other end doesn't negotiate CN payload and codec is G.711, VAD is deactivated in the call. By default, VAD is not negotiated. Whether this is active or not depends on the configuration executed through **vad** in the dial-peer menu; the payload used depends on rtp vad-rfc3389 telephony.

*Syntax:*

```
dial-peer 1 config>RTP PAYLOAD-TYPE CN <payload>
```

### 2.4.3.78 [NO] RTP PAYLOAD-TYPE RFC2833

Specifies RTP payload used to send dtmf digits through RTP through RFC 2833. The payload value can be between 97 and 127.

*Syntax:*

```
dial-peer 1 config>RTP PAYLOAD-TYPE RFC2833 <payload>
```

### 2.4.3.79 [NO] RTP PLAYOUT-DELAY MAXIMUM

Delay for RTP frames to be reproduced, adapts to the quality of the UDP/IP connection, specifically to the variance in the delay suffered by RTP packets. This command specifies the maximum delay at which RTP frames are reproduced. Default is 200 milliseconds.

*Syntax:*

```
dial-peer 1 config>RTP PLAYOUT-DELAY MAXIMUM <value>
```

### 2.4.3.80 [NO] RTP PLAYOUT-DELAY MINIMUM

Specifies the minimum delay to reproduce RTP frames. Default is 40 milliseconds. If you want a constant delay (not adaptive), simply configure the maximum and minimum to the same value. The reproduction delay will then be constant and equal to said value.

*Syntax:*

```
dial-peer 1 config>RTP PLAYOUT-DELAY MINIMUM <value>
```

### 2.4.3.81 [NO] RTP PORT-EXCLUSIVE

Configures the UDP port to be used in a call for RTP traffic so, when this dial-peer is used it tries to use said UDP port. If the configured port is being used, an error is produced and the call cannot be established. This admits values between 1024 and 65535. No UDP port is configured by default. To execute calls, the first free port is used starting from the UDP port configured through **rtp port** (explained in this manual). This command is only available in a direct dial-peer.

*Syntax:*

```
dial-peer 10 config>RTP PORT-EXCLUSIVE <port>
```

### 2.4.3.82 [NO] RTP RFC2833 SEND CNG

If a CNG caller fax tone is detected, notification is sent to the other end via a 36 digit message complying with RFC2833. This command only takes effect if RFC2833 has been negotiated in the call.

*Syntax:*

```
dial-peer 10 config>RTP RFC2833 SEND CNG
```

### 2.4.3.83 [NO] RTP RX-GAIN

Configures the gain in decibels to apply to RTP voice traffic received in this dial-peer. This is only applicable when codec is G711. Values can vary between -31 and 31 dB. Default is no gain configured (0dB).

*Syntax:*

```
dial-peer 10 config>RTP RX-GAIN ?
<-31..31>   Gain value in dB
dial-peer 10 config>RTP RX-GAIN 3
```

### 2.4.3.84 [NO] RTP THROUGH

This forces RTP packets in SIP-SIP calls, associated with this dial-peer, to pass through the device. In medium negotiation through SDP, the device substitutes the IPs associated with the calling RTP and the caller for its own IP so, RTP packets pass through this.

*Syntax:*

```
dial-peer 10 config>RTP THROUGH
```

### 2.4.3.85 [NO] RTP TIMEOUT [DROP-CALL / EVENT-INTERVAL]

If this command is configured, the router can detect when the time configured, without receiving an RTP packet in the call using this dial-peer, times out and generate an event reporting this. If it subsequently receives RTP once more, another event is displayed reporting recovery. If, in addition, the command has been configured with **drop-call**, after said timeout times out and the event is sent, the router drops the call.

Otherwise, if the command has been configured with **event-interval**, then the device repeats said event every certain period of time (preconfigured) while it isn't receiving RTP packets.

*Syntax:*

```
dial-peer 1 config>RTP TIMEOUT ?
<1s..1h>   Time value
dial-peer 1 config>RTP TIMEOUT 10s ?
drop-call   Drop the call after timeout is reached
event-interval Time between timeout events
<cr>
dial-peer 1 config>RTP TIMEOUT 10s DROP-CALL
```

```
dial-peer 1 config>RTP TIMEOUT 10s EVENT-INTERVAL ?
<1s..1h>    Time value
dial-peer 1 config>RTP TIMEOUT 10s EVENT-INTERVAL 5m
dial-peer 1 config>
```

### 2.4.3.86 [NO] RTP TX-GAIN

Configures the gain in decibels to apply to RTP voice traffic transmitted through this dial-peer. This is only applicable when codec is G711. The values can vary between -31 and 31 dB. Default is no gain configured (0dB).

#### Syntax:

```
dial-peer 10 config>RTP TX-GAIN ?
<-31..31>    Gain value in dB
dial-peer 10 config>RTP TX-GAIN 3
```

### 2.4.3.87 [NO] SRTP AUTH

Specifies a specific SRTP authentication method for calls using the dial-peer.

#### Syntax:

```
dial-peer 10 config>SRTP AUTH ?
none          No srtp authentication
hmac-sha1-32  Use HMAC SHA1 32 bits authentication
hmac-sha1-80  Use HMAC SHA1 80 bits authentication
system       Use global srtp configuration
dial-peer 10 config>
```

The options are **none** to deactivate authentication, **hmac-sha1-32** to use 32 bits, **hmac-sha1-80** to use 80 bits and **system** to use the authentication configured in the global telephony menu. This command is only available for SIP dial-peers. Default is **system**.

### 2.4.3.88 [NO] SRTP MODE

Specifies a specific SRTP negotiation method for calls using the dial-peer.

#### Syntax:

```
dial-peer 10 config>SRTP MODE ?
none          Do not use srtp
fallback      Use srtp when possible
force         Only use srtp
system       Use global srtp configuration
dial-peer 10 config>
```

Options are **none** to deactivate SRTP; **fallback** to negotiate SRTP and, if you can't not encrypt the call, **force** to compel the use of SRTP; **system** to use the mode configured in the global telephony menu. This command is only available for SIP dial-peers. Default is **system**.

### 2.4.3.89 [NO] SRTP RTCP

Specifies whether RTCP stream is encrypted or not in calls using the dial-peer.

#### Syntax:

```
dial-peer 10 config>SRTP RTCP ?
encrypted     Encrypt rtcp packets for this dial-peer
unencrypted   Do not encrypt rtcp packets for this dial-peer
system       Use global srtp configuration
dial-peer 10 config>
```

Options are **encrypted** to encrypt RTCP; **unencrypted** for non-encrypt; **system** to encrypt or not, depending on the configuration in the telephony global menu. This command is only available for SIP dial-peers. Default is **system**.

### 2.4.3.90 [NO] SDP HANDLE

For SIP-SIP calls associated with this dial-peer, the device processes SDP protocol embedded in SIP messages, to adjust negotiated parameters according to this configuration. Consequently, codec in SIP-SIP call can be limited.

If this mode is forced, only basic calls can operate, i.e. transfers cannot be carried out.

**Syntax:**

```
dial-peer 10 config>SDP HANDLE
```

**2.4.3.91 [NO] SDP TRASPARENT**

The device does not process SDP messages in SIP-SIP calls, it transparently resends it from one extreme to another. This is default.

**Syntax:**

```
dial-peer 10 config>SDP TRASPARENT
```

**2.4.3.92 [NO] SHUTDOWN**

Disables a dial-peer so it is not used in either incoming or outgoing calls.

**Syntax:**

```
dial-peer 1 config>SHUTDOWN
```

**2.4.3.93 [NO] SIP-CONTACT**

This command is only available for SIP dial-peers and configures the CONTACT field used on SIP messages with format: <user>@<host\_ip>.

If this command is not configured, the CONTACT field will include the calling number as <user> and the IP configured in the SIP Protocol as the <host\_ip>. (Please see manual *bintec Dm766-I SIP*).

**Syntax:**

```
dial-peer 2 config>sip-contact ?
<word>
```

**Example:**

```
dial-peer 2 config>sip-contact 100@192.168.1.1
```

**Command history:**

Release	Modification
11.01.03	This command was introduced as of version 11.01.03.

**2.4.3.94 [NO] SIP-HOLD**

Configures placing a SIP call matching a dial-peer on hold.

**Syntax:**

```
dial-peer 20 config>sip-hold ?
inactive    Use inactive to hold SIP calls
rfc2543     Use RFC2543 to hold SIP calls
rfc3261     Use RFC3261 to hold SIP calls
```

Options are: **inactive**, used by the inactive attribute in SDP; **rfc2543**, which indicates 0.0.0.0 as the IP to receive audio in SDP (recommended in SIP version described in RFC2543); **rfc3261** uses *sendonly* SDP attribute (recommended in the latest SIP version, RFC3261 and RFC3264). This command is only available in SIP dial-peers. Default is not configured. The global value configured through **hold** (from SIP protocol) is used. (Please see manual *bintec Dm766-I SIP*).

**2.4.3.95 [NO] SIP-URI PHONE-CONTEXT**

When this dial-peer is used in a SIP call, the value configured is added to the end of the SIP URI user field, preceded by the “;phone-context=” text. E.g., if the dial-peer corresponds to the SIP URI <sip:100@192.168.1.1> and the *oficina1* value, the new value of the SIP URI is <sip:100;phone-context=oficina1@192.168.1.1> . If the dial-peer matches as incoming, the URI affected is the one corresponding to the caller. However, if this is configured in an outgoing dial-peer, phone-context is added to the call destination URI. Default is not configured i.e. field not included.

**Syntax:**

```
dial-peer 1 config>sip-uri phone-context <text>
```

### 2.4.3.96 [NO] SIP-URI USER-PHONE

When this dial-peer is used in a SIP call, the `user=phone` parameter is added to the SIP URI. This parameter indicates SIP URI is a telephone number. E.g. if the dial-peer corresponds to SIP URI `<sip:100@192.168.1.1>`, the new SIP URI value is `<sip:100 @192.168.1.1;user=phone>`. If the dial-peer matches as incoming, the URI affected is the one corresponding to the caller. However, if this command is configured in the outgoing dial-peer, the parameter is added to the call destination URI. Default is not configured: this field isn't included.

*Syntax:*

```
dial-peer 1 config>sip-uri user-phone
```

### 2.4.3.97 [NO] STUN

Enables or disables the use of STUN for calls where this dial-peer intervenes.

For STUN to work, it must be active in both the dial-peer and SIP/H323 protocol global configuration. Enabling STUN is incompatible with the `dial-peer local-ip` command. If you enable both, the calls won't be correctly forwarded in said dial-peer.

*Syntax:*

```
dial-peer 1 config>STUN
```

### 2.4.3.98 [NO] TARGET DIRECTED-PICKUP PREFIX

If a call finds a facility outgoing dial-peer with target directed-pickup, this executes directed pickup. For this, configure a pattern through **destination-pattern**, which consists of a pickup prefix plus the extension you wish to pickup. To differentiate the extension prefix, configure the prefix length through `<length>` parameter for said command. The extension to be picked up is obtained by applying the configured number changes and eliminating pickup prefix from the called number. With the resulting number, you can discover the number you want to pick up. If this rings, it picks up, if not, an error is sent. For further information, please see section 2.5 Call Capture. This command is only available in facility dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET DIRECTED-PICKUP PREFIX <length>
```

### 2.4.3.99 [NO] TARGET DNS

Indicates destination associated to this dial-peer is a domain name. The device first tries to resolve the domain name and subsequently routes the call. This command is only available for SIP dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET DNS <dns-name>
```

### 2.4.3.100 [NO] TARGET DYNAMIC

Indicates the destination associated with this dial-peer is searched for among the dynamic dial-peers created when an SIP, H323, NOE or SCCP terminal was registered. If a dynamic dial-peer is found with the same number, the call is routed. If not, it fails. This command is only available for SIP, H323, NOE or SCCP dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET DYNAMIC
```

### 2.4.3.101 [NO] TARGET FORWARD-ALWAYS PREFIX

Activates unconditional forward available in FXS interfaces. If the call origin is different to an FXS terminal, the call fails. The extension the call is forwarded to is obtained by applying the configured number changes and eliminating the forward prefix (length is configured through the `<length>` parameter). So forwarding is admitted, there must be an outgoing dial-peer the call can go through to this number. If successful, you'll hear two confirmation tones in the telephone; contrariwise, you'll hear the error tone. For further information, please see section 2.6 Call Forwarding. This command is only available for facility dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET FORWARD-ALWAYS PREFIX <length>
```



### 2.4.3.102 [NO] TARGET FORWARD-ALWAYS-CLEAR

Deactivates unconditional forward in FXS interfaces. If call origin is different to an FXS terminal, the call fails. You hear two confirmation tones in the telephone. For further information, please see section 2.6 Call Forwarding. This command is only available in facility dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET FORWARD-ALWAYS-CLEAR
```

### 2.4.3.103 [NO] TARGET FORWARD-CLEAR

Deactivates all forwarding in FXS interfaces. If the call origin is different to an FXS terminal, the call fails. You hear two confirmation tones in the telephone. For further information, please see section 2.6 Call Forwarding. This command is only available for facility dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET FORWARD-CLEAR
```

### 2.4.3.104 [NO] TARGET FORWARD-NOANSWER PREFIX

Activates **forward** if no answer, available in FXS interfaces. If call origin is different to an FXS terminal, said call fails. The extension the call is forwarded to is obtained by applying the configured number changes and eliminating the forward prefix (length is configured through the **<length>** parameter). So forwarding is admitted, there must be an outgoing dial-peer through which the call can exit to this number. If successful, you hear two confirmation tones in the telephone, contrariwise, you'll hear the error tone. For further information, please see section 2.6 Call Forwarding.

The time taken in forwarding the call is configured per line through the voip interface command: **line <id> timing forward-no-answer <time>** (please see *bintec Dm770-I VoIP Interfaces*). This command is only available for facility dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET FORWARD-NOANSWER PREFIX <length>
```

### 2.4.3.105 [NO] TARGET FORWARD-NOANSWER-CLEAR

Deactivates forward if no answer is available in FXS interfaces. If the call origin is different to an FXS terminal, the call fails. You hear two confirmation tones in the telephone. For further information on this, please see section 2.6 Call Forwarding. This command is only available in facility dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET FORWARD-NOANSWER-CLEAR
```

### 2.4.3.106 [NO] TARGET GATEKEEPER

Indicates the call destination is the gatekeeper configured in the device. This command is only available for H323 dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET GATEKEEPER
```

### 2.4.3.107 [NO] TARGET GROUP

Indicates the destination associated with this dial-peer is a group of dial-peers. The call is simultaneously directed to all dial-peers within this group of dial-peers. The call establishes with the first that picks up. For further information, please see section 2.3 Simultaneous ringing of various telephones. This command is only available for group dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET GROUP <groupid>
```

### 2.4.3.108 [NO] TARGET GROUP-PICKUP PEER-GROUP

Action to take on a call matching an outgoing facility dial-peer with target group-pickup, is to capture the ringing call. This latter has an outgoing dial-peer pertaining to the configured peer-group. If there is a configured telephone associated with the dial-peer in the peer-group, it's important that not only is there a call ringing for said dial-peer, but also the called number must be the one configured in the peer-group. If **dynamic-peers** is configured in the peer-group, it's possible to capture any ringing dial-peer. Please see further information in section 2.4 Group Pickup. This command is only available in facility dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET GROUP-PICKUP PEER-GROUP <groupid>
```

### 2.4.3.109 [NO] TARGET IPV4

Indicates the call destination is the host specified by the IP configured through this command. This command is only available for H323 or SIP dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET IPV4 <ip>
```

### 2.4.3.110 [NO] TARGET LOC-RECORD

Configuring this **target** in a facility dial-peer allows you to record voice messages in device proprietor format. It's only possible to record voice messages from FXS interfaces.

On dialing the configured number in said dial-peer destination-pattern from an FXS interface, two confirmation tones are heard and the audio begins to be recorded. To finalize this, press a number on the telephone, which will be the number for the recorded message. The voice message is stored in the device flash with the name LOC <num>.loc and in the codec configured in the dial-peer facility. Subsequently, you hear two confirmation tones and the recorded voice message is then heard. The maximum length of the recorded voice messages is 30 seconds. This command is only available in facility dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET LOC-RECORD
```

### 2.4.3.111 [NO] TARGET SIP-PROXY

Indicates call destination is the SIP proxy configured in the device. If there is no proxy, or it's inactive, the dial-peer passes to inactive. This command is only available for SIP dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET SIP-PROXY
```

### 2.4.3.112 [NO] TARGET MAC

This command is used in IP telephony protocols where the telephone does not know its telephone number to assign it from the router. This applies to NOE and SCCP terminals. Consequently, this command is only available in noe and sccp dial-peers.

When a NOE terminal connects to the router and the NOE protocol *server* application is enabled, a search is executed among noe dial-peers to see if one has the terminal mac configured through **target mac**. If the mac is found, the telephone is assigned the extension number from the first **destination-pattern** configured in this same dial-peer. If you use NOE *survival* application, the router is capable of discovering the telephone extension, i.e. this command does not apply. For further information, please see the manual on the NOE protocol (*bintec Dm777-I NOE*).

When an SCCP terminal connects to the router, a search is executed among SCCP dial-peers to see if one has the terminal mac configured through **target mac**. If the mac is found, the telephone is assigned an extension number for each **destination-pattern** command, which doesn't have wildcards in the pattern in this dial-peer. If the telephone has been connected to the router as backup server, it's possible to discover the extension it had with the main server: if so this command doesn't need to be used. For further information, please see the manual on the SCCP protocol (*bintec Dm782-I SCCP*).

*Syntax:*

```
dial-peer 1 config>TARGET MAC <m>
```

### 2.4.3.113 [NO] TARGET VOICE-PORT

Specifies the destination, associated with this dial-peer, is a VoIP interface telephone line. This command is only available for voice-port dial-peers.

*Syntax:*

```
dial-peer 1 config>TARGET VOICE-PORT <interface> <line>
```

### 2.4.3.114 [NO] TO-REALM

Configures the realm used in the SIP messages TO field, generated in calls, and registers in this dial-peer. If this is not configured, the global realm used is that configured in SIP Protocol. (Please see manual *bintec Dm766-I SIP*).

*Syntax:*

```
dial-peer 1 config>TO-REALM <realm>
```

### 2.4.3.115 [NO] TRACK

This option activates/deactivates the dial-peer, depending on various states (explained further on). If you configure more than one type of **track** in the same dial-peer, the dial-peer activates as soon as one of them registers as active. Dial-peer status is a logical OR resulting from each individual TRACK.

### 2.4.3.116 [NO] TRACK IP-CALLS

If this option is configured the dial-peer is only active if the maximum IP calls limit (configured through **call threshold** from the telephony menu) has been reached. Otherwise, dial-peer is inactive.

*Syntax:*

```
dial-peer 1 config>TRACK IP-CALLS
```

### 2.4.3.117 [NO] TRACK NSLA

If this option is configured, the dial-peer is only activates if the nsla advisor (associated through this command) is active.

*Syntax:*

```
dial-peer 1 config>TRACK NSLA-ADVISOR <advisor-id>
```

### 2.4.3.118 [NO] TRACK SIP-REGISTRATIONS

If this option is configured, the dial-peer activates when SIP registrations, sent to the proxy, do not receive a response.

*Syntax:*

```
dial-peer 1 config>TRACK SIP-REGISTRATIONS
```

### 2.4.3.119 [NO] TRANSPORT SYSTEM

Transport, used for outgoing calls pertaining to this dial-peer, is that configured in the SIP global configuration. This is default. This command is only available in SIP dial-peers.

*Syntax:*

```
dial-peer 1 config>TRANSPORT SYSTEM
```

### 2.4.3.120 [NO] TRANSPORT TCP

Transport used for the outgoing calls pertaining to this dial-peer is TCP. This command is only available in SIP dial-peers.

*Syntax:*

```
dial-peer 1 config>TRANSPORT TCP
```

### 2.4.3.121 [NO] TRANSPORT TLS

Transport used for the outgoing calls pertaining to this dial-peer is TLS. This command is only available in SIP dial-peers. SIP must be configured with a user certificate so TLS can be used (please see manual *bintec Dm766-I SIP*).

*Syntax:*

```
dial-peer 1 config>TRANSPORT TLS
```

### 2.4.3.122 [NO] TRANSPORT UDP

Transport used for outgoing calls pertaining to this dial-peer is UDP. This command is only available in SIP dial-peers.

*Syntax:*

```
dial-peer 1 config>TRANSPORT UDP
```

### 2.4.3.123 [NO] UPDATE LEVEL-INDICATOR

Updates the level indicator <id> in the configured value <val>. This indicator increases the value <val> for each established call (that has this dial peer as incoming or outgoing), and decreases said value when the call is released. The level indicator is used as an entry for the NSLA feature (please see manual *bintec Dm754-I NSLA*). This updates the indicator for each call, **per-call**, or only for calls considered external to the device local network (**per-external-call**). In the case of the latter, configure which ones are local networks (call application **local-networks access-list** from the telephony menu).

SIP dial-peers have an additional <type> (**when-down**). When this type is used, the indicator increases <val> if the **options-keepalive** command is configured and the peer is considered unavailable.

*Syntax:*

```
dial-peer 1 config>UPDATE LEVEL-INDICATOR <id> VALUE <val> <type>
```

**Command history:**

Release	Modification
11.01.09	The <b>when-down</b> type was added for SIP dial-peers as of version 11.01.09.

### 2.4.3.124 [NO] VAD

Enables silence detection (Voice Activity Detector) for gateway calls that use this dial-peer. **no vad** disables this feature. Default is enabled.

*Syntax:*

```
dial-peer 1 config>VAD
```

### 2.4.3.125 [NO] VOICE-CLASS

Configures previously defined codecs class for this dial-peer. If you configure codecs class, the **codec** command for said dial-peer becomes ineffective as **voice-class** takes precedence. (Please see **voice-class** on the telephony menu.)

*Syntax:*

```
dial-peer 1 config>VOICE-CLASS <classid>
```

## 2.4.4 LIST

Lists the different configured parameters.

*Syntax:*

```
Telephony Config>LIST ?
parameters      Display the parameters associated to the telephone numbers
```

### 2.4.4.1 LIST PARAMETERS

Displays parameters associated with the telephones. When the motherboard is ISDN telephony, certain parameters are not necessary and consequently do not appear on the list.

**Example:**

```

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: 4
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
Telephony Config>

```

**Command history:**

Release	Modification
10.09.26	Parameters on the 5 FXO busy patterns are displayed.
11.00.05	Parameters on the 5 FXO busy patterns are displayed.
11.01.00	Parameters on the 5 FXO busy patterns are displayed.

## 2.4.5 [NO] TELEPHONY

Deletes all the configuration from the telephony menu.

**Syntax:**

```
Telephony Config>NO TELEPHONY          Clear all telephony configuration
```

## 2.4.6 [NO] PEER-GROUP

Defines groups of dial-peers. This is useful if you want various telephones to ring at the same time when calling a given number (using it as a group dial-peer target), or to define capture groups using it as target group-pickup peer-group in a facility dial-peer. These can also be used to define outgoing dial-peers permitted in a call (through **call application dial-plan**, telephony menu), and the **incoming dial-plan** (in the dial-peer menu). This allows you to have different dial-plans depending on the source of a call. Another possible use is to configure a dial-peer forward group, using **forward busy peer-group** if it's busy, and **forward no-answer peer-group** if this is due to no answer. For further information and peer-group application examples, please see section 2.

On executing this command, a peer-groups configuration menu appears. You can eliminate a peer-group from said configuration through **no peer-group <id>**.

**Syntax:**

```
Telephony Config>PEER-GROUP <1..10000>  Value in the specified range
```

### 2.4.6.1 [NO] DIAL-PEER

Adds or eliminates a dial-peer to a peer-group.

**Syntax:**

```
dial-peer group 1 config>DIAL-PEER <id> <tlph>
```

*id*: identifier for the dial-peer you want to add to the group, between 1 and 10000.

*tlph*: this is an optional parameter with different meanings depending on what you are using the peer-group for. It is the telephone number used to call this peer if you are using the peer-group as a target group dial-peer. If no number is indicated, the original called telephone number is used. If this is used in **forward busy** or **forward no-answer** for a dial-peer, it's the telephone number called in call forwarding. When using this in group pickup (through **target group-pickup**), this checks it matches the number called to validate said pickup. For other uses, this field is ignored.

**Example 1:**

On calling telephone number 200, you want lines 1 and 2 from the first card, to ring simultaneously.

To do this, define two dial-peers, one for each line and a peer-group to group them. Subsequently, define another dial-peer with destination-pattern 200, which has the peer-group as target.

```
dial-peer 1 voice-port
  destination-pattern 201
  target voice-port voip1/0 1
exit
;
dial-peer 2 voice-port
  destination-pattern 202
  target voice-port voip1/0 2
exit
;
dial-peer 3 group
  destination-pattern 200
  target group 1
exit
```

```

;
peer-group 1
  dial-peer 1
  dial-peer 2
exit
;

```

With this configuration, on calling number 201, line 1 rings, on calling 202, line 2 and on calling 200, both lines ring.

### 2.4.6.2 DYNAMIC-PEERS

Adds or eliminates dynamic dial peers from a peer-group, which are created through registers or SIP, H323, NOE or SCCP terminals. This is applicable when the peer-group is used to define a dial-plan and indicates if the device should look for the outgoing dial-peer from among the registered telephones. This command is also applied when the peer-group is used as a target group dial-peer. In this case, all dynamically created terminals ring at the same time.

Syntax:

```
dial-peer group 1 config>DYNAMIC-PEERS
```

*Example 2:*

We don't want any call to establish with SIP telephones registered in the device, except for those from a server in IP 10.0.0.1 (which can only establish calls with said telephones). Configure a peer-group, which includes all dial-peers except dynamic ones, and assign the default dial-plan to it (**call application dial-plan default**). Another peer-group is created containing the dynamic dial-peers only, which is assigned to incoming calls from IP 10.0.0.1, using a dial-peer matching incoming calls filtered by the access list. The configuration is as follows:

```

; Configuración de feature access-list
access-list 10
  entry 1 default
  entry 1 permit
  entry 1 source address 10.0.0.1 255.255.255.25
exit
;

; Configuración de tlphy
dial-peer 1 voice-port
  destination-pattern 201
  target voice-port voip1/0 1
exit
;
dial-peer 2 voice-port
  destination-pattern 202
  target voice-port voip1/0 2
exit
;
dial-peer 3 group
  destination-pattern 200
  target group 1
exit
;
dial-peer 4 sip
  destination-pattern T
  target sip-proxy
exit
;
dial-peer 5 sip
  incoming access-list 10
  incoming called number T
  incoming dial-plan peer-group 1
exit
;
peer-group 1
  dynamic-peers
exit
;
peer-group 2
  dial-peer 1

```

```

dial-peer 2
dial-peer 3
dial-peer 4
exit
;
call application dial-plan default peer-group 2

```

**Example 3:**

You want calls to 2000 to ring in all telephones dynamically registered in the device, and calls to 3000 ring in three of them. These are SIP telephones with numbers 11, 12 and 13:

```

dial-peer 1 sip
  destination-pattern ..
  target dynamic
exit
;
dial-peer 2 group
  destination-pattern 2000
  target group 1
exit
;
dial-peer 3 group
  destination-pattern 3000
  target group 2
exit
;
peer-group 1
  dynamic-peers
exit
;
peer-group 2
  dial-peer 1 11
  dial-peer 1 12
  dial-peer 1 13
exit
;

```

## 2.4.7 RING-SIGNAL

Configures ring signal parameters. These ring on an analog telephone connected to an FXS port on the device.

**Syntax:**

```

Telephony Config>RING-SIGNAL ?
ACTIVITY      Time, in tenths of seconds, during which the tone is active
SILENCE       Tenths of seconds, during which the tone is inactive

```

### 2.4.7.1 RING-SIGNAL ACTIVITY

Configures the time in tenths of seconds during which the call signal is maintained activate. This command is only applied to the device FXS interfaces. Admits values between 1 and 100. Default is 15 tenths of a second.

**Example:**

```

Telephony Config>RING-SIGNAL ACTIVITY 20
Telephony Config>

```

If you give a value outside the permitted range, an error message is displayed.

### 2.4.7.2 RING-SIGNAL SILENCE

Configures the time in tenths of seconds during which the call signal is maintained silent. This command is only applied to the device FXS interfaces. Admits values between 0 and 100. Default is 30 tenths of a second.

**Example:**

```

Telephony Config>RING-SIGNAL SILENCE 20
Telephony Config>

```



If you give a value outside the permitted range, an error message is displayed.

## 2.4.8 [NO] RTP

Configures parameters relative to RTP, for voice over IP transmission, in calls that behave as Media Gateway.

*Syntax:*

```
Telephony Config>RTP ?
flow-through      RTP passes through the media gateway
ip-tos            Configure IP TOS for RTP and RTCP packets
port              Configure rtp starting port
rtcp-onhold       Send RTCP on held calls
vad-rfc3389       Use 13 payload for CN in G.711
```

### 2.4.8.1 [NO] RTP FLOW-THROUGH

Forces RTP packets to pass through the device in SIP-SIP calls. This is achieved by substituting the IP associated with RTP in SDP messages. If the dial-peers associated with a call have RTP THROUGH or RTP AROUND configured, then this configuration takes precedence over the global configuration parameter.

*Syntax:*

```
Telephony Config>RTP FLOW-THROUGH
```

### 2.4.8.2 [NO] RTP IP-TOS

Configures TOS in RTP packets sent by the device. You configure complete TOS byte through its value in hexadecimal. Default is 0.

*Syntax:*

```
Telephony Config>RTP IP-TOS ?
<hex 0x0..0xff>   Hexadecimal value in the specified range
```

### 2.4.8.3 [NO] RTP PORT

Configures the lowest UDP port used by RTP. From this base value, each call established in H323 occupies 4 UDP ports and each call established in SIP, 2 ports. **no rtp port** restores base RTP port configuration (used in calls over IP) to default (16384).

*Syntax:*

```
Telephony Config>RTP PORT <base-port>
```

### 2.4.8.4 [NO] RTP RTCP-ONHOLD

If configured, the device continues to send RTCP packets in onhold calls. Default is deactivated.

*Syntax:*

```
Telephony Config >RTP RTCP-ONHOLD
```

### 2.4.8.5 [NO] RTP VAD-RF3389

Affects the sending of SID frames (Silence Descriptors), which generate comfort noise when VAD (Voice Activity Detection) is enabled. This occurs when the audio codec used is G.711. On enabling SID frames, they are sent with RTP 13 payload, instead of 0 or 8, corresponding to G.711. If this is disabled, SID frames are sent with the same payload as voice frames. Regardless of how this command is configured, the device accepts reception of both audio payload (0 or 8) and that specified from SID (13).

*Syntax:*

```
Telephony Config>RTP VAD-RFC3389
```

## 2.4.9 [NO] SRTP

Configures parameters relative to *Secure RTP* used to encrypt RTP and RTCP flows. This feature is only available in calls using SIP signaling. The device implements RFC 4568 *Session Description Protocol (SDP) Security Descriptions for Media Streams* to negotiate keys used in RTP encryption. This is a generic configuration for all calls, although you can execute a specific configuration for a dial-peer, which takes preference.

*Syntax:*

```
Telephony Config>SRTP ?
  auth          srtplib authentication configuration
  mode          Secure rtp behavior
  unencrypted-rtcp Do not encrypt RTCP packets
```

### 2.4.9.1 [NO] SRTP AUTH

Specifies the authentication method for SRTP sessions.

*Syntax:*

```
Telephony Config>SRTP AUTH ?
  none          No srtplib authentication
  hmac-sha1-32  Use HMAC SHA1 32 bits authentication
  hmac-sha1-80  Use HMAC SHA1 80 bits authentication
```

The options are **none** to deactivate the authentication, **hmac-sha1-32** to use 32 bits and **hmac-sha1-80** to use 80 bits. Default is **hmac-sha1-80**.

### 2.4.9.2 [NO] SRTP MODE

Configures how SRTP is used in the calls.

*Syntax:*

```
Telephony Config>SRTP MODE ?
  none          Do not use srtplib
  fallback      Use srtplib when possible
  force         Only use srtplib
```

Options are **none** to deactivate SRTP; **fallback** to try and negotiate SRTP (and if you can't manage not to encrypt the call) and **force** to compel the use of SRTP.

### 2.4.9.3 [NO] SRTP UNENCRYPTED-RTCP

If you configure this option, the RTCP stream is not encrypted.

*Syntax:*

```
Telephony Config>SRTP UNENCRYPTED-RTCP
```

## 2.4.10 TONES

Configures parameters relative to the various tones heard in analog voice interfaces: **dial**, **alert**, **busy** and **error**. You can configure repetitions for each tone or voice messages to substitute the tone through the **voice-message** subcommand. Voice messages played are either recorded from an FXS interface in proprietary format (where the file name is the configured number and .LOC extension) or .wave (loaded through **voice-message** in the telephony menu). Voice messages can be independently configured for each device voice port through **<id> <tone> voice-message <loc>** for the device voip interfaces (*bintec Dm770-I VoIP Interfaces*). You also need to configure analog interface timers and hook-on detection in FXO interfaces.

*Syntax:*

```
Telephony Config>TONES ?
ALERT          Configures the parameters relative to the alert tone
BUSY           Configures the parameters relative to the busy tone
DIAL           Configures the parameters relative to the dialing tone
DTMF-TIMEOUT  Time from having received a DTMF until dialing completes
ERROR         Configures the parameters relative to the error tone
FXO-BUSY-PATTERN Configures FXO busy tone pattern to hook on
FXO-SUPERVISION-PATTERN Configured FXO dial tone for supervision tasks
```

### 2.4.10.1 [NO] TONES ALERT

Configures parameters relative to tone alert.

*Syntax:*

```
Telephony Config>TONES ALERT ?
ACTIVITY          Time, in tenths of seconds, during which the tone is active
FREQUENCY         Tone frequency
SILENCE           Tenths of seconds, during which the tone is inactive
VOICE-MESSAGE     Permits substituting the tone for voice message
```

#### 2.4.10.1.1 [NO] TONES ALERT ACTIVITY

Configures the time, in tenths of seconds, during which tone alert is maintained. Admits values between 1 and 100. Default is 15 tenths of a second.

*Example:*

```
Telephony Config>TONES ALERT ACTIVITY 20
Telephony Config>
```

#### 2.4.10.1.2 [NO] TONES ALERT FREQUENCY

Configures tone alert frequency in the remote terminal. Admits values between 0 and 4000 Hz. Default is 425 Hz.

*Example:*

```
Telephony Config>TONES ALERT FREQUENCY 450
Telephony Config>
```

#### 2.4.10.1.3 [NO] TONES ALERT SILENCE

Configures the time, in tenths of seconds, during which the tone alert is maintained inactive. Admits values between 1 and 100. Default is 30 tenths of a second.

*Example:*

```
Telephony Config>TONES ALERT SILENCE 40
Telephony Config>
```

#### 2.4.10.1.4 [NO] TONES ALERT VOICE-MESSAGE

Substitutes a dialing tone for a voice message. Admits values between 1 and 9. Default is no voice message.

*Example:*

```
Telephony Config>TONES ALERT VOICE-MESSAGE 1
Telephony Config>
```

### 2.4.10.2 [NO] TONES BUSY

Configures parameters relative to the busy tone.

*Syntax:*

```
Telephony Config>TONES BUSY ?
ACTIVITY          Time, in tenths of seconds, during which the tone is active
FREQUENCY         Tone frequency
SILENCE           Tenths of seconds, during which the tone is inactive
VOICE-MESSAGE     Permits substituting the tone for voice message
```

#### 2.4.10.2.1 [NO] TONES BUSY ACTIVITY

Configures the time, in tenths of seconds, during which the busy tone is maintained active. Admits values between 1 and 100. Default is 2 tenths of a second.

*Example:*

```
Telephony Config>TONES BUSY ACTIVITY 3
Telephony Config>
```

#### 2.4.10.2.2 [NO] TONES BUSY FREQUENCY

Configures tone frequency in hertz, which indicates the remote terminal is busy. Admits values between 0 and 4000 Hz. Default is 425 Hz.

*Example:*

```
Telephony Config>TONES BUSY FREQUENCY 450
Telephony Config>
```

#### 2.4.10.2.3 [NO] TONES BUSY SILENCE

Configures the time, in tenths of seconds, during which the busy tone is maintained inactive. Admits values between 1 and 100. Default is 2 tenths of a second.

*Example:*

```
Telephony Config>TONES BUSY SILENCE 3
Telephony Config>
```

#### 2.4.10.2.4 [NO] TONES BUSY VOICE-MESSAGE

Substitutes a busy tone for a voice message. Admits values between 1 and 9. Default is no voice message.

*Example:*

```
Telephony Config>TONES BUSY VOICE-MESSAGE 1
Telephony Config>
```

### 2.4.10.3 [NO] TONES DIAL

Configures parameters relative to the dial tone.

*Syntax:*

```
Telephony Config>TONES DIAL ?
FREQUENCY      Tone frequency
VOICE-MESSAGE  Permits substituting the tone for voice message
```

#### 2.4.10.3.1 [NO] TONES DIAL FREQUENCY

Dial tone frequency. Admits values between 0 and 4000 Hz. Default is 425 Hz.

*Example:*

```
Telephony Config>TONES DIAL FREQUENCY 450
Telephony Config>
```

#### 2.4.10.3.2 [NO] TONES DIAL VOICE-MESSAGE

Substitutes a dial tone for a voice message. Admits values between 1 and 9. Default is no voice message.

*Example:*

```
Telephony Config>TONES DIAL VOICE-MESSAGE 1
Telephony Config>
```

#### 2.4.10.4 [NO] TONES DTMF-TIMEOUT

Configures the maximum wait time from when a DTMF tone has been received until dialing is complete. Admits values between 1 and 30 seconds. Default is 10 seconds.

*Example:*

```
Telephony Config>TONES DTMF-TIMEOUT 20
Telephony Config>
```

#### 2.4.10.5 [NO] TONES ERROR

Configures parameters relative to the error tone. Error tone is made up of the following sequence of elements: tone, silence 1, tone, silence 1, tone, silence 2. This sequence is continually repeated.

**Syntax:**

```
Telephony Config>TONES ERROR ?
ACTIVITY          Time, in tenths of seconds, during which the tone is active
FREQUENCY         Tone frequency
SILENCE1          Tenths of seconds, during which the tone is active
SILENCE2          Tenths of sc. during which the tone is actv each three tones
VOICE-MESSAGE     Permits substituting the tone for voice message
```

**2.4.10.5.1 [NO] TONES ERROR ACTIVITY**

Configures the time in, tenths of seconds, during which the error tone is maintained active. Admits values between 1 and 100. Default is 2 tenths of a second.

**Example:**

```
Telephony Config>TONES ERROR ACTIVITY 3
Telephony Config>
```

**2.4.10.5.2 [NO] TONES ERROR FREQUENCY**

Configures error tone frequency in hertz. Admits values between 0 and 4000 Hz. Default is 425 Hz.

**Example:**

```
Telephony Config>TONES ERROR FREQUENCY 450
Telephony Config>
```

**2.4.10.5.3 [NO] TONES ERROR SILENCE1**

This is the time in tenths of seconds during which a silence is maintained in the error tone. Admits values between 1 and 100. Default is 2 tenths of a second.

**Example:**

```
Telephony Config>TONES ERROR SILENCE1 3
Telephony Config>
```

**2.4.10.5.4 [NO] TONES ERROR SILENCE2**

This is the time in tenths of seconds during which a silence is maintained in the error tone each three error tones. Admits values between 1 and 100. Default is 6 tenths of a second.

**Example:**

```
Telephony Config>TONES ERROR SILENCE2 8
Telephony Config>
```

**2.4.10.5.5 [NO] TONES ERROR VOICE-MESSAGE**

Substitutes an error tone for a voice message. Admits values between 1 and 9. Default is no voice message.

**Example:**

```
Telephony Config>TONES ERROR VOICE-MESSAGE 1
Telephony Config>
```

**2.4.10.6 [NO] TONES FXO-BUSY-PATTERN**

Configures the times, levels and frequencies to detect the FXO is busy. This also configures tone detection levels. Up to five different detection patterns can be configured. By using the negative form of this command, you can set a pattern parameter by default, or define an entire pattern. For further information on how to configure hook-on detection in an FXO line, please see manual *bintec Dm770-I VoIP Interfaces*.

**Syntax:**

```
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
Telephony Config>

```

#### Command history:

Release	Modification
10.09.26	Configuration of multiple patterns (up to five) and the negative form of this command has been introduced.
11.00.05	Configuration of multiple patterns (up to five) and the negative form of this command has been introduced.
11.01.00	Configuration of multiple patterns (up to five) and the negative form of this command has been introduced.

#### 2.4.10.7 TONES FXO-SUPERVISION-PATTERN

Configures dialing tone frequencies and levels. FXO line supervision is based on detecting said tone each time a call is executed over an FXO line. For further information on how to configure supervision of an FXO line, please see manual *bintec Dm770-I VoIP Interfaces*.

##### Syntax:

```

Telephony Config>TONES FXO-SUPERVISION-PATTERN ?
freq-1-level   Freq 1 level
freq-2-level   Freq 1 level
frequency1     Tone frequency
frequency2     Tone frequency
Telephony Config>

```

#### 2.4.11 [NO] TRANSLATION

A translation is a transformation that can be applied to one of the telephone numbers intervening in a call (calling or called), as well as over a type of associated dial-plan.

A translation is identified by a number between 1 and 10000 and made up of various rules. In turn, each rule is composed of a pattern to search for and some elements to insert when the pattern matches the input data. The rules are applied in order of priority until one of them is verified. For further information on the construction of the rules, please see the rule command in this section.

Translations are always applied before applying expansions (prefix) and numerical compressions (strip-digits).

To configure a translation, execute **translation** followed by the number of the translation to configure in the translation configuration menu. **no translation** deletes the indicated translation.

##### Syntax:

```

Telephony Config>translation 1

TRNL config>?
no      Negates a command or sets its defaults
rule    Add a rule to the translation
test    Permit you to test the translation
exit

```

The following commands can be found in the translation configuration menu:

### 2.4.11.1 [NO] RULE

Adds or eliminates a rule for the active translation you are configuring. Rules are made up of a pattern to search for and a change to apply when a sequence matching the pattern is found. At the same time, the pattern being searched for is made up of a regular expression and optionally a type of dial-plan. The change comprises of a text string and, optionally, a new type of dial-plan.

The rule application mode is as follows: apply the regular expression over the telephone number and, if a sequence complying with this is found, the change is applied. The change could consist of substituting the sequence matching the regular expression for a new sequence, or adding the new sequence as a prefix or a suffix for the sequence being adjusted, and optionally substitute the type of dial-plan for that indicated in the change. If the resulting telephone number after applying the translation is greater than 15 characters, translation is not carried out.

The rules are applied in order of priority until one of them is complied with. The rules with a lowest order take the most priority. These are only applied once in the incoming telephone call.

A regular expression of a rule consists of a text sequence where the following characters have a special meaning:

.	Represents a single character.
*	Suffix, i.e. it does not operate alone but as a suffix of a simple or composed element. Represents the simple or composed element that follows 0 or multiple times.
+	Suffix. Represents the simple or composed element that follows 1 or multiple times.
?	Suffix. Represents the simple or composed element that follows 0 or 1 time.
[ ]	Defines a composed element that begins with character '[' and ends with ']'. Use the '-' character (between start and end elements) to include ranges of elements.
[^ ]	Located at the beginning of square brackets that operate over the composed element and complementing it. This behavior is not applicable if it is not located immediately after '['.
^	Prefix. Represents the preceding simple or composed element, if it is found at the beginning of an entry string.
\$	Suffix. Represents the simple or composed element that only follows if found at the end of the entry string.
	Defines an alternative between two regular expressions.
( )	Groups elements with various purposes: grouping alternatives for other expressions and grouping complex expressions for use with the '*', '+' and '?' suffixes.
\	Acts as an escape character, so any character following this is taken literally, its special meaning, should it have one, not having any affect.

*Examples:*

- (1) The expression 3.5 adjusts to 385 and 305.
- (2) The expression 48\* adjusts to 4, 48, 4888 etc.
- (3) The expression 47+0 adjusts to 470, 47770 but not to 40.
- (4) The expression 45?0 only adjusts to 40 and 450.
- (5) The expression [12-4] adjusts to 12, 13 and 14 but not to 15, 16 etc.
- (6) The expression [^6-9]0 adjusts to numbers that do not have the 60, 70, 80 or 90 sequence.
- (7) The expression ^091 adjusts to telephone numbers that begin with 091.
- (8) The expression 3\$ adjusts to telephone numbers that end in 3.
- (9) The expression 91/93 searches for the 91 or 93 sequences in a telephone number.
- (10) The expression 0(91/93) searches for sequences 091 or 093 in a telephone number.
- (11) The expression 0(91/93) adjusts to 0, 091 and 093 sequences.

Dial-plan types are defined by the ITU-T Q.931 standards and are as follows.

unknown	Network
International	Subscriber

National	Abbreviated
	Reserved

**Syntax:**

```
TRNL config>RULE <id> behaviour <behaviour>
                    id restrict
                    match <pattern> <new pattern>
                    pattern-type <type>
                    new-type <type>
TRNL config>
```

<b>ID</b>	The rule identifier. This defines the priority of the rule: a lower order implies a higher priority. Admits values between 1 and 10000.
<b>Match</b>	As a first parameter: command accepts a regular expression, which is applied over a telephone number. Second parameter: sequence to be inserted where a sequence, which adjusts to the rule pattern, is found.
<b>Behaviour</b>	The rule behavior can be: replace, suffix, prefix, don't-alter and blacklist.  With replace, the pattern found is substituted for the new pattern. For suffix or prefix, the new pattern is inserted either behind or in front of the found pattern. Dont- <i>alter</i> does not modify the original number and <i>blacklist</i> identifies the number as pertaining to a blacklist and the call is rejected.
<b>Id-restrict</b>	Hides calling number in calls matching this rule.
<b>Pattern-type</b>	The type of dial-plan that a telephone number must have so the translation can take place, i.e. so the second part of the rule can be applied. The following values are admitted: unknown, international, national, network, subscriber, abbreviated, reserved and any.  Default is <i>any</i> ; therefore any type of number applies the rule.
<b>New-type</b>	Type of dial-plan the telephone number has as a result of applying the translation. This admits the following values: unknown, international, national, network, subscriber, abbreviated, reserved and any.  If this is configured to <b>any</b> (default), the dial-plan type is not modified.

**Examples:**

**Example 1.** A rule that changes the numbers ending in 85 or 87 to numbers ending in 00. The type of entry dial-plan is unknown and left the same.

```
TRNL config>RULE 1 match (85|87)$ 00
TRNL config>RULE 1 pattern-type unknown
TRNL config>
```

**Example 2.** The rule only changes the dial-plan type, from network type to international type.

```
TRNL config>RULE 10 match "" ""
TRNL config>RULE 10 pattern-type network
TRNL config>RULE 10 new-type international
TRNL config>
```

**Example 3.** The rule changes the 6x6 prefixes for 99902 prefixes. Independently of the original dial-plan type, this changes to national.

```
TRNL config>RULE 20 match ^6.6 99902
TRNL config>RULE 20 new-type national
TRNL config>
```

**2.4.11.2 TEST**

Tests the current translation to check if the rules, making up the translation, adequately define its behavior.

Request a telephone number and a dial-plan type. If there is a rule matching this data, the resulting display is applied to the rule. If there is nothing that fits the rule, nothing is displayed. The telephone number admits a sequence of up to 15 characters.

**Syntax:**

```
TRNL config>TEST 918076565 unknown
TRNL config>
```



*Example:*

Here, you are going to check the defined rules (as an example) in the **rule** command from this menu.

```
TRNL config>TEST 8522222299 unknown
TRNL config>TEST 8522222285 unknown
      8522222200 (unknown)
TRNL config> TEST 8522222287 unknown
      8522222200 (unknown)
TRNL config> TEST 8522222288 unknown
TRNL config>TEST 932530222 network
      932530222 (international)
TRNL config>TEST 932530222 national
TRNL config>TEST 616900200 unknown
      99902900200 (national)
TRNL config> TEST 918076169 unknown
TRNL config>
```

**2.4.11.3 EXIT**

Returns to the telephony configuration menu.

*Syntax:*

```
TRNL config>exit
```

**2.4.12 [NO] VOICE-CLASS**

Defines a list of codecs. Once defined, said lists can be associated with a dial-peer so, when a call matches said dial-peer, the codecs defined in the list are used. **no voice-class** eliminates the indicated list.

*Syntax:*

```
Telephony Config>VOICE-CLASS <1..10000> Value in the specified range
```

At this point, you pass to a voice-class configuration menu. Default is there is no voice-class defined in the device.

**2.4.12.1 CODEC PREFERENCE**

Adds a codec to the list currently being configured.

*Syntax:*

```
voice class 1 config>CODEC-PREFERENCE <codec> FRAMES-PER-PACKET <1-6>
```

The frames-per-packet parameter indicates the number of voice frames sent in each rtp packet. If nothing is indicated, the default value is used and this depends on each codec:

G711: 2 frames in each RTP packet.

G729: 2 frames in each RTP packet.

G723: 1 frame in each RTP packet.

Please note: the higher the number of frames per rtp packet, the fewer number of packets are transmitted. This saves bandwidth but the delay is greater.

**2.4.12.2 NO CODEC PREFERENCE**

Deletes all codecs from said list.

*Syntax:*

```
voice class 1 config>NO CODEC-PREFERENCE
```

*Example:*

Definition of a list, which we will call 101, with three codecs, as the first option g711 law A is used, as the second g711 law u and as the third g729 with three frames per packet.

```
voice-class 101
  codec-preference g711alaw
```

```
codec-preference g711ulaw
codec-preference g729 frames-per-packet 3
exit
```

### 2.4.13 [NO] VOICE-MESSAGE

Associates a voice message identifier with a file name that must be in wav format. The encoding of said wav files must be G711 rule a or rule u, sampled at 8 KHz and a single channel (mono).

Subsequently, this voice message identifier can be used as a dialing tone, error, etc., in FXS/FXO lines; such as music on hold in any of the device's conventional telephony interfaces or as a voice message to be played in virtual interfaces (please see manual *bintec Dm770-I VoIP Interfaces*).

*Syntax:*

```
Telephony Config>VOICE-MESSAGE <id> filename <filename>
```

## Chapter 3 Generic Telephony Monitoring

### 3.1 Telephony Monitoring Commands

Telephony monitoring commands must be entered in the monitoring menu associated with telephony ( *Telephony Mon+*). To access this menu, enter **TELEPHONY** (general monitoring menu (+)).

```
+TELEPHONY
Telephony Monitor
Telephony Mon+
```

Once accessed, enter the commands described below.

#### 3.1.1 CALL-MAKE

Calls a phone number using one of the device telephone lines. To identify the line executing the call, you need the board slot, interface instance number and the line number.

*Syntax:*

```
Telephony Mon+call-make <slot> <instance> <line> <telephone number>
Telephony Mon+
```

Consequently, if the board is POTS, this is in spot 2 and wants to use line 3, the following data is needed:

```
Telephony Mon+call-make 2 0 3
```

If the interface does not have hardware, the voip interface number is used as the slot and the instance 0. Consequently, to make a call with voip 1000 interface, the following data is needed:

```
Telephony Mon+call-make 1000 0 1
```

If the call cannot be made, an error is given. This command hooks-off the line as a further part of the process of making the call. To terminate the call, hang-up the line through the monitoring command **hang-up** (explained in this chapter).

#### 3.1.2 CALL-WAITING

Places a call associated with an interface line on hold and place a new call to a given telephone number. This also allows you to return to a call that has been put on hold.

*Syntax:*

```
Telephony Mon+call-waiting ?
  end      Quit a call from hold
  start    Put a call on hold
Telephony Mon+
```

##### 3.1.2.1 CALL-WAITING END

Returns to a call that has been put on hold. To identify the line, you need the board slot, the interface instance number and line number.

*Syntax:*

```
Telephony Mon+call-waiting end <slot> <instance> <line>
Telephony Mon+
```

This command will return an error if there is no call on the line.

##### 3.1.2.2 CALL-WAITING START

Puts a call on hold and places a new call. To identify the line making the call, you need the board slot, the interface instance number and line number.

*Syntax:*

```
Telephony Mon+call-waiting start <slot> <instance> <line> <telephone_number>
```

```
Telephony Mon+
```

This command returns error if there is no call on the line.

### 3.1.3 CLEAR

Deletes the executed calls register.

*Syntax:*

```
Telephony Mon+clear ?
released-calls Delete all the released calls register
```

#### 3.1.3.1 CLEAR RELEASED-CALLS

Deletes the released calls register.

*Example:*

```
Telephony Mon+clear released-calls
Telephony Mon+
```

### 3.1.4 DISPLAY

Dynamically displays the status of the voice messages.

*Syntax:*

```
Telephony Mon+display ?
voice-messages Display the status of the voice messages
```

#### 3.1.4.1 DISPLAY VOICE-MSG

Displays voice messages present in the router RAM memory. A voice message is loaded in the memory on restarting the router if it has been configured as an error tone, communication tone, etc., or if, once the device has been started, a voice message has been recorded on disk.

Information is also displayed on the size in bytes, the codec the voice message was recorded with and format version used. Voice messages in proprietary format, recorded in an FXS interface, now have version 2 and the loaded wav files now appear as version 3.

*Example:*

```
Telephony Mon+display voice-msg

These voice messages are loaded in memory:
Voice number  SIZE          CODEC          VERSION
-----
1             3770             G729           2
2             7992             G723           1
3             3648             G723           2
4             5550             G729           2
1             23808            G711alaw      3
Telephony Mon+
```

### 3.1.5 DUMP-CDRS

If there is a telephone register recording file configured (CDRs), recording begins to said file, independently of the configured time interval (see **call application history file**).

*Syntax:*

```
Telephony Mon+dump-cdrs
```

### 3.1.6 HANG-UP

Hangs up a line that is not hook-on. If said line has a call associated, this ends it. If the line is hook-on, then operation has no effect. To identify the line, you need the board slot, interface instance number and line number.

**Syntax:**

```
Telephony Mon+hang-up <slot> <instance> <line> <telephone number>
Telephony Mon+
```

Consequently, to hang up line 2 on a POTS board located in slot 3, the data to provide is:

```
Telephony Mon+hang-up 3 0 2
Telephony Mon+
```

### 3.1.7 LIST

**Syntax:**

```
Telephony Mon+list ?
calls          List call information
causes        List the call release codes
cdr           Lists cdr information
dial-peers    List dial-peer information
statistics    List gateway statistics
```

#### 3.1.7.1 LIST CALLS

**Syntax:**

```
Telephony Mon+list calls?
active        List the calls that are currently active
```

##### 3.1.7.1.1 LIST CALLS ACTIVE

Lists currently active calls. Displays the associated dial-peers, calling and called parties, call type, codec, number of bytes transmitted and received, etc.

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

**Syntax:**

```
Telephony Mon+list calls active
```

**Example:**

```
Telephony Mon+list calls active
Call In: tel 12 dialpeer 2 Out: tel 23 dialpeer 3
Codec g711Alaw64k, Vad Yes, NOB 1, Fax No, Bytes 213520/800 (tx/rx)
In:
  buffered rx frames 0, missed rx frames 7 since last print
  Gateway Interface Peer voip1/0-2
Out:
  SIP peer IP 172.24.100.133
  RTP Delay: 90/90/90 (val/max/min), Variance 3/3/0 (val/max/min), RFC2833 DTMF: No,
  timestamps 0/215360 (in/out)
Telephony Mon+
```

#### 3.1.7.2 LIST CDR

**Syntax:**

```
Telephony Mon +list cdr ?
active        List the calls that are currently active
released     List last released calls
```

##### 3.1.7.2.1 LIST CDR ACTIVE

Lists all currently active calls. Displays the associated line, calling and called parties, call type, lines involved in the call, codec, number of bytes transmitted and received, time and date of the start of the call and time and date when the call connected.

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

**Syntax:**

```
Telephony Mon+list cdr active <registers to display>
```

**Example:**

```
Telephony Mon+list cdr active 1
CallId:          4 CallingAddress:      6001 CalledAddress:      1
CallType:        SIPServ CallingPeer:   172.24.100.129 CalledPeer: 172.24.27.129
Codec:           G711(ALaw) TxBytes:    0 RxBytes:          0 ReleaseCause:
SetupTime:       2006-01-02 18:21:35   ConnectTime: 2006-01-02 18:21:40
ReleaseTime:     Duration: 00:12:10
Telephony Mon+
```

**3.1.7.2 LIST CDR RELEASED**

Lists the latest 15 calls released, although this number is configurable. The information displayed is identical to active calls but also reports the date and time of disconnection and cause.

On listing the called telephone, information on the result of applying configured numeric expansions is displayed both for incoming and outgoing calls. The release cause appears in Q.850 format.

**Syntax:**

```
Telephony Mon+list cdr released <registers to display>
```

**Example:**

```
Telephony Mon+list cdr released 1
CallId:          0 CallingAddress:      613 CalledAddress:      0
CallType:        H323 CallingPeer:     voip1/0-1 CalledPeer: 172.24.100.129
Codec:           G723.1(5K3) TxBytes:  805540 RxBytes:    804300 ReleaseCause: 19
SetupTime:       2006-01-02 17:06:29   ConnectTime: 2006-01-02 17:06:35
ReleaseTime:     2006-01-02 17:26:43   Duration: 00:20:08
Telephony Mon+
```

**3.1.7.3 LIST DIAL-PEER****Syntax:**

```
Telephony Mon+list dial-peers?
  brief      Display information on the active dial-peers
  detailed   Display detailed information on the active dial-peers
```

**3.1.7.3.1 LIST DIAL-PEER BRIEF**

Displays information on active dial-peers. Reports dial-peer destination, target, and ID, and the number of times it has matched a search.

**Example:**

```
Telephony Mon+list dial-peers brief

Peer Id 1 type voice-port matched 0 times
Destination 11
Target: voip1/0-1

Peer Id 2 type voice-port matched 1 times
Destination 12
Target: voip1/0-2

Peer Id 3 type sip matched 4 times
Destination 2.
Target: ipv4 172.24.100.130 port 5060

Telephony Mon+
```

**3.1.7.3.2 LIST DIAL-PEER DETAILED**

Displays detailed information on active dial-peers. This also includes information relative to SIP registers over the previous command.

**Example:**

```

Telephony Mon+list dial-peers detailed

Peer Id 1 type voice-port matched 0 times
Destination 11
Target: voip1/0-1
Sip Info: Next register in 30 seconds, registered No
          User: , Cseq: 0, CallId:

Peer Id 2 type voice-port matched 1 times
Destination 12
Target: voip1/0-2
Sip Info: Next register in 30 seconds, registered No
          User: , Cseq: 0, CallId:

Peer Id 3 type sip matched 4 times
Destination 2.
Target: ipv4 172.24.100.130 port 5060
Sip Info: Next register in 0 seconds, registered No
          User: , Cseq: 0, CallId:

Telephony Mon+

```

**3.1.7.4 LIST STATISTICS**

Lists several statistics relating to both the device and last calls made per line.

**Syntax:**

```

Telephony Mon+list statistics ?
gw      List gateway statistics

```

**3.1.7.4.1 LIST STATISTICS GW**

Lists statistics common to all device lines. This displays date and time the system started up and whether voice service is available. It also shows the total number of calls since the system started, number of active calls and number of possible calls. Finally, the average length of calls is shown.

**Example:**

```

Telephony Mon+list statistics gw

Startup time: 16:01:27          10/ 5/99
Voice Service: disabled
Total calls: 0 (0)
Active calls: 0
Possible additional calls: 4
Conversation mean time: 0 seconds.

Telephony Mon+

```

**3.1.8 PICK-UP**

Picks up a hook-on line. If the line is already hook-off, then this operation will have no affect. To identify the line, you need the board slot, the interface instance number and line number.

**Syntax:**

```

Telephony Mon+pick-up <slot> <instance> <line> <telephone number>
Telephony Mon+

```

Therefore, to pick up line 2 on a POTS board located in slot 3, the following data is needed:

```

Telephony Mon+pick-up 3 0 2
Telephony Mon+

```

This command only picks up a line; it does not carry out any dialing.

## 3.1.9 SRTP

### 3.1.9.1 SRTP DEBUG

Activates the SRTP protocol debugging events generation and displays them through the TLPHY.028 event.

## 3.1.10 TRACE

### 3.1.10.1 TRACE RCD

Debugging command for company use. This should only be used by our Technical Support personnel.

## 3.1.11 VOICE

Reproduces a previously recorded voice message.

*Syntax:*

```
Telephony Mon+voice <slot> <instance> <line>
```



## Chapter 4 Third Party Software

### 4.1 Third Party Software

Implementation of the SRTP protocol uses the libSRTP library code.

The following is a copy of the libSRTP license.

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