40- Voice Configuration Commands

40.1 Voice Configuration Command Tree

Description

This chapter gives an overview of nodes that are handled by "Voice Configuration Commands".

Command Tree

----configure
    ----voice
        ----sip
            ----[no] server
                - (name)
                - [no] admin-status
                - address
                - [no] port
                - [no] server-role
                - [no] priority
                - [no] weight
                - [no] site
                - [no] transproto
                - [no] dnsname-type
                - admin-domain-name
                - [no] tls-port
                - [no] tls-cafile
            ----lineid-syn-prof
                - (name)
                - [no] syntax-pattern
                - [no] pots-syntax
                - [no] isdn-syntax
                - [no] cas-r2-syntax
                - [no] cas-r1-syntax
            ----[no] user-agent
                - (name)
                - [no] ip-addr-policy
                - [no] ip-config-mode
                - [no] signal-gateway-ip
                - signal-vlan-id
                - [no] signal-dscp
                - [no] signal-pbits
                - [no] signal-link-mtu
                - [no] rtp-gateway-ip
                - [no] rtp-vlan-id
                - [no] rtp-dscp
                - [no] rtp-pbits
                - [no] rtp-link-mtu
                - [no] dhcp-optreq-list
                - [no] admin-status
                - [no] provider-name
                - [no] lsa-rtp-gw-ip
                - [no] lsa-rtp-vlan-id
                - [no] lsa-md5-realm
                - [no] lsa-md5-password

- [no] rtp-ipv6-gateway
- [no] rtp-ipv6-vlan-id
- [no] rtp-ipv6-dscp
- [no] rtp-ipv6-pbits
- [no] rtp-ipv6-link-mtu

----[no] vsp

- **(name)**
- domain-name
- [no] admin-status
- [no] tinfo
- [no] ta4
- [no] ttir1
- [no] t-acm-delta
- [no] access-held-time
- [no] awaiting-time
- [no] digit-send-mode
- [no] overlap-484-act
- [no] dmpm-intdgt-expid
- [no] dial-start-timer
- [no] dial-long-timer
- [no] dial-short-timer
- [no] uri-type
- [no] rfc2833-pl-type
- [no] rfc2833-process
- [no] min-data-jitter
- [no] init-data-jitter
- [no] max-data-jitter
- [no] release-mode
- [no] dyn-pt-nego-type
- [no] vbd-g711a-pl-type
- [no] vbd-g711u-pl-type
- [no] vbd-mode
- [no] warmline-dl-timer
- timer-b
- timer-f
- timer-t1
- timer-t2
- [no] reg-sub
- [no] dtmf-sip-info
- [no] sub-period
- [no] sub-head-start
- [no] t38-same-udp
- [no] dhcp-option82
- [no] sspprofile
- [no] signaling-ipmode
- [no] tls-cafile
- [no] media-ipmode

----user-agent-ap

- **(ua-name)**
- **slot-id**
- signal-ip
- [no] rtp-ip
- [no] dhcp-fqdn
- [no] dhcp-customer-id
- [no] admin-status
- [no] lsa-rtp-ip
- [no] rtp-ipv6-address

----[no] dialplan

- **- (name)**
- [no] pre-activated
- [no] static-prefix
- [no] static-suffix
- [no] digitmap-mode
- [no] provider-name

----[no] digitmap

- **- (name)**
- **- type**
- **- rule**
- [no] access-type

----[no] termination

- **- (if-index)**
- [no] directory-number
- [no] user-name
- [no] display-name
- [no] uri
- [no] direct-uri
- [no] line-feed
- [no] md5-realm
- [no] md5-password
- [no] aka-secret-key
- [no] admin-status
- [no] clip-mode
- [no] telc-clip-mode
- [no] anti-tapping
- [no] impedance
- [no] rx-gain
- [no] tx-gain
- [no] warmline-service
- [no] linesig-remanswer
- [no] line-id
- [no] force-hold
- [no] callwait-service
- [no] callhold-service
- [no] callconf-service
- [no] calltras-service
- [no] fast-guard
- [no] clear-forward
- [no] testaccessstate
- [no] busyoverwrite
- [no] accessontimeout
- [no] provider-name
- [no] register_recall
- [no] sip-rule-set
- [no] reserved
- [no] pulse-dialing
- [no] auto-answer

----tca

  - [no] enable
  - [no] high-jbfl
  - [no] low-jbfl

----[no] sharedline

- **- (if-index)**
- [no] directory-number
- [no] user-name
- [no] display-name
- [no] uri

- [no] admin-status
- [no] clip-mode
- [no] callwait-service
- [no] callhold-service
- [no] callconf-service
- [no] calltras-service

----register

- **(provider-name)**
- [no] register-uri
- [no] register-intv
- [no] reg-retry-intv
- [no] reg-prev-ava-intv
- [no] reg-head-start
- [no] reg-start-min
- [no] init-reg-delay

----[no] transport

- **(trans-protocol)**
- **provider-name**
- [no] admin-status
- [no] port-rcv
- [no] tcp-idle-time
- [no] max-out-udp-size
- [no] tls-port-rcv

----redundancy

- **(admin-domain-name)**
- [no] support-redun
- [no] dns-purge-timer
- [no] dns-ini-retr-int
- [no] dns-max-retr-nbr
- [no] fg-monitor-method
- [no] fg-monitor-int
- [no] bg-monitor-method
- [no] bg-monitor-int
- [no] stable-obs-period
- [no] fo-hystersis
- [no] del-upd-threshold
- [no] auto-server-fo
- [no] auto-server-fb
- [no] auto-sos-fo
- [no] auto-sos-fb
- [no] rtry-after-thrsh
- [no] options-max-fwd
- [no] dns-redun-mode
- [no] fail-obs-timer
- [no] fg-intv-503
- [no] time-thrsh-503
- [no] nbr-thrsh-503
- [no] auto-srv-fo-timer

----system

  ----session-timer

  - **X** [no] enable
  - [no] status
  - [no] min-se-time
  - [no] se-time
  - [no] admin-status

----[no] dnsserver

- **(name)**
- [no] admin-status
- address

- [no] priority
- [no] site
- admin-domain-name
- [no] port

----[no] dhcp-authent-para

- **(ua-name)**
- **secret-id**
- [no] key
- [no] action-type

----redundancy-cmd

- **(domain-name)**
- [no] start-time
- [no] end-time
- fail-x-type

----statistics

- [no] stats-5min-config
- [no] cdr-config

----stats-config

- [no] per-line
- [no] per-board
- [no] per-system
- [no] per-call
- [no] out-any-rsp
- [no] out-180-rsp
- [no] out-200-rsp
- [no] in-any-rsp
- [no] in-180-rsp
- [no] in-200-rsp

----[no] isdn-cas-term

- **(if-index)**
- [no] directory-number
- [no] user-name
- [no] display-name
- [no] uri
- [no] md5-realm
- [no] md5-password
- [no] aka-secret-key
- [no] admin-status
- [no] line-id
- [no] testaccessstate
- [no] busyoverwrite
- [no] accessontimeout
- [no] provider-name
- [no] signalling-type
- [no] cpn-screen
- [no] default-cpc
- [no] restrict-acc
- [no] hunting-way
- [no] sip-rule-set
- [no] cas-exch-type
- [no] q-value
- [no] reg-mode
- [no] e1-hunting-way
- [no] e1-cluster-itf
- [no] e1-cluster-name
- [no] isdn-nsm-profile
- [no] cas-nsm-profile

----[no] pri-cas-tksg-term

- (if-index)
- tksg-id
- [no] directory-number
- [no] user-name
- [no] display-name
- [no] uri
- [no] realm
- [no] password
- [no] aka-secret-key
- [no] admin-status
- [no] line-id
- [no] provider-name
- [no] ds0-hunting-way
- [no] itf-hunting-way
- [no] sip-rule-set
- [no] q-value
- [no] reg-mode
- [no] extension
- trunk-group-name
- [no] signalling-type
- [no] cas-cpn-screen
- [no] cas-default-cpc
- [no] cas-restrict-acc
- [no] cas-exch-type
- [no] isdn-nsm-profile
- [no] cas-nsm-profile
- [no] uus-profile
- [no] cas-timer-profile

----[no] manipulate-rule
- (rule-id)
- [no] admin-status
- [no] rule-name
- rule
- [no] rule-type

----cas-r2-timer
- (if-index)
- [no] seizure-ack
- [no] disconnect-ack
- [no] force-rel-ack
- [no] pres-fwd-sig
- [no] recept-fwd-sig
- [no] re-answer
- [no] call-process

----[no] isdn-cas-cls-puid
- (name)
- directory-number
- [no] user-name
- [no] uri
- [no] md5-password
- [no] aka-secret-key
- [no] display-name

----[no] pri-cas-tksg-puid
- (if-index)
- tksg-id
- directory-number
- [no] user-name
- [no] uri
- [no] password
- [no] aka-secret-key

- [no] display-name

----[no] isdn-nsm-prof
- **(prof-name)**
- [no] version-nbr
- [no] outg-from-no-cgpn
- [no] outg-privacy
- [no] outg-from-pi
- [no] inc-from-noclip
- [no] inc-from-clip-pri
- [no] inc-from-clip-ua
- [no] inc-npi-to-cpn
- [no] inc-ton-to-cpn
- international-prefix
- [no] national-prefix
- country-code
- outg-cpn-length
- [no] inc-cgpn-add-ac
- [no] inc-ton-to-cdpn

----[no] cas-nsm-prof
- **(prof-name)**
- [no] version-nbr
- [no] outg-from-no-cgpn
- international-prefix
- [no] national-prefix
- country-code
- outg-cpn-length

----[no] user-to-user-prof
- **(prof-name)**
- [no] version-nbr
- [no] service-type
- [no] act-estab-call
- [no] act-stable-call
- [no] uus3-t1-timer

----[no] tksg
- **(if-index)**
- **tksg-id**
- **interface**
- tksg-name
- ds0-bitmap
- [no] admin-status
- [no] ds0-hunting-way

----[no] phys-itf
- **(if-index)**
- [no] admin-status
- [no] interface-type
- [no] signalling-type
- [no] e1-crc4frame
- [no] t1-framemode
- [no] t1-tx-linecode
- [no] t1-rx-linecode
- [no] t1-lbo-value
- [no] blocking-bitmap
- [no] cap-name
- [no] cas-sig-prof

----[no] cas-sig-prof
- **(prof-name)**
- [no] version-nbr
- [no] idle
- [no] seize

- [no] seize-ack
- [no] answer
- [no] fw-disconnect
- [no] fw-disconnect-ack
- [no] bw-disconnect
- [no] bw-disconnect-ack
- [no] fault
- [no] block
- [no] meter
- [no] reject-call
- [no] bw-disc-reanswer

----[no] cap

- **(cap)**
- [no] cap-name
- [no] ne-name
- [no] mode
- [no] location
- [no] admin-status
- [no] ip
- [no] udp-port
- [no] gateway-ip
- [no] provider-name

----[no] ne-board

- **cap-id**
- **ne-id**
- **board**
- [no] admin-status
- xhub-port

----[no] cas-timer-prof

- **(prof-name)**
- [no] version-nbr
- appl-bitmap
- [no] seize-ack
- [no] clear-back-ack
- [no] clear-fw-ack
- [no] presence-fw-sig
- [no] receipt-fw-sig
- [no] re-answer
- [no] call-process
- [no] delay-dial-min
- [no] delay-dial-max
- [no] seize-state-time
- [no] reg-recall-min
- [no] reg-recall-max
- [no] decadic-pulse-dur
- [no] decadic-break-dur
- [no] inter-digit-peri
- [no] ring-fw
- [no] receipt-nbr-rcvd
- [no] meter-pulse-dur

----[no] network-element

- **cap-id**
- **ne-id**
- ne-name
- [no] role
- [no] admin-status
- [no] ip
- [no] vlan-id

----[no] call-query

- (id)
- [no] timeout-period
- target-dn
- [no] remote-dn
- [no] type

----cluster

- (cluster-id)

----ip
- [no] ivps-ip
- [no] netmask
- [no] router-ip
- vlan-id
- [no] ip-mode
- [no] dhcpoption60
- [no] private-ip
- [no] private-netmask
- [no] private-vlan-id

----[no] equipment

- (equip-id)
- asam-id
- ip-address
- [no] next-hop

----[no] board

- (board-id)
- planned-type
- lanx-port

----[no] termination

- (port-id)
- [no] type
- [no] isdn-codec
- [no] switch-type
- [no] activate-type
- termination-id
- media-gateway-id
- [no] admin-status
- [no] line-feed
- [no] rx-gain
- [no] tx-gain
- [no] impedance
- [no] rtp-dscp
- [no] rtp-pbits
- [no] clip-mode
- [no] metering-type
- [no] directory-number
- [no] voice-service
- [no] reserved

----tca
- [no] tca-enable
- [no] rtp-pktloss-thres
- [no] rtp-jitter-thres
- [no] rtp-delay-thres

----[no] li
- [no] state
- src-port
- dest-ip
- dest-port

----[no] sharedlineterm

- (port-id)
- sharedlineterm-id

- [no] admin-status
- [no] directory-number
- [no] voice-service

----[no] media-gateway
- **(media-gateway-id)**
- [no] name
- [no] ip-mode
- [no] dhcpoption60
- [no] ip-address
- [no] netmask
- [no] udp-port
- [no] router-ip
- vlan-id
- [no] mgc-type
- prim-mgc-ip
- [no] mgc-id
- [no] prim-mgc-udp
- [no] sec-mgc-ip
- [no] sec-mgc-udp
- [no] tert-mgc-ip
- [no] tert-mgc-udp
- [no] quat-mgc-ip
- [no] quat-mgc-udp
- [no] esa-mgc-service
- [no] mg-mid-type
- [no] mg-domain-name
- [no] svcreason-format
- [no] mg-profile-name
- [no] admin-status
- [no] termid-type
- [no] pstn-term-format
- isdn-term-format
- isdn-suffix1
- isdn-suffix2
- [no] max-transhandling
- [no] max-network-delay
- [no] max-retrans
- [no] red-bat-delay
- [no] release-delay
- [no] release-type
- [no] wt-rls-delay
- [no] active-heartbeat
- [no] passive-heartbeat
- [no] retrans
- [no] max-waiting-delay
- [no] prov-rpl-time
- [no] signal-dscp
- [no] signal-pbits
- [no] rtp-dscp
- [no] rtp-pbits
- [no] event-req-id
- [no] stml-stdsg-evt
- [no] al-of-evt
- [no] al-on-evt
- [no] al-of-strict-evt
- [no] al-on-strict-evt
- [no] mg-overload-evt
- [no] mg-dummy-evt

- [no] rfc2833-pl-type
- [no] rfc2833-process
- [no] dial-start-timer
- [no] dial-long-timer
- [no] dial-short-timer
- [no] min-data-jitter
- [no] init-data-jitter
- [no] max-data-jitter
- [no] ephe-term-prefix
- [no] ephe-term-min
- [no] ephe-term-max

----[no] signal-gateway

- (signal-gateway-id)
- prim-asp-ip
- prim-sctp-port
- [no] sec-asp-ip
- [no] sec-sctp-port
- [no] tert-asp-ip
- [no] tert-sctp-port
- [no] quat-asp-ip
- [no] quat-sctp-port
- ip-address
- sgi-user-label
- sgi-mgi
- [no] admin-status

----eont

----[no] sip-useragent

- (ont-idx)
- [no] udp-port
- proxy-pri-ip
- [no] proxy-pri-port
- proxy-sec-ip
- [no] proxy-sec-port
- reg-pri-ip
- [no] reg-pri-port
- reg-sec-ip
- [no] reg-sec-port
- outbound-ip
- [no] outbound-port
- [no] reg-intval
- [no] heartbeat-mode
- [no] heartbeat-cycle
- [no] heartbeat-count
- digit-map-prof

----[no] pots

- (uni-idx)
- [no] admin-up
- [no] user-account
- [no] user-name
- [no] user-pwd
- tid-name

----[no] comm-para

- (ont-idx)
- [no] ip-mode
- ip
- mask
- gateway
- [no] pppoe-mode

- [no] pppoe-username
- [no] pppoe-pwd
- [no] tagged-mode
- cvlan
- [no] svlan
- [no] pbit
- [no] t38-admin
- [no] fax

----[no] media-gateway

- **(ont-idx)**
- [no] udp-port
- mgc-pri-ip
- [no] mgc-pri-port
- [no] mgc-sec-ip
- [no] mgc-sec-port
- [no] reg-mode
- gw-id
- [no] heartbeat-mode
- [no] heartbeat-cycle
- [no] heartbeat-count
- tid-num
- tid-prefix
- tid-digit-begin
- tid-mode
- tid-digit-len

----lsa-server

----[no] system

- **(name)**
- slot-id
- country-code
- area-code
- [no] admin-status

----[no] instance

- **(name)**
- slot-id
- sign-ip
- [no] sign-port
- sign-vlan
- [no] sign-gw-ip
- [no] admin-status
- [no] sg-name
- [no] lsa-md5-realm
- [no] lsa-md5-password

----[no] emergency-service

- **dn**
- **gateway-dn**
- slot-id
- [no] name
- [no] sg-name
- [no] gateway-ddidn
- [no] gateway-dn-type

----[no] extline

- **(if-index)**
- local-ip
- [no] gateway-ip
- [no] ip-addr-policy
- vlan-id
- fxo-ip
- fxo-port

- [no] admin-status
- [no] codec
- [no] ptime
- [no] echo-cancel
- [no] silence-suppr
- [no] impedance
- [no] rx-gain
- [no] tx-gain
- [no] dscp
- [no] pbits

----[no] pseudo-wire

- **(if-index)**
- channelset
- [no] ip-addr-policy
- local-ip
- dest-ip
- dest-udp-port
- [no] gateway-ip
- vlan-id
- [no] p-bits
- [no] dscp
- [no] admin-status
- [no] data-jitter
- p-time
- [no] payload-type

40.2 Voice Sip Server Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip Server profile.

IPv6 is actually not supported. All references to IPv6 must be ignored since mentioned for future usage only.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no server (name)) | (server (name) [no admin-status | admin-status <Sip::ServerAdminStatus>] address <Sip::ServerAddr> [no port | port <Sip::ServerPort>] [no server-role | server-role <Sip::ServerRole>] [no priority | priority <Sip::ServerPriority>] [no weight | weight <Sip::ServerWeight>] [no site | site <Sip::ServerSite>] [no transproto | transproto <Sip::ServerTransproto>] [no dnsname-type | dnsname-type <Sip::DnsDomNameType>] admin-domain-name <Sip::AdminDomName> [no tls-port | tls-port <Sip::ServerPort>] [no tls-cafile | tls-cafile <Sip::ServerCAFileName>])

Command Parameters

Table 40.2-1 "Voice Sip Server Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(name)	Format: - uniquely identifies this voice server - length: x<=32	uniquely identify of this voice server

Table 40.2-2 "Voice Sip Server Configuration Commands" Command Parameters

Parameter	Type	Description
[no] admin-status	Parameter type: <Sip::ServerAdminStatus> Format: (up | down) Possible values: - up : unlock the sip server - down : lock the sip server	*optional parameter with default value: "down"* administrative status of this sip server
address	Parameter type: <Sip::ServerAddr> Format: (ipv4 : <Ip::V4Address> | ipv6 : <Ip::V6Address> | dns : <Sip::ServerAddr>) Possible values: - ipv4 : the address type of the server is IPv4 - ipv6 : the address type of the server is IPv6 - dns : the address type of the server is DNS	*mandatory parameter* The address(IPv4 or IPv6 or DNS or FQDN) of this SIP server

Parameter	Type	Description
	Field type <Ip::V4Address> - IPv4-address Field type <Ip::V6Address> - IPv6-address Field type <Sip::ServerAddr> - name of the SipServer that can be resolved to an IP address via a DNS Server - length: 7<=x<=64	
[no] port	Parameter type: <Sip::ServerPort> Format: - the SIP server port - range: [1...65534]	*optional parameter with default value: 5060* port of voice server
[no] server-role	Parameter type: <Sip::ServerRole> Format: (proxy-server | registrar-server | registrar-and-proxy | lsa-server) Possible values: - proxy-server : proxy server - registrar-server : registrar server - registrar-and-proxy : served as a registrarServer and a proxyServer - lsa-server : served as a LSAServer	*optional parameter with default value: "proxy-server"* The role of voice application server. When redundancy is DISABLED, the operator can configure 4 different server roles, role = Proxy server, role = Registrar server, role = Proxy server AND Registrar server, role = LSA server. System will behave in accordance with configured role. When redundancy is ENABLED, the system always assume role = Proxy server AND Registrar server, irrespective of what has been configured by the operator. Thus the system always behaves according to the role = proxyAndRegistrarServer(20), and does not look at the value been configured by the operator
[no] priority	Parameter type: <Sip::ServerPriority> Format: - the SIP server priority - range: [0...65535]	*optional parameter with default value: 100* The priority of voice application server
[no] weight	Parameter type: <Sip::ServerWeight> Format: - the SIP server weight - range: [0...65535]	*optional parameter with default value: 100* The weight of voice application server
[no] site	Parameter type: <Sip::ServerSite> Format: (primary-site | geo-backup-site | sos-backup-site) Possible values: - primary-site : the server belongs to the GEO primary site - geo-backup-site : the server belongs to the GEO secondary site - sos-backup-site : the server belongs to the SOS secondary site	*optional parameter with default value: "primary-site"* The site of voice application server
[no] transproto	Parameter type: <Sip::ServerTransproto> Format: (udp | tcp	*optional parameter with default value: "udp"* The transport protocol of voice application server

Parameter	Type	Description
	| udp_tcp | tls_over_tcp | udp_tls | tcp_tls | udp_tcp_tls | udp_swo_tcp) Possible values: - udp : the transport protocol used in connecting to the server is UDP - tcp : the transport protocol used in connecting to the server is TCP - udp_tcp : the transport protocol used in connecting to the server is TCP or UDP, TCP is the preferred transport protocol - tls_over_tcp : the transport protocol used in connecting to the server is TLSoTCP - udp_tls : the transport protocol used in connecting to the server is UDP or TLSoTCP. TLSoTCP is the preferred transport protocol - tcp_tls : the transport protocol used in connecting to the server is TCP or TLSoTCP. TLSoTCP is the preferred transport protocol - udp_tcp_tls : the transport protocol used in connecting to the server is UDP, TCP or TLSoTCP. TLSoTCP is the preferred transport protocol. The next preferred transport protocol is TCP and finally UDP. If UDP would be used and the size of the message exceeds a configured threshold then a switch can be made to TCP - udp_swo_tcp : the transport protocol used in connecting to the server is UDP or TCP, when size of message exceeds a configured threshold will switchover to TCP	
[no] dnsname-type	Parameter type: <Sip::DnsDomNameType> Format: (none | dn | fqdn) Possible values: - none : If sipServerAddrType is provisioned with value ipv4 or ipv6, this object should set to none - dn : If sipServerAddrType is provisioned with value dns, it means the sipserver address is provisioned with DNS. - fqdn : If sipServerAddrType is provisioned with value dns, it means the sipserver address is provisioned with FQDN.	*optional parameter with default value: "none"* The clarification of whether a Domain Name (dn) or Fully qualified Domain Name (fqdn) was provisioned for the object sipServerAddr.
admin-domain-name	Parameter type: <Sip::AdminDomName> Format: - uniquely name of this element - length: x<=32	*mandatory parameter* The administrative domain name of the (farm of) SIP First hop(s). The administrative domain might be the VoIP Access Seeker Network or the Connectivity Serving Network.
[no] tls-port	Parameter type: <Sip::ServerPort> Format: - the SIP server port - range: [1...65534]	*optional parameter with default value: 5061* TLSoTCP port of voice server
[no] tls-cafile	Parameter type: <Sip::ServerCAFileName> Format: - the ca file name	*optional parameter with default value: ""* The name of the Certification

Parameter	Type	Description
	- length: x<=13	Authority (CA) file that applies when TLSoTCP used as underlying transport protocol for SIP

40.3 Voice Sip LineId Syntax Profile Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip LineId Syntax profile. This command can help operator to get different types of sip termination contact-user-info: physicallineid or terminationuriordn. If the syntax is 'physicalLineId', then the system will construct a value according the syntax as specified in the next parameters (depending of the type of line being pots or isdn). In case the value is 'terminationuriordn', then the value will be taken from the corresponding parameter of the SipTermination object (chapter 38.9): the uri will be taken firstly if it is valid, otherwise the dn can be taken. The keywords apply to the profile syntax include: "Access_Node_ID", "Rack", "Frame", "Slot", "ShSlt", "Port","ShPrt", "Channel". The sip termination contact-user-info can be learned using the show command: show voice sip termination.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip lineid-syn-prof (name) [no syntax-pattern | syntax-pattern <SIP::LineIdSyntaxPattern>] [no pots-syntax | pots-syntax <SIP::PotsSyntax>] [no isdn-syntax | isdn-syntax <SIP::IsdnSyntax>] [no cas-r2-syntax | cas-r2-syntax <SIP::CasR2Syntax>] [no cas-r1-syntax | cas-r1-syntax <SIP::CasR1Syntax>]

Command Parameters

Table 40.3-1 "Voice Sip LineId Syntax Profile Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(name)	Format: - the lineid syntax profile name, can only be profile1 - length: x<=32	uniquely identify of this lineid syntax profile, can only be profile1

Table 40.3-2 "Voice Sip LineId Syntax Profile Configuration Commands" Command Parameters

Parameter	Type	Description
[no] syntax-pattern	Parameter type: <SIP::LineIdSyntaxPattern> Format: (rpregisterdialog | rpregister | physicallineid | terminationuriordn) Possible values: - rpregisterdialog : random per register and dialog - rpregister : random per register - physicallineid : physical line id - terminationuriordn : termination uri	*optional parameter with default value: "terminationuriordn"* pattern of the sip LineID Syntax

Parameter	Type	Description
[no] pots-syntax	Parameter type: <SIP::PotsSyntax> Format: - syntax of the POTS SIP LineID - length: x<=128	*optional parameter with default value:* *"al/Channel/Port/Slot/Frame/Rack/Acce* syntax of the Pots sip LineID
[no] isdn-syntax	Parameter type: <SIP::IsdnSyntax> Format: - syntax of the ISDN SIP LineID - length: x<=128	*optional parameter with default value:* *"pra/Channel/Port/Slot/Frame/Rack/Acc* syntax of the Isdn sip LineID
[no] cas-r2-syntax	Parameter type: <SIP::CasR2Syntax> Format: - syntax of the Cas R2 SIP LineID - length: x<=128	*optional parameter with default value:* *"r2/Channel/Port/Slot/Frame/Rack/Acce* syntax of the Cas R2 sip LineID
[no] cas-r1-syntax	Parameter type: <SIP::CasR1Syntax> Format: - syntax of the Cas R1 SIP LineID - length: x<=128	*optional parameter with default value:* *"r1/Channel/Port/Slot/Frame/Rack/Acce* syntax of the Cas R1 sip LineID

40.4 Voice Sip User Agent Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip user agent profile.

DESCRIPTION FOR THE COMMAND PARAMETER ip-addr-policy:
- *The parameter allows to configure the internal ip address policy of the voice node can be set to the values 'distributed' or 'centralized'.*

DISTRIBUTED: Each of the voice LT boards owns (a) different IP address(es) for the signaling and media plane.
- *This mode isn't applicable to NGVR and ANSI MDU equipment i.e. only applicable to ISAM FD and ISAM FX equipment.*
- *In case the distinct VLAN approach applies, each of the voice LT boards owns 2 IP addresses i.e. a signaling IP address and a media IP address and these are different IP addresses. It is mandatory to configure for each of the SIP User Access Points a signaling and media IP address.*
- *In case the shared VLAN approach applies, each of the voice LT boards owns a single IP address i.e. the signaling IP address and the media IP address are the same. It is mandatory to configure for each of the SIP User Access Points a shared signaling and media IP address.*

CENTRALIZED: All voice LT boards shared the same IP address(es) for the signaling and media plane.
- *ISAM FD and ISAM FX equipment: the configuration of the signaling and media IP addresses in the SIP User Agent Points is optional. If configured and the distinct VLAN approach applies, all SIP User Agent points share the same signaling IP address and all SIP User Agent Access points share the same media IP address. The signaling IP address and the media IP address are the same as configured at the xHUB. The IP address of the signaling and the IP address of the media plane are different. If configured and the shared VLAN approach applies, then all SIP User Agent Access points must be configured with the same shared signaling/media IP address.*
- *NGVR and ANSI MDU equipment: the configuration of the signaling and media IP address in the SIP User Agent Access Point is mandatory. In case the distinct VLAN approach applies, the signaling and media plane own a different IP address. In case the shared VLAN approach applies, the signaling and media plane share the same IP address.*

DESCRIPTION FOR THE COMMAND PARAMETER ip-config-mode:
- *The parameter allows to configure the configuration mode of the IP related data required by the SIP User Agent, the SIP User Agent Access Point, the SIP server and the DNS server management entities and can be set to the values 'dhcp' or 'manual'.*

DHCP: The IP related data is acquired from the DHCP server.
- *ISAM FD and ISAM FX equipment: this mode can only be set when (1) the parameter ip-addr-policy is configured as 'distributed' and (2) the shared signaling/media VLAN approach applies.*
- *NGVR and ANSI MDU equipment: this mode can only be set when the shared signaling/media VLAN approach applies.*

MANUAL: The IP related data must be manually configured.

DESCRIPTION FOR THE COMMAND PARAMETER signal-gateway-ip:
- *The parameter allows to configure the IPv4 or IPv6 address of the signalling IP gateway.*
- *ISAM FD and ISAM FX equipment: A mandatory input when the parameter ip-config-mode is configured as 'manual' and the parameter ip-addr-policy is configured as 'distributed'.*
- *NGVR and ANSI MDU equipment: A mandatory input when the parameter ip-config-mode is configured as*

'manual'.

DESCRIPTION FOR THE COMMAND PARAMETER rtp-gateway-ip:

- *The parameter that allows to configure the IPv4 address of the media IP gateway.*
- *ISAM FD and ISAM FX equipment: A mandatory input parameter when the parameter ip-config-mode is configured as 'manual' and the parameter ip-addr-policy is configured as 'distributed' and the distinct VLAN approach applies. In case of the shared VLAN approach, the value configured for the parameter signal-gateway-ip is inherited.*
- *NGVR and ANSI MDU equipment: A mandatory input parameter when parameter ip-config-mode is configured as 'manual' and the distinct VLAN approach applies. In case of the shared VLAN approach, the value configured for the parameter signal-gateway-ip is inherited.*

DESCRIPTION FOR THE COMMAND PARAMETER rtp-link-mtu:

- *The parameter allows to configure the link mtu of the IPv4 media vlan. Only relevant in case of the distinct signaling/media VLAN approach. In case of the shared signaling/media VLAN approach, the value configured for the parameter signal-link-mtu is inherited.*

DESCRIPTION FOR THE COMMAND PARAMETER dhcp-optreq-list:

- *The parameter allows to configure the list of DHCPv4 options the SIP User Agent wants the DHCPv4 server to return in option 55.*
- *ISAM FD and ISAM FX equipment: only relevant in case the parameter ip-config-mode is configured as 'dhcp' (parameter ip-addr-policy is configured as 'distributed' and the shared signaling/media VLAN approach applies).*
- *NGVR and ANSI MDU equipment: only relevant in case the parameter ip-config-mode is configured as 'dhcp' (parameter ip-addr-policy is configured as 'centralized' and the shared signaling/media VLAN approach applies).*

DESCRIPTION FOR THE COMMAND PARAMETER lsa-md5-realm:

- *The parameter allows to configure the realm to be used for the registration of SIP terminations to the LSA server.*
- *The parameter lsa-md5-realm must be configured with the same value as configured for the parameter lsa-md5-realm of the LSA server instance.*

DESCRIPTION FOR THE COMMAND PARAMETER lsa-md5-password:

- *The parameter allows to configure the password to be used for the registration of SIP terminations to the LSA server.*
- *The parameter lsa-md5-password must be configured with the same value as configured for the parameter lsa-md5-password of the LSA server instance.*

DESCRIPTION FOR THE COMMAND PARAMETER rtp-ipv6-gateway:

- *The parameter allows to configure the IPv6 address of the media IP gateway.*
- *ISAM FD and ISAM FX equipment: A mandatory input parameter when (1) the parameter media-ipmode is configured as 'ipv6' and (2) the parameter ip-config-mode is configured as 'manual' and (3) the parameter ip-addr-policy is configured as 'distributed' and (4) the distinct VLAN approach applies. In case of the shared VLAN approach, the parameter rtp-ipv6-gateway inherits the value configured for the parameter signal-gateway-ip.*
- *NGVR and ANSI MDU equipment: A mandatory input parameter when (1) the parameter media-ipmode is configured as 'ipv6' and (2) the parameter ip-config-mode is configured as 'manual' and (3) the distinct VLAN approach applies. In case of the shared VLAN approach, the parameter rtp-Ipv6-gateway inherits the value configured for the parameter signal-gateway-ip.*

DESCRIPTION FOR THE COMMAND PARAMETER rtp-ipv6-link-mtu:

- *The parameter allows to configure the link mtu of the IPv6 media vlan.*
- *This parameter is only relevant when (1) the parameter media-ipmode is configured as 'ipv6' and (2) the distinct signaling/media VLAN approach applies. In case of the shared signaling/media VLAN approach, the parameter rtp-ipv6-link-mtu inherits the value configured for the parameter signal-link-mtu.*

The parameters rtp-ipv6-gateway, rtp-ipv6-vlan-id, rtp-ipv6-dscp, rtp-ipv6-pbits, rtp-ipv6-link-mtu are defined for *future usage only and thus actually not supported. IPv6 is actually not supported. All references to IPv6 must be ignored since mentioned for future usage only.*

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no user-agent (name)) | (user-agent (name) [no ip-addr-policy | ip-addr-policy <Sip::IpAddrPolicy>] [no ip-config-mode | ip-config-mode <Sip::IpConfigMode>] [no signal-gateway-ip | signal-gateway-ip <Sip::UserAgentIpAddr>] signal-vlan-id <Sip::UserAgentVlanId> [no signal-dscp | signal-dscp <Sip::UserAgentSignalDscp>] [no signal-pbits | signal-pbits <Sip::UserAgentSignalPbits>] [no signal-link-mtu | signal-link-mtu <Sip::UserAgentLinkMTU>] [no rtp-gateway-ip | rtp-gateway-ip <Sip::UserAgentIPv4Addr>] [no rtp-vlan-id | rtp-vlan-id <Sip::UserAgentRtpVlanId>] [no rtp-dscp | rtp-dscp <Sip::UserAgentRtpDscp>] [no rtp-pbits | rtp-pbits <Sip::UserAgentRtpPbits>] [no rtp-link-mtu | rtp-link-mtu <Sip::UserAgentLinkMTU>] [no dhcp-optreq-list | dhcp-optreq-list <Sip::UserAgentDHCPOptionReqList>] [no admin-status | admin-status <Sip::UserAgentAdminStatus>] [no provider-name | provider-name <Sip::UserAgentProviderName>] [no lsa-rtp-gw-ip | lsa-rtp-gw-ip <Sip::UserAgentIPv4Addr>] [no lsa-rtp-vlan-id | lsa-rtp-vlan-id <Sip::UALSAVlanId>] [no lsa-md5-realm | lsa-md5-realm <Sip::MD5Realm>] [no lsa-md5-password | lsa-md5-password <Security::Password4>] [no rtp-ipv6-gateway | rtp-ipv6-gateway <Sip::UserAgentIPv6Addr>] [no rtp-ipv6-vlan-id | rtp-ipv6-vlan-id <Sip::UserAgentRtpVlanId>] [no rtp-ipv6-dscp | rtp-ipv6-dscp <Sip::UserAgentRtpDscp>] [no rtp-ipv6-pbits | rtp-ipv6-pbits <Sip::UserAgentRtpPbits>] [no rtp-ipv6-link-mtu | rtp-ipv6-link-mtu <Sip::UserAgentV6LinkMTU>])

Command Parameters

Table 40.4-1 "Voice Sip User Agent Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(name)	Format: - uniquely identifies the User Agent - length: 1<=x<=32	uniquely identify of this user agent

Table 40.4-2 "Voice Sip User Agent Configuration Commands" Command Parameters

Parameter	Type	Description
[no] ip-addr-policy	Parameter type: <Sip::IpAddrPolicy> Format: (distributed | centralized) Possible values: - distributed : Each of the voice LT boards owns (a) different IP address(es) for the signaling and media plane. - centralized : All voice LT boards shared the same IP address(es) for the signaling and media plane.	*optional parameter with default value: "distributed"* the internal ip address policy of the voice node
[no] ip-config-mode	Parameter type: <Sip::IpConfigMode> Format: (dhcp | manual) Possible values: - dhcp : The IP related data is acquired from the DHCP server. - manual : The IP related data must be manually configured.	*optional parameter with default value: "dhcp"* the configuration mode of the IP related data.

Parameter	Type	Description
[no] signal-gateway-ip	Parameter type: <Sip::UserAgentIpAddr> Format: (ipv6 : <Ip::V6Address> | <Ip::V4Address>) Possible values: - ipv6 : IPv6-address Field type <Ip::V6Address> - IPv6-address Field type <Ip::V4Address> - IPv4-address	*optional parameter with default value: "0.0.0.0"* The IPv4 or IPv6 address of the signalling IP gateway.
signal-vlan-id	Parameter type: <Sip::UserAgentVlanId> Format: - vlan id - range: [1...4093]	*mandatory parameter* the vlan id for IPv4 or IPv6 signaling traffic
[no] signal-dscp	Parameter type: <Sip::UserAgentSignalDscp> Format: - dscp mark for rtp or rtcp packets - range: [0...63]	*optional parameter with default value: 46* the dscp mark for the transmitted sip IPv4 or IPv6 signaling traffic
[no] signal-pbits	Parameter type: <Sip::UserAgentSignalPbits> Format: - the dot-1p bit value - range: [0...7]	*optional parameter with default value: 6* the 802.1p bits for the transmitted sip IPv4 or IPv6 signaling traffic
[no] signal-link-mtu	Parameter type: <Sip::UserAgentLinkMTU> Format: - minimum value: 576 for IPv4 - range: [576...1500]	*optional parameter with default value: 1500* the linkmtu for the IPv4 or IPv6 signaling vlan
[no] rtp-gateway-ip	Parameter type: <Sip::UserAgentIPv4Addr> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* IPv4 address of the media IP gateway.
[no] rtp-vlan-id	Parameter type: <Sip::UserAgentRtpVlanId> Format: - rtp vlan id - range: [0...4093]	*optional parameter with default value: 0* vlan id for IPv4 media traffic
[no] rtp-dscp	Parameter type: <Sip::UserAgentRtpDscp> Format: - dscp mark for rtp or rtcp packets - range: [0...63]	*optional parameter with default value: 46* dscp mark for transmitted IPv4 rtp or rtcp traffic
[no] rtp-pbits	Parameter type: <Sip::UserAgentRtpPbits> Format: - the dot-1p bit value - range: [0...7]	*optional parameter with default value: 6* 802.1p bits for transmitted IPv4 rtp or rtcp traffic
[no] rtp-link-mtu	Parameter type: <Sip::UserAgentLinkMTU> Format: - minimum value: 576 for IPv4 - range: [576...1500]	*optional parameter with default value: 1500* link mtu of the IPv4 media vlan.
[no] dhcp-optreq-list	Parameter type: <Sip::UserAgentDHCPOptionReqList> Format: - the list of DHCPv4 options to be returned in option 55. - length: x<=63	*optional parameter with default value: "1,3,6,120"* the list of DHCPv4 options to be returned in option 55.
[no] admin-status	Parameter type: <Sip::UserAgentAdminStatus> Format: (up | down) Possible values: - up : unlock the sip user agent - down : lock the sip user agent	*optional parameter with default value: "down"* administrative status of this sip user agent

Parameter	Type	Description
[no] provider-name	Parameter type: <Sip::UserAgentProviderName> Format: - uniquely identifies the user agent provider name - length: x<=32	*optional parameter with default value: "vsp1"* the provider name
[no] lsa-rtp-gw-ip	Parameter type: <Sip::UserAgentIPv4Addr> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* media gateway IP address for the LSA mode
[no] lsa-rtp-vlan-id	Parameter type: <Sip::UALSAVlanId> Format: - lsa vlan id - range: [0...4093]	*optional parameter with default value: 0* media vlan ID for the LSA mode
[no] lsa-md5-realm	Parameter type: <Sip::MD5Realm> Format: - the realm identifier (Due to legacy reasons, the MD5 character string was included in the object name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication realm identifier.) - length: x<=64	*optional parameter with default value: ""* realm for the registration of SIP terminations to the LSA server.
[no] lsa-md5-password	Parameter type: <Security::Password4> Format: (prompt | plain : <Security::PlainPassword4>) Possible values: - prompt : prompts the operator for a password - plain : the password in plain text, may contain any printable character defined by the US-ASCII character set and that has a decimal code in the range 33 to 126, and the length should be less than or equal to 64 (Due to legacy reasons, the MD5 character string was included in the parameter name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication password.) Field type <Security::PlainPassword4> - the password, may contain any printable character defined by the US-ASCII character set and that has a decimal code in the range 33 to 126, and the length should be less than or equal to 64 (Due to legacy reasons, the MD5 character string was included in the parameter name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication password.) - length: x<=64	*optional parameter with default value: "plain : "* password for the registration of SIP terminations to the LSA server.
[no] rtp-ipv6-gateway	Parameter type: <Sip::UserAgentIPv6Addr> Format: - IPv6-address	*optional parameter with default value: " : : "* IPv6 address of the media IP gateway.
[no] rtp-ipv6-vlan-id	Parameter type: <Sip::UserAgentRtpVlanId> Format: - rtp vlan id - range: [0...4093]	*optional parameter with default value: 0* vlan id for IPv6 media traffic

Parameter	Type	Description
[no] rtp-ipv6-dscp	Parameter type: <Sip::UserAgentRtpDscp> Format: - dscp mark for rtp or rtcp packets - range: [0...63]	*optional parameter with default value: 46* dscp mark for the transmitted IPv6 rtp and rtcp traffic
[no] rtp-ipv6-pbits	Parameter type: <Sip::UserAgentRtpPbits> Format: - the dot-1p bit value - range: [0...7]	*optional parameter with default value: 6* 802.1p bits for the transmitted IPv6 rtp and rtcp traffic
[no] rtp-ipv6-link-mtu	Parameter type: <Sip::UserAgentV6LinkMTU> Format: - minimum value: 1280 for IPv6 - range: [1280...1500]	*optional parameter with default value: 1500* linkmtu of the IPv6 media vlan

40.5 Sip Voice Service Gateway Configuration Commands

Command Description

This command allows the operator to configure the Voice Service Gateway.

The parameters signaling-ipmode and media-ipmode are defined for future usage only and thus actually not supported. IPv6 is actually not supported. All references to IPv6 must be ignored since mentioned for future usage only.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no vsp (name)) | (vsp (name) domain-name <Sip::VspDomainName> [no admin-status | admin-status <Sip::VspAdminStatus>] [no tinfo | tinfo <Sip::VspTinfo>] [no ta4 | ta4 <Sip::VspTa4>] [no ttir1 | ttir1 <Sip::VspTtir1>] [no t-acm-delta | t-acm-delta <Sip::VspTAcmDelta>] [no access-held-time | access-held-time <Sip::VSPAccessHeldTimer>] [no awaiting-time | awaiting-time <Sip::VSPAwaitingTimer>] [no digit-send-mode | digit-send-mode <Sip::VSPDigitSendingMode>] [no overlap-484-act | overlap-484-act <Sip::VSPOverlap484Action>] [no dmpm-intdgt-expid | dmpm-intdgt-expid <Sip::VSPDMPMIntDgtExpid>] [no dial-start-timer | dial-start-timer <Sip::VSPDialStartTimer>] [no dial-long-timer | dial-long-timer <Sip::VSPDialLongTimer>] [no dial-short-timer | dial-short-timer <Sip::VSPDialShortTimer>] [no uri-type | uri-type <Sip::VSPURIType>] [no rfc2833-pl-type | rfc2833-pl-type <Sip::VSPRfc2833PayloadType>] [no rfc2833-process | rfc2833-process <Sip::VSPRfc2833Process>] [no min-data-jitter | min-data-jitter <Sip::VSPDataJitter>] [no init-data-jitter | init-data-jitter <Sip::VSPDataJitter>] [no max-data-jitter | max-data-jitter <Sip::VSPDataJitter>] [no release-mode | release-mode <Sip::VSPReleaseMode>] [no dyn-pt-nego-type | dyn-pt-nego-type <Sip::VSPDynamicPTNegoType>] [no vbd-g711a-pl-type | vbd-g711a-pl-type <Sip::VSPVbdG711APayloadType>] [no vbd-g711u-pl-type | vbd-g711u-pl-type <Sip::VSPVbdG711UPayloadType>] [no vbd-mode | vbd-mode <Sip::VSPVbdMode>] [no warmline-dl-timer | warmline-dl-timer <Sip::VSPWarmlineDelayTimer>] [timer-b <Sip::CommonCfgTimerB>] [timer-f <Sip::CommonCfgTimerF>] [timer-t1 <Sip::CommonCfgTimerT1>] [timer-t2 <Sip::CommonCfgTimerT2>] [no reg-sub | reg-sub <Sip::VSPRegSubscribe>] [no dtmf-sip-info | dtmf-sip-info <Sip::VSPDtmfRelaySipInfo>] [no sub-period | sub-period <Sip::VSPSubscribePeriod>] [no sub-head-start | sub-head-start <Sip::VSPSubscribeHeadStart>] [no t38-same-udp | t38-same-udp <Sip::VSPT38withSameUDP>] [no dhcp-option82 | dhcp-option82 <Sip::DHCPOption82>] [no sspprofile | sspprofile <Sip::SSPProfile>] [no signaling-ipmode | signaling-ipmode <Sip::VSPSignalingIpMode>] [no tls-cafile | tls-cafile <Sip::VSPCAFileName>] [no media-ipmode | media-ipmode <Sip::VSPMediaIpMode>])

Command Parameters

Table 40.5-1 "Sip Voice Service Gateway Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(name)	Format: - uniquely identify of this SIP voice service gateway - length: x<=32	uniquely identify of this sip voice service gateway

Table 40.5-2 "Sip Voice Service Gateway Configuration Commands" Command Parameters

Parameter	Type	Description
domain-name	Parameter type: <Sip::VspDomainName> Format: - domain name of the SIP VSG - length: 1<=x<=64	*mandatory parameter* domain name of the sip VSG
[no] admin-status	Parameter type: <Sip::VspAdminStatus> Format: (up | down) Possible values: - up : unlock the sip VSG - down : lock the sip VSG	*optional parameter with default value: "down"* administrative status of this sip voice service gateway
[no] tinfo	Parameter type: <Sip::VspTinfo> Format: - tinfo of this SIP voice service gateway - range: [1...3600000]	*optional parameter with default value: 1000* value of tinfo in milli-second of this sip voice service gateway, the timer to collect digits before an INVITE is sent to reduce the amount of INVITE requests sent as defined in ETSI TS 183 036 V0.10.1
[no] ta4	Parameter type: <Sip::VspTa4> Format: - Ta4 timer expire of this SIP voice service gateway - unit: millisecond - range: [1...3600000]	*optional parameter with default value: 4000* value of Ta4 timer expire in milli-second. The timer Ta4 starts on receipt of first dialed digit. If the dialed digits are not matched with digitmap, the expiry of Ta4 will trigger the sending of initial INVITE.
[no] ttir1	Parameter type: <Sip::VspTtir1> Format: - ttir1 of this SIP voice service gateway - range: [1...3600000]	*optional parameter with default value: 100* value of ttir1 in milli-second of this sip voice service gateway, in the case 'from-change' is indicated, UA has to wait for an UPDATE, the timer assures that the call can continue if the UPDATE is missing as defined in ETSI TS 183 036 V0.10.1
[no] t-acm-delta	Parameter type: <Sip::VspTAcmDelta> Format: - T-acm-delta timer expire of this SIP voice service gateway - unit: millisecond - range: [1...3600000]	*optional parameter with default value: 18000* value of T-acm-delta timer expire in milli-second. The timer T-acm-delta starts on the situation that T-interdigit timer expires and there is INVITE transaction ongoing. The system will start T-acm-delta. The expiry of T-acm-delta will end the attemp of the call establishment.
[no] access-held-time	Parameter type: <Sip::VSPAccessHeldTimer>	*optional parameter with default*

Parameter	Type	Description
	Format: - Value of access-held-timer expires which is used in the release control procedure. - unit: second - range: [1...65535]	*value: 600* this value is used in the release control procedure. it is started when receiving INVITE with No Ring and no SDP and stopped when sending 200 OK (INVITE).
[no] awaiting-time	Parameter type: <Sip::VSPAwaitingTimer> Format: - value of awaiting-timer expires which is used in the release control procedure. - unit: second - range: [1...65535]	*optional parameter with default value: 60* this value is used in the release control procedure. it is started when receiving 200 OK (Bye) with X-Service-indicator and stopped when receiving INVITE with No Ring and No SDP.
[no] digit-send-mode	Parameter type: <Sip::VSPDigitSendingMode> Format: (en-bloc | overlap-invite | overlap-indialog) Possible values: - en-bloc : en-block mode - overlap-invite : overlap-invite mode - overlap-indialog : overlap-indialog mode	*optional parameter with default value: "en-bloc"* digit sending mode of this sip voice service gateway
[no] overlap-484-act	Parameter type: <Sip::VSPOverlap484Action> Format: (release-call | continue) Possible values: - release-call : terminate the call. - continue : continue the digit collection for the call attempt.	*optional parameter with default value: "release-call"* release call or not when 484 is received for INVITE before early dialog is established for digit sending in overlap dialing in-dialog method.
[no] dmpm-intdgt-expid	Parameter type: <Sip::VSPDMPMIntDgtExpid> Format: (send-invite | release-call) Possible values: - send-invite : send out invite with the collected digits. - release-call : terminate the call.	*optional parameter with default value: "send-invite"* The action if inter-digit timer expired when digitmap partial matched.
[no] dial-start-timer	Parameter type: <Sip::VSPDialStartTimer> Format: - Dialing start timer in second - unit: second - range: [1...60]	*optional parameter with default value: 10* Default maximum waiting time in seconds for dialing the first digit
[no] dial-long-timer	Parameter type: <Sip::VSPDialLongTimer> Format: - dialing long timer in second - unit: second - range: [1...60]	*optional parameter with default value: 20* Default maximum waiting time in seconds for dialing when no matching found in the digitmap
[no] dial-short-timer	Parameter type: <Sip::VSPDialShortTimer> Format: - dialing short timer in second - unit: second - range: [1...60]	*optional parameter with default value: 5* Default maximum waiting time in seconds for dialing when matching found in the digitmap
[no] uri-type	Parameter type: <Sip::VSPURIType> Format: (sip-uri | tel-uri | sips-uri)	*optional parameter with default value: "sip-uri"* uri type of this sip voice service gateway

Parameter	Type	Description
	Possible values: - sip-uri : sip-uri mode - tel-uri : tel-uri mode - sips-uri : sips-uri mode	
[no] rfc2833-pl-type	Parameter type: <Sip::VSPRfc2833PayloadType> Format: - payload type of rfc2833 - range: [96...127]	*optional parameter with default value: 96* payload type of rfc2833
[no] rfc2833-process	Parameter type: <Sip::VSPRfc2833Process> Format: (audio | rfc2833 | both | mandatory-audio) Possible values: - audio : processing procedure is audio - rfc2833 : processing procedure is rfc2833 - both : processing procedure is both - mandatory-audio : processing procedure is audio first	*optional parameter with default value: "audio"* processing procedure for the dtmf event tones
[no] min-data-jitter	Parameter type: <Sip::VSPDataJitter> Format: - data jitter buffer for calls working in rtp mode in millisecond - unit: millisecond - range: [0...200]	*optional parameter with default value: 50* minimum jitter buffer for data calls working in rtp mode
[no] init-data-jitter	Parameter type: <Sip::VSPDataJitter> Format: - data jitter buffer for calls working in rtp mode in millisecond - unit: millisecond - range: [0...200]	*optional parameter with default value: 50* initial jitter buffer for data calls working in rtp mode
[no] max-data-jitter	Parameter type: <Sip::VSPDataJitter> Format: - data jitter buffer for calls working in rtp mode in millisecond - unit: millisecond - range: [0...200]	*optional parameter with default value: 50* maximum jitter buffer for data calls working in rtp mode
[no] release-mode	Parameter type: <Sip::VSPReleaseMode> Format: (normal | caller | callee | both) Possible values: - normal : sending BYE immediately, applicable for the normal call release procedure - caller : postpone BYE until access-held-time times out for caller, only postpone BYE in case the subscriber is caller - callee : postpone BYE until access-held-time times out for callee, only postpone BYE in case the subscriber is callee - both : postpone BYE until access-held-time times out for caller and callee, postpone BYE in case the subscriber is caller or callee	*optional parameter with default value: "normal"* send BYE immediately or not after caller/callee/both onhook
[no] dyn-pt-nego-type	Parameter type: <Sip::VSPDynamicPTNegoType> Format: (far-end | asymmetric)	*optional parameter with default value: "far-end"* how to negotiate dynamic payload type

Parameter	Type	Description
	Possible values: - far-end : negotiate with supported by far-end - asymmetric : negotiate with different payload type for sending and receiving	
[no] vbd-g711a-pl-type	Parameter type: <Sip::VSPVbdG711APayloadType> Format: - dynamic payload type of Vbd G711A - range: [96...127]	*optional parameter with default value: 97* dynamic payload type of vbd g711a
[no] vbd-g711u-pl-type	Parameter type: <Sip::VSPVbdG711UPayloadType> Format: - dynamic payload type of Vbd G711U - range: [96...127]	*optional parameter with default value: 98* dynamic payload type of vbd g711u
[no] vbd-mode	Parameter type: <Sip::VSPVbdMode> Format: (autoswitch | renegotiation | v152fb-autoswitch | v152fb-reneg) Possible values: - autoswitch : auto switch without negotiation - renegotiation : renegotiation - v152fb-autoswitch : V152. If V152 negotiation fails, act as autoswitch (ISDN and CASR2 not support this) - v152fb-reneg : V152. If V152 negotiation fails, act as renegotiation (ISDN and CASR2 not support this)	*optional parameter with default value: "autoswitch"* vbd negotiation mode
[no] warmline-dl-timer	Parameter type: <Sip::VSPWarmlineDelayTimer> Format: - warmline delay timer in second - unit: second - range: [1...60]	*optional parameter with default value: 4* warmline-delay-timer,range from 1 to 60 sec
timer-b	Parameter type: <Sip::CommonCfgTimerB> Format: - value of SIP CommonCfgTimerB for rfc4780 - unit: millisecond - range: [2000...64000]	*optional parameter* This object reflects the maximum time a SIP entity will wait to receive a response to an INVITE. The timer is started upon transmission of the initial INVITE request.
timer-f	Parameter type: <Sip::CommonCfgTimerF> Format: - value of SIP CommonCfgTimerF for rfc4780 - unit: millisecond - range: [2000...64000]	*optional parameter* This object reflects the maximum time a SIP entity will wait to receive a final response to a non-INVITE request. The timer is started upon transmission of the initial request.
timer-t1	Parameter type: <Sip::CommonCfgTimerT1> Format: - value of SIP CommonCfgTimerT1 for rfc4780 - unit: millisecond - range: [200...10000]	*optional parameter* This object reflects the T1 timer for a SIP entity. T1 is an estimate of the round-trip time (RTT) between the client and server transactions.
timer-t2	Parameter type: <Sip::CommonCfgTimerT2> Format: - value of SIP CommonCfgTimerT2 for rfc4780 - unit: millisecond - range: [200...10000]	*optional parameter* This object reflects the T2 timer for a SIP entity. T2 is the maximum retransmit interval for non-INVITE requests and INVITE responses. It is used in various parts of the protocol to

Parameter	Type	Description
		reset other Timer* objects to this value.
[no] reg-sub	Parameter type: <Sip::VSPRegSubscribe> Format: (disable | enable) Possible values: - disable : do not send reg SUBSCRIBE - enable : send reg SUBSCRIBE	*optional parameter with default value: "disable"* This object indicates whether the system should send the reg SUBSCRIBE method.
[no] dtmf-sip-info	Parameter type: <Sip::VSPDtmfRelaySipInfo> Format: (disable | enable) Possible values: - disable : do not send DTMF Info - enable : send DTMF Info	*optional parameter with default value: "disable"* This objects indicates whether dtmf-events will be sent on top of RFC2833 configuration also in the SIP signalling path in the message-body of SIP INFO message with content-type set to application/dtmf-relay.
[no] sub-period	Parameter type: <Sip::VSPSubscribePeriod> Format: - This object indicates subscription expiration time that the client will propose by including it in an Expires header of a SUBSCRIBE request. If this property is set to a value of zero, the client SHALL NOT refresh a subscription even if the server specifies an expiration interval. Accept value greater than or equal to 60 [0|(60..86400)] - unit: second - range: [0,60...86400]	*optional parameter with default value: 3600* This object indicates subscription expiration time that the client will propose by including it in an Expires header of a SUBSCRIBE request. If this property is set to a value of zero, the client SHALL NOT refresh a subscription even if the server specifies an expiration interval.
[no] sub-head-start	Parameter type: <Sip::VSPSubscribeHeadStart> Format: - This object indicates number of seconds prior to expiration of a subscription at which the client sends a SUBSCRIBE request to refresh the subscription. Accept value greater than or equal to 60 [0|(32..86400)] - unit: second - range: [0,32...86400]	*optional parameter with default value: 600* This object indicates number of seconds prior to expiration of a subscription at which the client sends a SUBSCRIBE request to refresh the subscription.
[no] t38-same-udp	Parameter type: <Sip::VSPT38withSameUDP> Format: (disable | enable) Possible values: - disable : T38 use different port as voice - enable : T38 use same port as voice	*optional parameter with default value: "disable"* Whether T38 use same UDP port with voice,disable(1),enable(2)
[no] dhcp-option82	Parameter type: <Sip::DHCPOption82> Format: (enable | disable) Possible values: - enable : enable to send option 82 in DHCP mode - disable : disable to send option 82 in DHCP mode	*optional parameter with default value: "disable"* dhcp option82 mode
[no] sspprofile	Parameter type: <Sip::SSPProfile> Format: - The name of the SIP Service Profile assigned to a voice service gateway - length: x<=64	*optional parameter with default value: ""* The name of the SIP Service Profile assigned to a VSG
[no] signaling-ipmode	Parameter type: <Sip::VSPSignalingIpMode>	*optional parameter with default*

Parameter	Type	Description
	Format: (legacy-ipv4 | ipv4 | ipv6) Possible values: - legacy-ipv4 : the SIP signaling path is based on IPv4 addressing, for backward compatibility of legacy CLI management interface. Only allowed when media-ipmode set as 'legacy-ipv4' - ipv4 : the SIP signaling path is based on IPv4 addressing, for changed CLI management interface. Only allowed when media-ipmode set as 'ipv4' - ipv6 : the SIP signaling path is based on IPv6 addressing, for changed CLI management interface. Only allowed when media-ipmode set as 'ipv6'	*value: "legacy-ipv4"* The variable allows the operator to specify the IP mode of the SIP signaling path.
[no] tls-cafile	Parameter type: <Sip::VSPCAFileName> Format: - the ca file name - length: x<=13	*optional parameter with default value: ""* The name of Certification Authority (CA) file configured for VSP entry.
[no] media-ipmode	Parameter type: <Sip::VSPMediaIpMode> Format: (legacy-ipv4 | ipv4 | ipv6) Possible values: - legacy-ipv4 : the media path is based on IPv4 addressing, for backward compatibility of legacy CLI management interface. Only allowed when signaling-ipmode set as 'legacy-ipv4' - ipv4 : the media path is based on IPv4 addressing, for changed CLI management interface. Only allowed when signaling-ipmode set as 'ipv4' - ipv6 : the media path is based on IPv6 addressing, for changed CLI management interface. Only allowed when signaling-ipmode set as 'ipv6'	*optional parameter with default value: "legacy-ipv4"* The variable allows the operator to specify the IP mode of the SIP media path.

40.6 Voice Sip User Agent Access Point Configuration Command

Command Description

This command allows the operator to configure the Voice Sip User Agent Access Point. This command depends on the Voice Sip UserAgent Configuration command. When a SIP User Agent is created/deleted, the user agent access point(s) will be created/deleted automatically according to the SIP UA ip-address-policy. The user agent access point(s) can be learned using the show command: show voice sip user-agent-ap.

The parameter rtp-ipv6-address is defined for future usage only and thus actually not supported. IPv6 is actually not supported. All references to IPv6 must be ignored since mentioned for future usage only.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip user-agent-ap (ua-name) slot-id <SIP::SlotIndex> [signal-ip <SIP::IpAddressWithPrefix>] [no rtp-ip | rtp-ip <SIP::IpAddressAndMask>] [no dhcp-fqdn | dhcp-fqdn <Sip::UaApDHCPFQDN>] [no dhcp-customer-id | dhcp-customer-id <Sip::UaApDHCPCustomerID>] [no admin-status | admin-status <Sip::UaApAdminStatus>] [no lsa-rtp-ip | lsa-rtp-ip <SIP::IpAddressAndMask>] [no rtp-ipv6-address | rtp-ipv6-address <SIP::IPv6AddressWithPrefix>]

Command Parameters

Table 40.6-1 "Voice Sip User Agent Access Point Configuration Command" Resource Parameters

Resource Identifier	Type	Description
(ua-name)	Format: - uniquely identifies the User Agent - length: 1<=x<=32	uniquely identify of the user agent
slot-id	Parameter type: <SIP::SlotIndex> Format: (lt : <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> | nt | ntio | <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::EqSlotId>) Possible values: - lt : lt-slot - nt : nt-slot - ntio : ntio-slot Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId>	slot id associated with this user agent access point

Resource Identifier	Type	Description
	- the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::EqSlotId> - the equipment slot number	

Table 40.6-2 "Voice Sip User Agent Access Point Configuration Command" Command Parameters

Parameter	Type	Description
signal-ip	Parameter type: <SIP::IpAddressWithPrefix> Format: (ipv6 : <Ip::V6Address> / <Ip::V6PrefixLength> | <Ip::V4Address> / <Sip::PrefixLength>) Possible values: - ipv6 : IPv6-address Field type <Ip::V6Address> - IPv6-address Field type <Ip::V4Address> - IPv4-address Field type <Ip::V6PrefixLength> - prefix length of IPv6-address - range: [0...128] Field type <Sip::PrefixLength> - prefix length of the subnet - range: [0...128]	*optional parameter* signalling ip address and prefix length of the sip user agent access point
[no] rtp-ip	Parameter type: <SIP::IpAddressAndMask> Format: <Ip::V4Address> / <Sip::PrefixLength> Field type <Ip::V4Address> - IPv4-address Field type <Sip::PrefixLength> - prefix length of the subnet - range: [0...128]	*optional parameter with default value: "0.0.0.0/0"* rtp ip address and prefix length of the sip user agent access point
[no] dhcp-fqdn	Parameter type: <Sip::UaApDHCPFQDN> Format: - the string for FQDN in Option81 When UA send out DHCP request - length: x<=255	*optional parameter with default value: ""* the FQDN in Option81 When UA send out DHCP request
[no] dhcp-customer-id	Parameter type: <Sip::UaApDHCPCustomerID> Format: - the string for Customer id in Option82 When UA send out DHCP request - length: x<=64	*optional parameter with default value: "Physical-ID"* the Customer id in Option82 When UA send out DHCP request
[no] admin-status	Parameter type: <Sip::UaApAdminStatus> Format: (up | down) Possible values: - up : unlock the sip Ua AccessPoint - down : lock the sip Ua AccessPoint	*optional parameter with default value: "down"* administrative status of this sip user agent access point
[no] lsa-rtp-ip	Parameter type: <SIP::IpAddressAndMask> Format: <Ip::V4Address> / <Sip::PrefixLength> Field type <Ip::V4Address> - IPv4-address Field type <Sip::PrefixLength> - prefix length of the subnet - range: [0...128]	*optional parameter with default value: "0.0.0.0/0"* media IP address + prefix length used when connected to the LSA server

Parameter	Type	Description
[no] rtp-ipv6-address	Parameter type: <SIP::IPv6AddressWithPrefix> Format: <Ip::V6Address> / <Ip::V6PrefixLength> Field type <Ip::V6Address> - IPv6-address Field type <Ip::V6PrefixLength> - prefix length of IPv6-address - range: [0...128]	*optional parameter with default value: " : : /0"* rtp IPv6 address and prefix length of the sip user agent access point

40.7 Voice Sip Dial Plan Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip dial plan profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no dialplan (name)) | (dialplan (name) [no pre-activated | pre-activated <Sip::DialPlanPreActivated>] [no static-prefix | static-prefix <Sip::DialPlanStaticPrefix>] [no static-suffix | static-suffix <Sip::DialPlanStaticSuffix>] [no digitmap-mode | digitmap-mode <Sip::DialPlanDigitMapMode>] [no provider-name | provider-name <Sip::DialPlanProviderName>])

Command Parameters

Table 40.7-1 "Voice Sip Dial Plan Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(name)	Format: - uniquely identifies the dial plan - length: 1<=x<=32	unique identity of a voice sip dial plan

Table 40.7-2 "Voice Sip Dial Plan Configuration Commands" Command Parameters

Parameter	Type	Description
[no] pre-activated	Parameter type: <Sip::DialPlanPreActivated> Format: (on | off) Possible values: - on : Prefix activated of dial plan is on - off : Prefix activated of dial plan is off	*optional parameter with default value: "off"* identify the status for sip dial plan prefix
[no] static-prefix	Parameter type: <Sip::DialPlanStaticPrefix> Format: - static prefix added to all valid DNs - length: x<=32	*optional parameter with default value: ""* identify static prefix(uri type + area code) added to valid DNs.If the format is 'uri type + area code',the uri type will overwrite the uri type in vsp; If the format is area code without uri type, uri type will depend on the uri type in vsp.
[no] static-suffix	Parameter type: <Sip::DialPlanStaticSuffix> Format: - static suffix added to all valid DNs	*optional parameter with default value: ""* identify static suffix(area code)

Parameter	Type	Description
	- length: x<=32	added all valid DNs
[no] digitmap-mode	Parameter type: <Sip::DialPlanDigitMapMode> Format: (maximum | minimum) Possible values: - maximum : standard match mode - minimum : system should send out INVITE immediately when an exact full match dial string is detected	*optional parameter with default value: "maximum"* identify the digit match mode for dial plan
[no] provider-name	Parameter type: <Sip::DialPlanProviderName> Format: - uniquely identifies the dial plan provider name - length: x<=32	*optional parameter with default value: "vsp1"* identify the sip dial plan provider name

40.8 Voice Sip Dial Plan Digitmap Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip dialplan digitmap profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no digitmap (name) type <Sip::DialPlanDigitmapType> rule <Sip::DialPlanDigitmapValue>) | (digitmap (name) type <Sip::DialPlanDigitmapType> rule <Sip::DialPlanDigitmapValue> [no access-type | access-type <Sip::DialPlanDigitmapAccessType>])

Command Parameters

Table 40.8-1 "Voice Sip Dial Plan Digitmap Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(name)	Format: - uniquely identifies the dial plan - length: 1<=x<=32	identify voice application dial plan within the scope
type	Parameter type: <Sip::DialPlanDigitmapType> Format: regular Possible values: - regular : digitmap type	identify the type of sip dial plan digitmap
rule	Parameter type: <Sip::DialPlanDigitmapValue> Format: - identity the value of digitmap - length: 1<=x<=100	identify the value of sip dial plan digitmap

Table 40.8-2 "Voice Sip Dial Plan Digitmap Configuration Commands" Command Parameters

Parameter	Type	Description
[no] access-type	Parameter type: <Sip::DialPlanDigitmapAccessType> Format: allowed Possible values: - allowed : the digitmap can be used	*optional parameter with default value: "allowed"* *The parameter is not visible during creation.* The access type of digitmap

40.9 Voice Sip Termination Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip termination profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no termination (if-index)) | (termination (if-index) [no directory-number | directory-number <Sip::TermDnumber>] [no user-name | user-name <Sip::TermUserName>] [no display-name | display-name <Sip::TermDisplayName>] [no uri | uri <Sip::TermUri>] [no direct-uri | direct-uri <Sip::TermUri>] [no line-feed | line-feed <Sip::TermLineCharacter>] [no md5-realm | md5-realm <Sip::MD5Realm>] [no md5-password | md5-password <Security::Password4>] [no aka-secret-key | aka-secret-key <Security::Password8>] [no admin-status | admin-status <Sip::TermAdminStatus>] [no clip-mode | clip-mode <Sip::TermETSIClipDataMode>] [no telc-clip-mode | telc-clip-mode <Sip::TermTelcordiaClipDataMode>] [no anti-tapping | anti-tapping <Sip::TermAntiTapping>] [no impedance | impedance <Sip::TermImpedance>] [no rx-gain | rx-gain <Sip::TermRxGain>] [no tx-gain | tx-gain <Sip::TermTxGain>] [no warmline-service | warmline-service <Sip::TermWarmlineService>] [no linesig-remanswer | linesig-remanswer <Sip::TermLineSignalOnRemoteAnswer>] [no line-id | line-id <Sip::TermLineId>] [no force-hold | force-hold <Sip::TermService>] [no callwait-service | callwait-service <Sip::TermCallwaitingService>] [no callhold-service | callhold-service <Sip::TermSupplementaryService>] [no callconf-service | callconf-service <Sip::TermSupplementaryService>] [no calltras-service | calltras-service <Sip::TermSupplementaryService>] [no fast-guard | fast-guard <Sip::TermFastGuardOnInvite>] [no clear-forward | clear-forward <Sip::TermClearForwardOnBye>] [no testaccessstate | testaccessstate <Sip::TestAccessState>] [no busyoverwrite | busyoverwrite <Sip::TestAccessBusyOverWrite>] [no accessontimeout | accessontimeout <Sip::TestAccessonTimeout>] [no provider-name | provider-name <Sip::TermUserProviderName>] [no register_recall | register_recall <Sip::TermRegisterRecall>] [no sip-rule-set | sip-rule-set <Sip::PotsManipulationRuleSet>] [no reserved | reserved <Sip::TermReserved>] [no pulse-dialing | pulse-dialing <Sip::TermPulseDialing>] [no auto-answer | auto-answer <Sip::TermAutoAnswer>])

Command Parameters

Table 40.9-1 "Voice Sip Termination Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(if-index)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId>	The unique internal identifier of the termination port

Resource Identifier	Type	Description
	- the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::PortId> - the port number	

Table 40.9-2 "Voice Sip Termination Configuration Commands" Command Parameters

Parameter	Type	Description
[no] directory-number	Parameter type: <Sip::TermDnumber> Format: - identify the DN of the subscriber line - length: x<=32	*optional parameter with default value: ""* identify the directory (telephone) number of the subscriber line
[no] user-name	Parameter type: <Sip::TermUserName> Format: - identify the user name for this port - length: x<=64	*optional parameter with default value: ""* identify the user name for this termination port
[no] display-name	Parameter type: <Sip::TermDisplayName> Format: - identify the display name for this port - length: x<=64	*optional parameter with default value: ""* display name for this termination port
[no] uri	Parameter type: <Sip::TermUri> Format: - SIP uri by which the user agent identifies subscriber line - length: x<=80	*optional parameter with default value: ""* sip uri by which user agent identifies subscriber line
[no] direct-uri	Parameter type: <Sip::TermUri> Format: - SIP uri by which the user agent identifies subscriber line - length: x<=80	*optional parameter with default value: ""* a call will be automatically established to this SIP-URI after an off-hook
[no] line-feed	Parameter type: <Sip::TermLineCharacter> Format: (25 | 40) Possible values: - 25 : the line character of this port is 25, unit: ma - 40 : the line character of this port is 40, unit: ma	*optional parameter with default value: "25"* the characteristic of the subscriber line
[no] md5-realm	Parameter type: <Sip::MD5Realm> Format: - the realm identifier (Due to legacy reasons, the MD5 character string was included in the object name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication realm identifier.) - length: x<=64	*optional parameter with default value: ""* realm identifier corrsponding to service gateway domain
[no] md5-password	Parameter type: <Security::Password4> Format: (prompt | plain : <Security::PlainPassword4>) Possible values: - prompt : prompts the operator for a password - plain : the password in plain text, may contain any printable character defined by the US-ASCII character set and that has a decimal code in the range 33 to 126, and the length should be less than or equal to 64 (Due to legacy reasons, the MD5 character string was included in the	*optional parameter with default value: "plain : "* the password associated with the user

Parameter	Type	Description
	parameter name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication password.) Field type <Security::PlainPassword4> - the password, may contain any printable character defined by the US-ASCII character set and that has a decimal code in the range 33 to 126, and the length should be less than or equal to 64 (Due to legacy reasons, the MD5 character string was included in the parameter name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication password.) - length: x<=64	
[no] aka-secret-key	Parameter type: <Security::Password8> Format: (prompt | plain : <Security::PlainPassword8>) Possible values: - prompt : prompts the operator for AKA shared secret key - plain : the AKA shared secret key in plain text, each character should be 0-f, and the length can only be 0 or 32 Field type <Security::PlainPassword8> - the AKA shared secret key, each character should be 0-f, and the length can only be 0 or 32 - length: x<=32	*optional parameter with default value: "plain : "* the AKA shared secret key
[no] admin-status	Parameter type: <Sip::TermAdminStatus> Format: (up | down) Possible values: - up : unlock the sip termination - down : lock the sip termination	*optional parameter with default value: "down"* identify the status of this port administration
[no] clip-mode	Parameter type: <Sip::TermETSIClipDataMode> Format: (cdevalue | fsk | dtmf) Possible values: - cdevalue : cde configuration will be applied for etsi clip - fsk : fsk will be applied for etsi clip - dtmf : dtmf will be applied for etsi clip	*optional parameter with default value: "cdevalue"* the etsi clip data transmission protocol of this sip termination
[no] telc-clip-mode	Parameter type: <Sip::TermTelcordiaClipDataMode> Format: (cdevalue | fsk) Possible values: - cdevalue : cde configuration will be applied for Telcordia clip - fsk : fsk will be applied for Telcordia clip	*optional parameter with default value: "cdevalue"* the TELCORDIA clip data transmission protocol of this sip termination
[no] anti-tapping	Parameter type: <Sip::TermAntiTapping> Format: (enable | disable) Possible values:	*optional parameter with default value: "disable"* enable/disable the anti-tapping service of this sip termination

Parameter	Type	Description
	- enable : Enable Anti-Tapping service - disable : Disable Anti-Tapping service	
[no] impedance	Parameter type: <Sip::TermImpedance> Format: (default | 200 | 220 | 220minisplitter | 270 | 300 | 370 | 600 | 900 | 370minisplitter | 600splitter | 370nvlsasplitter | 370skinnysplitter | 270duratelsplitter | 2703msplitter | 100 | 150 | 220splitter | reserv19 | reserv20 | reserv21) Possible values: - default : the impedance is default value - 200 : the line impedance is 200 ohm - 220 : the line impedance is 220 ohm - 220minisplitter : the line impedance is 220 ohm+ miniSplitter - 270 : the line impedance is 270 ohm - 300 : the line impedance is 300 ohm - 370 : the line impedance is 370 ohm - 600 : the line impedance is 600 ohm - 900 : the line impedance is 900 ohm - 370minisplitter : the line impedance is 370 ohm+ miniSplitter - 600splitter : the line impedance is 600 ohm+splitter - 370nvlsasplitter : the line impedance is 370 ohm+ NVLSASplitter - 370skinnysplitter : the line impedance is 370 ohm+ SkinnySplitter - 270duratelsplitter : the line impedance is 270 ohm+ DuratelSplitter - 2703msplitter : the line impedance is 270 ohm+ 3MDplitter - 100 : the line impedance is 100 ohm - 150 : the line impedance is 150 ohm - 220splitter : the line impedance is 220 ohm+ splitter - reserv19 : reserved for future use 19 - reserv20 : reserved for future use 20 - reserv21 : reserved for future use 21	*optional parameter with default value: "default"* configure the line impedance of this sip termination
[no] rx-gain	Parameter type: <Sip::TermRxGain> Format: - the line rx_gain range -14...6 - range: [-14...6]	*optional parameter with default value: "0"* configure the line rx_gain of this sip termination

<table>
<tr><th>Parameter</th><th>Type</th><th>Description</th></tr>
<tr><td>[no] tx-gain</td><td>Parameter type: <Sip::TermTxGain>
Format:
- the line tx_gain range -14...6
- range: [-14...6]</td><td>optional parameter with default value: "0"
configure the line tx_gain of this sip termination</td></tr>
<tr><td>[no] warmline-service</td><td>Parameter type: <Sip::TermWarmlineService>
Format:
(disabled
| activated
| subscribed)
Possible values:
- disabled : When warmline-service=disabled and direct-uri is null, warmline service is controlled by SIP Service Profile. The SSP will tell whether this service is controlled by UA-Profile or the MIB disable value is applied (e.g. when the IMS core does not support the UA-profile approach).
- activated : Warmline is activated
- subscribed : Warmline can be subscribed by end user</td><td>optional parameter with default value: "disabled"
warmline administrative status</td></tr>
<tr><td>[no] linesig-remanswer</td><td>Parameter type: <Sip::TermLineSignalOnRemoteAnswer>
Format:
(no-signal
| line-reversal
| single-meteringpulse)
Possible values:
- no-signal : send no signal
- line-reversal : send line reversal signal
- single-meteringpulse : send single metering pulse signal</td><td>optional parameter with default value: "no-signal"
line signal on remote answer</td></tr>
<tr><td>[no] line-id</td><td>Parameter type: <Sip::TermLineId>
Format:
- The cli-timing (cli) and/or VBD rx-gain/tx-gain (vbdrx/ vbdtx) and/or the Fax Mode (fm) and/or the Voice Activity Detection (vad) and/or the warmline delay timer (wldt) and/or the processing procedure for the dtmf event tones(2833) and/or the vbd-mode (vm) and/or the UDP port usage for T38 to voice switching (and vice versa) (udp) and/or the Pani header line ID.\n Format :\n [cli:prior|cli:during][;][vbdtx:(value(-14:6))][;][vbdrx:(value(-14:6))][;][fm:(value(1:3))][;][vad:e|vad:d] header line-id value)]\n A semi-colon must be inserted between the configuration input strings. The configuration input must be given in lower case characters. No space allowed anywhere; no leading zero(es) allowed anywhere.\n In case the configuration input for one or multiple input strings must be changed, the configuration input for all other input strings must be repeated too i.e. the new configuration input for the line id parameter fully overwrites the previous configuration input.\n - Keyword : cli (Only applicable to CLIP ETSI standard.)\n Values : prior | during.\n prior: clip-procedure is priorDT (for FSK) or priorNoLR (for DTMF).\n during: clip-procedure is duringRinging.\n In case this keyword is not found in the configuration input of the line-id parameter, the pre-provisioned value in the CDE file applies.\n - Keyword : vbdtx \n Values : range [-14:6]\n The '+' sign is not allowed; only the '-' sign is allowed.\n In case this keyword is not found in the configuration input of the line-id parameter, for a voice call, the value configured at SIP termination level applies. After switching to VBD, the hardcoded value Tx gain = 0 applies.\n - Keyword : vbdrx\n</td><td>optional parameter with default value: ""
identify the line id for this termination port</td></tr>
</table>

Parameter	Type	Description
	Values : range [-14:6]\n The '+' sign is not allowed; only the '-' sign is allowed.\n In case this keyword is not found in the configuration input of the line-id parameter, for a voice call, the value configured at SIP termination level applies. After switching to VBD, the hardcoded value Rx gain = 0 applies.\n - Keyword : fm\n Values : range [1:3]\n 1 : g711vbd \n 2 : t.38-with-g711vbd-fallback-with-signaling \n 3 : t.38-with-g711vbd-fallback-without-signaling\n In case this keyword is not found in the configuration input of the line-id parameter, the pre-provisioned value in the CDE file applies.\n - Keyword : vad\n Values : e | d\n e : vad is enabled (sending comfort noise packets during silent periods)\n d : vad is disabled (do not send comfort noise packets during silent periods)\n In case this keyword is not found in the configuration input of the line-id parameter, the pre-provisioned value in the CDE file applies.\n - Keyword : wldt\n Values : range [1:60] seconds\n In case this keyword is not found in the configuration input of the line-id parameter, the system wide configuration input applies.\n - Keyword : 2833\n Values : range [1:4]]\n 1 : the processing procedure is audio\n 2 : the processing procedure is rfc2833\n 3 : the processing procedure is both, audio and rfc2833\n 4 : audio has priority in the processing procedure\n In case this keyword is not found in the configuration input of the line-id parameter, the system wide configuration input applies.\n - Keyword : vm\n Values: range [1:4]\n 1 : auto switch without negotiation\n 2 : renegotiation\n 3 : If V152 negotiation fails, autoswitch applies\n 4 : If V152 negotiation fails, renegotiation applies\n In case this keyword is not found in the configuration input of the line-id parameter, the system wide configuration input applies.\n - Keyword : udp\n Values : e | d \n e : the same UDP port is used for Voice and T38\n d : a different UDP port is used for Voice and T38\n In case this keyword is not found in the configuration input of the line-id parameter, the system wide configuration input applies.\n - Pani header line id :\n must always appear at the end of the configuration string; The configuration input is not bounded to any format i.e. It is the responsibility of the customer to define the desired line ID format; There is no restriction on the character set being used - length: x<=63	
[no] force-hold	Parameter type: <Sip::TermService> Format: (disable | enable) Possible values: - disable : field is disabled - enable : field is enabled	*optional parameter with default value: "disable"* force hold functionality
[no] callwait-service	Parameter type: <Sip::TermCallwaitingService> Format: (disabled | activated | subscribed) Possible values: - disabled : Callwaiting is disabled - activated : Callwaiting is activated	*optional parameter with default value: "disabled"* callwaiting administrative status

Parameter	Type	Description
	- subscribed : Callwaiting can be subscribed by end user	
[no] callhold-service	Parameter type: <Sip::TermSupplementaryService> Format: (disabled | activated) Possible values: - disabled : the service is disabled - activated : the service is activated	*optional parameter with default value: "disabled"* callhold administrative status
[no] callconf-service	Parameter type: <Sip::TermSupplementaryService> Format: (disabled | activated) Possible values: - disabled : the service is disabled - activated : the service is activated	*optional parameter with default value: "disabled"* callconference administrative status
[no] calltras-service	Parameter type: <Sip::TermSupplementaryService> Format: (disabled | activated) Possible values: - disabled : the service is disabled - activated : the service is activated	*optional parameter with default value: "disabled"* calltransfer administrative status
[no] fast-guard	Parameter type: <Sip::TermFastGuardOnInvite> Format: (cde-profile | enabled | disabled) Possible values: - cde-profile : legacy behaviour that applies CDE profile settings - enabled : enable Fast Guard on incoming INVITE - disabled : disable Fast Guard on incoming INVITE	*optional parameter with default value: "cde-profile"* fast guard administrative status
[no] clear-forward	Parameter type: <Sip::TermClearForwardOnBye> Format: (sip-service-profile | enabled | disabled) Possible values: - sip-service-profile : legacy behaviour that applies SSP profile settings - enabled : enable Clear Forward on Bye - disabled : disable Clear Forward on Bye	*optional parameter with default value: "sip-service-profile"* clear forward administrative status
[no] testaccessstate	Parameter type: <Sip::TestAccessState> Format: (on | off) Possible values: - on : turn on TestAccessState - off : turn off TestAccessState	*optional parameter with default value: "off"* test access status
[no] busyoverwrite	Parameter type: <Sip::TestAccessBusyOverWrite> Format: (true | false) Possible values: - true : enable TestAccessBusyOverWrite parameter - false : disable TestAccessBusyOverWrite parameter	*optional parameter with default value: "false"* test access busyoverwrite
[no] accessontimeout	Parameter type: <Sip::TestAccessonTimeout>	*optional parameter with default*

Parameter	Type	Description
	Format: - value of test access on expires. - unit: second - range: [0...900]	*value: "900"* time to go until test access timeout. In case test access state is off, the value will be 0. In case test access state is on, this parameter is optional with default initial value 900. The value will start counting down, until value 0 and test access state will be off again.
[no] provider-name	Parameter type: <Sip::TermUserProviderName> Format: - identify the user VSG name for this port - length: x<=32	*optional parameter with default value: "vsp1"* identify the user VSG name for this termination port
[no] register_recall	Parameter type: <Sip::TermRegisterRecall> Format: - identify the register_recall value.\n Format: min-max. such as: 100-300, range for min and max is [30...2000]. CDE value will work if configure \"\" or no register_recall. Any other formats are all invalid. - unit: milliseconds - length: x<=16	*optional parameter with default value: ""* overrule register_recall and on_hook_timer in CDE for this port
[no] sip-rule-set	Parameter type: <Sip::PotsManipulationRuleSet> Format: - the POTS manipulation rule set, at most 32 rule ids can be configured - length: x<=32	*optional parameter with default value: ""* identify the rules this termination can apply
[no] reserved	Parameter type: <Sip::TermReserved> Format: - reserved for future usage for this port - length: x<=32	*optional parameter with default value: ""* reserved for future usage for this port
[no] pulse-dialing	Parameter type: <Sip::TermPulseDialing> Format: (disabled | enabled) Possible values: - disabled : the pulse dialing is disabled - enabled : the pulse dialing is enabled	*optional parameter with default value: "enabled"* Termination pulse dialing status
[no] auto-answer	Parameter type: <Sip::TermAutoAnswer> Format: (disable | enable) Possible values: - disable : The auto-answer feature is disabled - enable : The auto-answer feature is enabled	*optional parameter with default value: "disable"* This object allows to enable the auto-answer feature for this termination

40.10 Voice Sip Termination TCA Threshold Configuration Command

Command Description

This command allows the operator to enable/disable tca or configure the high/low threshold for jitter buffer fill level of the sip termination.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip termination (if-index) tca [[no] enable] [no high-jbfl | high-jbfl <Sip::JBFL>] [no low-jbfl | low-jbfl <Sip::JBFL>]

Command Parameters

Table 40.10-1 "Voice Sip Termination TCA Threshold Configuration Command" Resource Parameters

Resource Identifier	Type	Description
(if-index)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::PortId> - the port number	The unique internal identifier of the termination port

Table 40.10-2 "Voice Sip Termination TCA Threshold Configuration Command" Command Parameters

Parameter	Type	Description
[no] enable	Parameter type: boolean	*optional parameter* Enable the reporting of tca for sip termination
[no] high-jbfl	Parameter type: <Sip::JBFL> Format: - average of jitter buffer fill level percentage for the termination - unit: percentage	*optional parameter with default value: 90* the high threshould for jitter buffer fill level. it can not be smaller than low-jbfl. 100 mean

Parameter	Type	Description
	- range: [0...100]	that this parameter shall be omitted by the system.
[no] low-jbfl	Parameter type: <Sip::JBFL> Format: - average of jitter buffer fill level percentage for the termination - unit: percentage - range: [0...100]	*optional parameter with default value: 80* the low threshould for jitter buffer fill level. it can not be bigger than high-jbfl. 0 mean that this parameter shall be omitted by the system.

40.11 Voice Sip Shared Line Termination Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip shared line termination profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no sharedline (if-index)) | (sharedline (if-index) [no directory-number | directory-number <Sip::TermDnumber>] [no user-name | user-name <Sip::TermUserName>] [no display-name | display-name <Sip::TermUserName>] [no uri | uri <Sip::TermUri>] [no admin-status | admin-status <Sip::TermAdminStatus>] [no clip-mode | clip-mode <Sip::TermETSIClipDataMode>] [no callwait-service | callwait-service <Sip::TermCallwaitingService>] [no callhold-service | callhold-service <Sip::TermSupplementaryService>] [no callconf-service | callconf-service <Sip::TermSupplementaryService>] [no calltras-service | calltras-service <Sip::TermSupplementaryService>])

Command Parameters

Table 40.11-1 "Voice Sip Shared Line Termination Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(if-index)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::PortId> - the port number	The unique internal identifier of the termination port

Table 40.11-2 "Voice Sip Shared Line Termination Configuration Commands" Command Parameters

Parameter	Type	Description
[no] directory-number	Parameter type: <Sip::TermDnumber> Format: - identify the DN of the subscriber line - length: x<=32	*optional parameter with default value: ""* identify the directory (telephone) number of the subscriber line

Parameter	Type	Description
[no] user-name	Parameter type: <Sip::TermUserName> Format: - identify the user name for this port - length: x<=64	*optional parameter with default value: ""* identify the user name for this termination port
[no] display-name	Parameter type: <Sip::TermUserName> Format: - identify the user name for this port - length: x<=64	*optional parameter with default value: ""* display name for this termination port
[no] uri	Parameter type: <Sip::TermUri> Format: - SIP uri by which the user agent identifies subscriber line - length: x<=80	*optional parameter with default value: ""* sip uri by which user agent identifies subscriber line
[no] admin-status	Parameter type: <Sip::TermAdminStatus> Format: (up | down) Possible values: - up : unlock the sip termination - down : lock the sip termination	*optional parameter with default value: "down"* identify the status of this port administration
[no] clip-mode	Parameter type: <Sip::TermETSIClipDataMode> Format: (cdevalue | fsk | dtmf) Possible values: - cdevalue : cde configuration will be applied for etsi clip - fsk : fsk will be applied for etsi clip - dtmf : dtmf will be applied for etsi clip	*optional parameter with default value: "cdevalue"* the etsi clip data transmission protocol of this sip termination
[no] callwait-service	Parameter type: <Sip::TermCallwaitingService> Format: (disabled | activated | subscribed) Possible values: - disabled : Callwaiting is disabled - activated : Callwaiting is activated - subscribed : Callwaiting can be subscribed by end user	*optional parameter with default value: "disabled"* callwaiting administrative status
[no] callhold-service	Parameter type: <Sip::TermSupplementaryService> Format: (disabled | activated) Possible values: - disabled : the service is disabled - activated : the service is activated	*optional parameter with default value: "disabled"* callhold administrative status
[no] callconf-service	Parameter type: <Sip::TermSupplementaryService> Format: (disabled | activated) Possible values: - disabled : the service is disabled - activated : the service is activated	*optional parameter with default value: "disabled"* callconference administrative status
[no] calltras-service	Parameter type: <Sip::TermSupplementaryService> Format: (disabled | activated) Possible values: - disabled : the service is disabled	*optional parameter with default value: "disabled"* calltransfer administrative status

Parameter	Type	Description
	- activated : the service is activated	

40.12 Sip Voice Register Configuration Commands

Command Description

This command allows the operator to configure the SIP Voice Register parameters .

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip register (provider-name) [no register-uri | register-uri <Sip::RegisterUri>] [no register-intv | register-intv <Sip::RegisterIntv>] [no reg-retry-intv | reg-retry-intv <Sip::RegisterRetryIntv>] [no reg-prev-ava-intv | reg-prev-ava-intv <Sip::RegisterPrevAvaIntv>] [no reg-head-start | reg-head-start <Sip::RegisterHeadStart>] [no reg-start-min | reg-start-min <Sip::RegisterStartMin>] [no init-reg-delay | init-reg-delay <Sip::RegisterDelayInitialRegister>]

Command Parameters

Table 40.12-1 "Sip Voice Register Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(provider-name)	Format: - provider name - length: x<=32	uniquely identify register provider name

Table 40.12-2 "Sip Voice Register Configuration Commands" Command Parameters

Parameter	Type	Description
[no] register-uri	Parameter type: <Sip::RegisterUri> Format: - The registration URI to be used by all SIP terminations that have a service agreement with this SIP Voice Service Gateway - length: x<=80	*optional parameter with default value: ""* The registration URI to be used by all SIP terminations that have a service agreement with this SIP Voice Service Gateway.
[no] register-intv	Parameter type: <Sip::RegisterIntv> Format: - The registration expiration time that UA will propose in the Expires header of a REGISTER request, unless the value is 0. If the value is set to 0 the UA shall not refresh a registration even if the server specifies an expiration interval. Accept value greater than or equal to 60 [0|(60..86400)] - unit: second - range: [0,60...86400]	*optional parameter with default value: 3600* The registration expiration time that UA will propose in the Expires header of a REGISTER request, unless the value is 0. If the value is set to 0 the UA shall not refresh a registration even if the server specifies an expiration interval.
[no] reg-retry-intv	Parameter type: <Sip::RegisterRetryIntv> Format: - The interval between successive registration retries after a	*optional parameter with default value: 60* The interval between successive

Parameter	Type	Description
	failed registration. The value of 0 shall disable registration retry.The valid input would be [0,30..86400] - unit: second - range: [0,30...86400]	registration retries after a failed registration. The value of 0 shall disable registration retry.
[no] reg-prev-ava-intv	Parameter type: <Sip::RegisterPrevAvaIntv> Format: - The magnitude of time interval that must be awaited before the system is allowed to initiate another register request. - unit: millisecond - range: [30...32000]	*optional parameter with default value: 500* the magnitude of time interval that must be awaited before the system is allowed to initiate another register request.
[no] reg-head-start	Parameter type: <Sip::RegisterHeadStart> Format: - The time prior to expiration of a registration at which the UA sends a registration refresh. If the value of register_head_start is greater than (1 - start_min%) of the value for register_period, then the register_head_start value will be ignored and a timer equal to start_min% of the value for register_period will be used for this function,a value 0 means register_head_start will not be used - unit: second - range: [0...86400]	*optional parameter with default value: 600* The time prior to expiration of a registration at which the UA sends a registration refresh,if the value of register_head_start is greater than (1 - start_min%) of the value for register_period, then the register_head_start value will be ignored and a timer equal to start_min% of the value for register_period will be used for this function,a value 0 means, this value will not be used,valid input would be [0,32..86400]
[no] reg-start-min	Parameter type: <Sip::RegisterStartMin> Format: - The time prior to expiration of a registration at which a registration refresh is sent, if the value of register_head_start is greater than (1 - start_min%) of the value for the Expires value received in the 200 OK response to last REGISTER request for this line (either the Expires parameter in the Contact header or, if that is not present, in an Expires header), then the register_head_start value is ignored and a timer equal to start_min% of the value for register_period is used for this function - unit: percentage - range: [30...70]	*optional parameter with default value: 50* The time prior to expiration of a registration at which a registration refresh is sent. If the value of register_head_start is greater than (1 - start_min%) of the value for the Expires value received in the 200 OK response to last REGISTER request for this line (either the Expires parameter in the Contact header or, if that is not present, in an Expires header), then the register_head_start value is ignored and a timer equal to start_min% of the value for register_period is used for this function.
[no] init-reg-delay	Parameter type: <Sip::RegisterDelayInitialRegister> Format: - Configurable delayed initial register time for failure cases with response without Retry-After header included. range is [0,10...400] - unit: second	*optional parameter with default value: 180* Configurable delayed initial register time for failure cases with response without Retry-After header included. This object only applies to POTS. range is [0,10...400]

40.13 Voice Sip Transport Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip Transport params.

IPv6 is actually not supported. All references to IPv6 must be ignored since mentioned for future usage only.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no transport (trans-protocol) provider-name <Sip::TransportProviderName>) | (transport (trans-protocol) provider-name <Sip::TransportProviderName> [no admin-status | admin-status <Sip::TransportAdminStatus>] [no port-rcv | port-rcv <Sip::TransportPortRcv>] [no tcp-idle-time | tcp-idle-time <Sip::TransportTCPIdleTime>] [no max-out-udp-size | max-out-udp-size <Sip::TransportMaxOutgUdpReqSize>] [no tls-port-rcv | tls-port-rcv <Sip::TransportPortTLSRcv>])

Command Parameters

Table 40.13-1 "Voice Sip Transport Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(trans-protocol)	Format: (udp | tcp | udp_tcp | tls_over_tcp | udp_tls | tcp_tls | udp_tcp_tls | udp_swo_tcp) Possible values: - udp : transport protocol is UDP - tcp : transport protocol is TCP - udp_tcp : transport protocol is UDP or TCP. TCP is the preferred transport protocol. If UDP would be used and the size of the message exceeds a configured threshold then a switch can be made to TCP - tls_over_tcp : transport protocol is TLSoTCP - udp_tls : transport protocol is UDP or TLSoTCP. TLSoTCP is the preferred transport protocol - tcp_tls : transport protocol is TCP or TLSoTCP. TLSoTCP	Terminations will use this transport for establishing the calls

Resource Identifier	Type	Description
	is the preferred transport protocol - udp_tcp_tls : transport protocol is UDP, TCP or TLSoTCP. TLSoTCP is the preferred transport protocol. The next preferred transport protocol is TCP and finally UDP. If UDP would be used and the size of the message exceeds a configured threshold then a switch can be made to TCP - udp_swo_tcp : transport protocol is UDP or TCP. Basically, UDP is to be used. When size of message exceeds a configured threshold then a switch is made to TCP	
provider-name	Parameter type: <Sip::TransportProviderName> Format: - uniquely identifies the transport provider name - length: x<=32	uniquely identify transport provider name

Table 40.13-2 "Voice Sip Transport Configuration Commands" Command Parameters

Parameter	Type	Description
[no] admin-status	Parameter type: <Sip::TransportAdminStatus> Format: (up | down) Possible values: - up : unlock the dns server - down : lock the dns server	*optional parameter with default value: "down"* change administrative status
[no] port-rcv	Parameter type: <Sip::TransportPortRcv> Format: - Defines transport protocol port the UserAgent has to listen to for incoming SIP requests. - range: [0...65534]	*optional parameter with default value: 5060* Defines transport protocol port the User Agent has to listen to for incoming SIP requests.
[no] tcp-idle-time	Parameter type: <Sip::TransportTCPIdleTime> Format: - Define the max time period that a TCP shall be kept alive without exchanging any msg. - unit: second - range: [32...3600]	*optional parameter with default value: 64* Define the max time period that a tcp shall be kept alive without exchanging any msg.
[no] max-out-udp-size	Parameter type: <Sip::TransportMaxOutgUdpReqSize> Format: - Define the maximum SIP message size that can be sent over UDP. The minimum value of IPv4 is 576, the minimum value for IPv6 is 1280 - unit: bytes - range: [576...65535]	*optional parameter with default value: 65535* Define the maximum SIP message size that can be sent over udp
[no] tls-port-rcv	Parameter type: <Sip::TransportPortTLSRcv> Format: - Define transport protocol port to which the SIP User Agent will listen when TLSoTCP applies as underlying transport protocol for SIP. - range: [0...65534]	*optional parameter with default value: 5061* Define transport protocol port to which the SIP User Agent will listen when TLSoTCP applies as underlying transport protocol for SIP.

40.14 Voice Sip Redundancy Table Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip Redundancy table.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip redundancy (admin-domain-name) [no support-redun | support-redun <Sip::NetwRedunSupported>] [no dns-purge-timer | dns-purge-timer <Sip::NetwRedunDnsPurgeTimer>] [no dns-ini-retr-int | dns-ini-retr-int <Sip::NetwRedunDnsInitRetranTimer>] [no dns-max-retr-nbr | dns-max-retr-nbr <Sip::NetwRedunDnsMaxRetrNbr>] [no fg-monitor-method | fg-monitor-method <Sip::NetwRedunFgMonitorMethod>] [no fg-monitor-int | fg-monitor-int <Sip::NetwRedunFgMonitorInt>] [no bg-monitor-method | bg-monitor-method <Sip::NetwRedunBgMonitorMethod>] [no bg-monitor-int | bg-monitor-int <Sip::NetwRedunBgMonitorInt>] [no stable-obs-period | stable-obs-period <Sip::NetwRedunStableObsPeriod>] [no fo-hystersis | fo-hystersis <Sip::NetwRedunFailoverHystersis>] [no del-upd-threshold | del-upd-threshold <Sip::NetwRedunDeliUpdThreshold>] [no auto-server-fo | auto-server-fo <Sip::NetwRedunAutoFailOver>] [no auto-server-fb | auto-server-fb <Sip::NetwRedunAutoFailBack>] [no auto-sos-fo | auto-sos-fo <Sip::NetwRedunAutoFailOver>] [no auto-sos-fb | auto-sos-fb <Sip::NetwRedunAutoSosFailback>] [no rtry-after-thrsh | rtry-after-thrsh <Sip::NetwRedunRetryAfterThreshold>] [no options-max-fwd | options-max-fwd <Sip::NetwRedunOPTIONSMaxForward>] [no dns-redun-mode | dns-redun-mode <Sip::NetwRedundnsredunmode>] [no fail-obs-timer | fail-obs-timer <Sip::NetwRedunFailureObservationTimer>] [no fg-intv-503 | fg-intv-503 <Sip::NetwRedun503FGMonitorIntv>] [no time-thrsh-503 | time-thrsh-503 <Sip::NetwRedun503MaxTimeThreshold>] [no nbr-thrsh-503 | nbr-thrsh-503 <Sip::NetwRedun503MaxNbrThreshold>] [no auto-srv-fo-timer | auto-srv-fo-timer <Sip::netwRedunAutoSIPServerFOTimer>]

Command Parameters

Table 40.14-1 "Voice Sip Redundancy Table Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(admin-domain-name)	Format: - uniquely name of this element - length: x<=32	The administrative domain name of the (farm of) SIP First hop(s)

Table 40.14-2 "Voice Sip Redundancy Table Configuration Commands" Command Parameters

Parameter	Type	Description
[no] support-redun	Parameter type: <Sip::NetwRedunSupported> Format: (enable	*optional parameter with default value: "disable"* This object allows the

Parameter	Type	Description
	| disable) Possible values: - enable : support Redundancy - disable : doesn't support Redundancy	administrator to define whether the Voice Service Gateway network supports redundancy or not.
[no] dns-purge-timer	Parameter type: <Sip::NetwRedunDnsPurgeTimer> Format: - Expire of the DNS Purge Timer, a value 0 means that the DNS purge timer is disabled - unit: second - range: [0...86400]	*optional parameter with default value: 86400* Expire of the DNS Purge Timer, a value 0 means that the DNS purge timer is disabled.
[no] dns-ini-retr-int	Parameter type: <Sip::NetwRedunDnsInitRetranTimer> Format: - Initial DNS retransmission Interval - unit: milliseconds - range: [200...1000]	*optional parameter with default value: 500* The initial DNS query retransmission interval(in ms).
[no] dns-max-retr-nbr	Parameter type: <Sip::NetwRedunDnsMaxRetrNbr> Format: - Maximum DNS retransmission times - range: [0...4]	*optional parameter with default value: 2* The maximum DNS query retransmission times.
[no] fg-monitor-method	Parameter type: <Sip::NetwRedunFgMonitorMethod> Format: (register-method | options-method | passive-heartbeat) Possible values: - register-method : Foreground Service Healthy Monitoring mode : register-method - options-method : Foreground Service Healthy Monitoring mode : options-method - passive-heartbeat : Foreground Service Healthy Monitoring mode : passive-heartbeat	*optional parameter with default value: "register-method"* The Foreground healthy monitoring method.
[no] fg-monitor-int	Parameter type: <Sip::NetwRedunFgMonitorInt> Format: - The Foreground healthy monitoring interval, range is [0,30..3600],0 means disable. - unit: second	*optional parameter with default value: 90* The Foreground healthy monitoring interval, range is [0,30..3600],0 means disable.
[no] bg-monitor-method	Parameter type: <Sip::NetwRedunBgMonitorMethod> Format: options-method Possible values: - options-method : Background Service Healthy Monitoring mode : options-method	*optional parameter with default value: "options-method"* The Background healthy monitoring method.
[no] bg-monitor-int	Parameter type: <Sip::NetwRedunBgMonitorInt> Format: - The Background healthy monitoring interval,range is [0,60..86400], 0 means disable. - unit: second	*optional parameter with default value: 3600* The Background healthy monitoring interval,range is [0,60..86400], 0 means disable.
[no] stable-obs-period	Parameter type: <Sip::NetwRedunStableObsPeriod> Format: - Configurable stable operation observation period, a value 0 means that the Stable-Operation Observation Period is disabled - unit: second - range: [0...86400]	*optional parameter with default value: 21600* Configurable stable operation observation period, a value 0 means that the Stable-Operation Observation Period is disabled.
[no] fo-hystersis	Parameter type: <Sip::NetwRedunFailoverHystersis> Format: - Fail-Over Hysteresis Threshold, a value 0 means that the	*optional parameter with default value: 0* The Fail-Over hysteresis,a value

Parameter	Type	Description
	Fail-Over Hysteresis is disabled - range: [0...5]	0 means that the Fail-Over Hysteresis is disabled.
[no] del-upd-threshold	Parameter type: <Sip::NetwRedunDeliUpdThreshold> Format: - The Deliberate Update Threshold. When set to 0, then upon a SIP server priority change or a change of the list of SIP servers. DU is immediately triggered for idle SIP terminations. DU is triggered upon call completion for SIP terminations involved in a call, with a maximum time limit of 60 sec. i.e. if after 60 sec the call is still not finished, then the call is aborted and DU is performed for that SIP termination. - unit: second - range: [0...86400]	*optional parameter with default value: 0* The Deliberate Update Threshold. When set to 0, then upon a SIP server priority change or a change of the list of SIP servers. DU is immediately triggered for idle SIP terminations. DU is triggered upon call completion for SIP terminations involved in a call, with a maximum time limit of 60 sec. i.e. if after 60 sec the call is still not finished, then the call is aborted and DU is performed for that SIP termination.
[no] auto-server-fo	Parameter type: <Sip::NetwRedunAutoFailOver> Format: (enable | disable) Possible values: - enable : Support autonomous Fail-Over - disable : Not support autonomous Fail-Over	*optional parameter with default value: "disable"* This object allows to enable/disable autonomous SIP Server Fail-Over.
[no] auto-server-fb	Parameter type: <Sip::NetwRedunAutoFailBack> Format: - support/unsupport Autonomous SIP server Fail-Back, a value 0 means disable this functionality, not 0 means that SIP server Fail-Back is enabled. The value indicates the time period by which the SIP server Fail-Back must be done - unit: second - range: [0...86400]	*optional parameter with default value: 0* This object allows to enable/disable autonomous SIP Server Fail-Back.If value != 0, the value indicates the time period by which the SIP server Fail-Back must be completed, a value 0 means disable
[no] auto-sos-fo	Parameter type: <Sip::NetwRedunAutoFailOver> Format: (enable | disable) Possible values: - enable : Support autonomous Fail-Over - disable : Not support autonomous Fail-Over	*optional parameter with default value: "disable"* This object allows to enable/disable autonomous SOS Fail-Over.
[no] auto-sos-fb	Parameter type: <Sip::NetwRedunAutoSosFailback> Format: - support/unsupport Autonomous SOS Fail-Back, a value 0 means disable this functionality,not 0 means that SIP SOS Fail-Back is enabled. The value indicates the time period by which the SIP SOS Fail-Back must be done - unit: second - range: [0...86400]	*optional parameter with default value: 0* This object allows to enable/disable autonomous SOS Fail-Back.If value != 0, the value indicates the time period by which the SOS Fail-Back must be completed,a value 0 means disable
[no] rtry-after-thrsh	Parameter type: <Sip::NetwRedunRetryAfterThreshold> Format: - Retry-after threshold. - unit: milliseconds - range: [0...30000]	*optional parameter with default value: 0* The retry after Threshold.
[no] options-max-fwd	Parameter type: <Sip::NetwRedunOPTIONSMaxForward> Format:	*optional parameter with default value: 0*

Parameter	Type	Description
	- Max-Forward of OPTION, Value - 0: Must not forward this OPTIONS, Value - 1: Allow to forward once. - range: [0...1]	Max-forwarding is used to configure max-forward attribute for the forwarding property of the SIP OPTIONS message.
[no] dns-redun-mode	Parameter type: <Sip::NetwRedundnsredunmode> Format: (dns-redun-primary | dns-redun-success) Possible values: - dns-redun-primary : dns redun mode is dns-redun-primary - dns-redun-success : dns redun mode is dns-redun-success	*optional parameter with default value: "dns-redun-primary"* DNS Redundancy Mode, dns-redun-primary means the DNS query will be sent to primary DNS server always firstly,dns-redun-success Means the DNS query will be sent to the DNS server where the previous request was successfully respond.
[no] fail-obs-timer	Parameter type: <Sip::NetwRedunFailureObservationTimer> Format: - The failure observation timer, range is [0,60..480]. - unit: second	*optional parameter with default value: 240* Configurable the failure observation timer for failure cases, The failure observation timer allowing to verify whether after the receipt of a recoverable error, a SIP register is still successfully replied. This object only applies to POTS. range is [0,60..480].
[no] fg-intv-503	Parameter type: <Sip::NetwRedun503FGMonitorIntv> Format: - The 503-restoration-mode FGHM interval, range is [0,2..3600]. - unit: second	*optional parameter with default value: 2* The configurable 503-restoration-mode FGHM interval that applies from the moment a 503 error response is received to the FGHM re-register request. This object has only relevance in case the overall restoration behaviour regarding a 503 response to a SIP register request is enabled in the SIP service Profile. This object only applies to POTS. range is [0,2..3600].
[no] time-thrsh-503	Parameter type: <Sip::NetwRedun503MaxTimeThreshold> Format: - The maximum elapse time during which a consecutive series of 503 responses to initial/re-register requests is allowed, range is [0,180..7200]. - unit: second	*optional parameter with default value: 0* The maximum elapse time during which a consecutive series of 503 responses to initial/re-register requests is allowed. The system will trigger a fail-over when EITHER the 503MaxTimeThreshold OR the 503MaxNbrThreshold is reached i.e. only one of the conditions must be fulfilled to trigger a fail-over. This object only applies to POTS. range is [0,180..7200].
[no] nbr-thrsh-503	Parameter type: <Sip::NetwRedun503MaxNbrThreshold> Format: - The maximum number of consecutive 503 responses to	*optional parameter with default value: 1* The maximum number of

Parameter	Type	Description
	initial/re-register requests that is allowed, range is [0,1,2..144].	consecutive 503 responses to initial/re-register requests that is allowed. The system will trigger a fail-over when EITHER the 503MaxTimeThreshold OR the 503MaxNbrThreshold is reached i.e. only one of the conditions must be fulfilled to trigger a fail-over. This object only applies to POTS. range is [0,1,2..144].A value 1 means that the 503 Max Number Threshold equals [the number of configured and administratively enabled SIP terminations at the voice LT board * 2].
[no] auto-srv-fo-timer	Parameter type: <Sip::netwRedunAutoSIPServerFOTimer> Format: - The configured elapse time by which a graceful autonomous SIP Server Fail-Over/SOS Fail-Over must be completed for all involved SIP terminations. When set to the default value '86400': The netwRedunAutoSIPServerFOTimer is considered as 'disabled'. For those SIP terminations that are involved in an ongoing call, the fail-over is postponed till the ongoing call has terminated. (backward compatible behaviour.) When set to the value '0': The fail-over (for POTS as well as ISDN PRI and CAS R2 SIP terminations) is immediately triggered. Calls that are ongoing at the moment the fail-over is triggered are immediately dropped. When set to a value [1..86399]: For those POTS SIP terminations that are involved in an ongoing call at the moment the Fail-Over is triggered, the Fail-Over will be postponed until either the call is released (while the FO timer is still running) or the FO timer has expired or session-refresh failure or SIP in-dialog communication failure. Ongoing calls are immediately dropped upon the expiry of the FO timer. For ISDN PRA and CAS R2 terminations, the Fail-Over is immediately triggered, however ongoing calls are kept until either the call is released (while the FO timer is still running) or the FO timer has expired or session-refresh failure or SIP in-dialog communication failure. Ongoing calls are immediately dropped upon the expiry of the FO timer. New calls are established via E1 interface/B-channels that already made a fail-over. - unit: second - range: [0...86400]	*optional parameter with default value: 86400* The configured elapse time by which a graceful autonomous SIP Server Fail-Over/SOS Fail-Over must be completed for all involved SIP terminations. When set to the default value '86400': The netwRedunAutoSIPServerFOTimer is considered as 'disabled'. For those SIP terminations that are involved in an ongoing call, the fail-over is postponed till the ongoing call has terminated. (backward compatible behaviour.) When set to the value '0': The fail-over (for POTS as well as ISDN PRI and CAS R2 SIP terminations) is immediately triggered. Calls that are ongoing at the moment the fail-over is triggered are immediately dropped. When set to a value [1..86399]: For those POTS SIP terminations that are involved in an ongoing call at the moment the Fail-Over is triggered, the Fail-Over will be postponed until either the call is released (while the FO timer is still running) or the FO timer has expired or session-refresh failure or SIP in-dialog communication failure. Ongoing calls are immediately dropped upon the expiry of the FO timer. For ISDN PRA and CAS R2 terminations, the Fail-Over is immediately triggered, however ongoing calls are kept until either the call is

Parameter	Type	Description
		released (while the FO timer is still running) or the FO timer has expired or session-refresh failure or SIP in-dialog communication failure. Ongoing calls are immediately dropped upon the expiry of the FO timer. New calls are established via E1 interface/B-channels that already made a fail-over.

40.15 Voice Sip System Session Timer Configuration Commands

Command Description

this command allows the operator to manage the voice sip session timer profile. The timer is used to consult with remote, and at last, if session timer was enabled, the refresh timer will be started by IPTK. The SIP Session Timer feature adds the capability to periodically refresh SIP sessions by sending repeated INVITE requests. The repeated INVITE requests, or re-INVITEs, are sent during an active call leg to allow user agents (UA) or SIP proxies to determine the status of a SIP session. Without this keep alive mechanism, proxies that remember incoming and outgoing requests (stateful proxies) may continue to retain call state needlessly. If a UA fails to send a BYE message at the end of a session or if the BYE message is lost because of network problems, a stateful proxy does not know that the session has ended. The re-INVITES ensure that active sessions stay active and completed sessions are terminated.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip system session-timer [[no] enable] [no status | status <Sip::SessionTimerAdminState>] [no min-se-time | min-se-time <Sip::SessionTimerMinSE>] [no se-time | se-time <Sip::SessionTimerSessionExpire>] [no admin-status | admin-status <Sip::SysObjectsAdminStatus>]

Command Parameters

Table 40.15-2 "Voice Sip System Session Timer Configuration Commands" Command Parameters

Parameter	Type	Description
[no] enable	Parameter type: boolean	*obsolete parameter replaced by parameter "status"* Prefix activated of session timer is enable
[no] status	Parameter type: <Sip::SessionTimerAdminState> Format: (enable | disable | enable-uas) Possible values: - enable : Prefix activated of session timer is enable - disable : Prefix activated of session timer is disable - enable-uas : Prefix activated of session timer is enableAsUas	*optional parameter with default value: "disable"* To configure the SIP Session Timer feature capability. If it's disable, shall not request session expiration in initial INVITE requests and 200 OK responses If it's enable, shall request session expiration in initial INVITE requests and 200 OK responses, and in both cases, if an incoming initial INVITE request contains a

Parameter	Type	Description
		session expiration header, but does not include the refresher value, then shall use uac for the refresher parameter. If it's enable-uas, shall request session expiration in initial INVITE requests and 200 OK responses, and in both cases, if an incoming initial INVITE request contains a session header, but does not include the refresher value, then shall use uas for the refresher parameter.
[no] min-se-time	Parameter type: <Sip::SessionTimerMinSE> Format: - value of min-se when use session time - unit: second - range: [90...65535]	*optional parameter with default value: "90"* Minimum Session Expires time. The lower bound for the session refresh interval. Because of the processing load of INVITE requests, the SIP proxy, User Agent Client and User Agent Server can have a configured minimum refresh timer value that they can accept. It is conveyed in the Min-SE header in the initial INVITE request. When making a call, the presence of the Min-SE header informs the UAS and any proxy of the minimum value that the UAC accepts for the session timer duration, in units of delta-seconds.
[no] se-time	Parameter type: <Sip::SessionTimerSessionExpire> Format: - value of session-expires when use sessiontime - unit: second - range: [90...65535]	*optional parameter with default value: "1800"* Session Expires Time. The upper bound for the session refresh interval. It is conveyed in the Session-Expires header in the initial INVITE request. If a session refresh request is not received before the interval passes, the session is considered terminated.
[no] admin-status	Parameter type: <Sip::SysObjectsAdminStatus> Format: (up | down) Possible values: - up : unlock the sipSysObjects - down : lock the sipSysObjects	*optional parameter with default value: "down"* The administrative status of this sipSysObjects

40.16 Voice Sip DNS Server Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip DNS Server profile.

IPv6 is actually not supported. All references to IPv6 must be ignored since mentioned for future usage only.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no dnsserver (name)) | (dnsserver (name) [no admin-status | admin-status <Sip::sipDNSServerAdminStatus>] address <Sip::sipDNSServerAddr> [no priority | priority <Sip::sipDNSServerPriority>] [no site | site <Sip::sipDNSServerSite>] admin-domain-name <Sip::AdminDomName> [no port | port <Sip::sipDNSServerPort>])

Command Parameters

Table 40.16-1 "Voice Sip DNS Server Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(name)	Format: - uniquely identifies this voice dns server - length: 1<=x<=32	uniquely identify of this sip DNS server

Table 40.16-2 "Voice Sip DNS Server Configuration Commands" Command Parameters

Parameter	Type	Description
[no] admin-status	Parameter type: <Sip::sipDNSServerAdminStatus> Format: (up | down) Possible values: - up : unlock the dns server - down : lock the dns server	*optional parameter with default value: "down"* administrative status of this DNS server
address	Parameter type: <Sip::sipDNSServerAddr> Format: (ipv4 : <Ip::V4Address> | ipv6 : <Ip::V6Address>) Possible values: - ipv4 : the address type of the server is IPv4 - ipv6 : the address type of the server is IPv6 Field type <Ip::V4Address>	*mandatory parameter* The address(IPv4 or IPv6) of this DNS server

Parameter	Type	Description
	- IPv4-address Field type <Ip::V6Address> - IPv6-address	
[no] priority	Parameter type: <Sip::sipDNSServerPriority> Format: - the SIP dns server priority, lower value with higher priority - range: [0...65535]	*optional parameter with default value: 100* The priority of sip DNS server,lower value with higher priority
[no] site	Parameter type: <Sip::sipDNSServerSite> Format: (primary-site | geo-backup-site) Possible values: - primary-site : the server belongs to the GEO primary site - geo-backup-site : the server belongs to the GEO secondary site	*optional parameter with default value: "primary-site"* The site which the sip DNS server is belonged to
admin-domain-name	Parameter type: <Sip::AdminDomName> Format: - uniquely name of this element - length: x<=32	*mandatory parameter* The administrative domain name of the DNS server. The administrative domain might be the VoIP Access Seeker Network or the Connectivity Serving Network.
[no] port	Parameter type: <Sip::sipDNSServerPort> Format: - - range: [1...65534]	*optional parameter with default value: 53* The remote port of sip DNS server, a value 0 means use default value 53

40.17 Voice Sip DHCP Authentication Params Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip DHCP authentication params.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no dhcp-authent-para (ua-name) secret-id <Sip::DHCPAuthentParaSecretId>) | (dhcp-authent-para (ua-name) secret-id <Sip::DHCPAuthentParaSecretId> [no key | key <Sip::DHCPAuthentParaKey>] [no action-type | action-type <Sip::DHCPAuthentParaActionType>])

Command Parameters

Table 40.17-1 "Voice Sip DHCP Authentication Params Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(ua-name)	Format: - uniquely identifies the User Agent - length: 1<=x<=32	uniquely identify of the user agent
secret-id	Parameter type: <Sip::DHCPAuthentParaSecretId> Format: - Unique identifier of the DHCP message authentication parameter record - range: [1...4294967295]	A unique identifier of the DHCP message authentication parameter record

Table 40.17-2 "Voice Sip DHCP Authentication Params Configuration Commands" Command Parameters

Parameter	Type	Description
[no] key	Parameter type: <Sip::DHCPAuthentParaKey> Format: - DHCP secret key used for DHCP message authentication. Every letter in key must be in '0'~'9', 'a'~'f', 'A'~'F' and the string length must be even. - length: x<=32	*optional parameter with default value: ""* DHCP secret key used for DHCP message authentication. Every letter in key must be in '0'~'9', 'a'~'f', 'A'~'F' and the string length must be even.
[no] action-type	Parameter type: <Sip::DHCPAuthentParaActionType> Format: (normal-req-key | force-discover	*optional parameter with default value: "normal-send-req"* the action type for the configured DHCP secret keys

Parameter	Type	Description
	| force-request | normal-force-dis | normal-force-req) Possible values: - normal-req-key : used when send DHCP request, not used for response - force-discover : used when force renew received with this key, will send DHCP discover - force-request : used when force renew received with this key, will send DHCP renew request - normal-force-dis : used when send DHCP request, not used for response, and when force renew received with this key, will send DHCP discover - normal-force-req : used when send DHCP request, not used for response, and when force renew received with this key, will send DHCP renew request	

40.18 Voice Sip Restoration Failover/Failback Type/Mode Configuration Commands

Command Description

This command allows the operator to change the Failover/Failback type or mode.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip redundancy-cmd (domain-name) [no start-time | start-time <Sip::FailXStartTime>] [no end-time | end-time <Sip::FailXEndTime>] [fail-x-type <Sip::FailXType>]

Command Parameters

Table 40.18-1 "Voice Sip Restoration Failover/Failback Type/Mode Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(domain-name)	Format: - uniquely identifies the transport provider name - length: x<=32	uniquely identify of admin domain name

Table 40.18-2 "Voice Sip Restoration Failover/Failback Type/Mode Configuration Commands" Command Parameters

Parameter	Type	Description
[no] start-time	Parameter type: <Sip::FailXStartTime> Format: - The absolute time when the manually triggered GEO Fail-Over / Fail-Back needs to start. The unit is in seconds. - unit: second - range: [0...4294967295]	*optional parameter with default value: 0* The absolute time when the manually triggered GEO Fail-Over / Fail-Back needs to start. The unit is in seconds.
[no] end-time	Parameter type: <Sip::FailXEndTime> Format: - The absolute time when the manually triggered graceful GEO Fail-Over / Fail-Back needs to be completed. The unit is in seconds. - unit: second - range: [0...4294967295]	*optional parameter with default value: 0* The absolute time when the manually triggered graceful GEO Fail-Over / Fail-Back needs to be completed. The unit is in seconds.
fail-x-type	Parameter type: <Sip::FailXType> Format: (geo-fail-over | geo-fail-back)	*optional parameter* Fail X type, failover or failback.

Parameter	Type	Description
	Possible values: - geo-fail-over : geo fail over - geo-fail-back : geo fail back	

40.19 Voice Statistics Configure Command

Command Description

Set statistics configuration.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip statistics [no stats-5min-config | stats-5min-config <Sip::FiveMStats>] [no cdr-config | cdr-config <Sip::CallRecord>]

Command Parameters

Table 40.19-2 "Voice Statistics Configure Command" Command Parameters

Parameter	Type	Description
[no] stats-5min-config	Parameter type: <Sip::FiveMStats> Format: (disable | enable) Possible values: - disable : disable 5m stats function - enable : enable 5m stats function	*optional parameter with default value: "disable"* enable/disable the 5-Min Statistics
[no] cdr-config	Parameter type: <Sip::CallRecord> Format: (disable | enable) Possible values: - disable : disable call record function - enable : enable call record function	*optional parameter with default value: "disable"* enable/disable the call record

40.20 Voice Statistics Mode Configuration Command

Command Description

Change voice statistics Mode.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip statistics stats-config [[no] per-line] [[no] per-board] [[no] per-system] [[no] per-call] [[no] out-any-rsp] [[no] out-180-rsp] [[no] out-200-rsp] [[no] in-any-rsp] [[no] in-180-rsp] [[no] in-200-rsp]

Command Parameters

Table 40.20-2 "Voice Statistics Mode Configuration Command" Command Parameters

Parameter	Type	Description
[no] per-line	Parameter type: boolean	*optional parameter* enable per line statistics function
[no] per-board	Parameter type: boolean	*optional parameter* enable per board statistics function
[no] per-system	Parameter type: boolean	*optional parameter* enable per system statistics function
[no] per-call	Parameter type: boolean	*optional parameter* enable per call statistics function
[no] out-any-rsp	Parameter type: boolean	*optional parameter* enable Arbitrary Response Mode for out-going call answered
[no] out-180-rsp	Parameter type: boolean	*optional parameter* enable 180 Response Mode for out-going call answered
[no] out-200-rsp	Parameter type: boolean	*optional parameter* enable 200 Response Mode for out-going call answered
[no] in-any-rsp	Parameter type: boolean	*optional parameter* enable Arbitrary Response Mode for in-coming call answered
[no] in-180-rsp	Parameter type: boolean	*optional parameter* enable 180 Response Mode for

Parameter	Type	Description
		in-coming call answered
[no] in-200-rsp	Parameter type: boolean	*optional parameter* enable 200 Response Mode for in-coming call answered

40.21 Voice Sip Isdn/Cas Termination Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip Isdn/Cas termination profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no isdn-cas-term (if-index)) | (isdn-cas-term (if-index) [no directory-number | directory-number <Sip::TermDnumber>] [no user-name | user-name <Sip::TermUserName>] [no display-name | display-name <Sip::TermUserName>] [no uri | uri <Sip::TermUri>] [no md5-realm | md5-realm <Sip::MD5Realm>] [no md5-password | md5-password <Security::Password4>] [no aka-secret-key | aka-secret-key <Security::Password8>] [no admin-status | admin-status <Sip::IsdnTermAdminStatus>] [no line-id | line-id <Sip::ISDNTermLineId>] [no testaccessstate | testaccessstate <Sip::TestAccessState>] [no busyoverwrite | busyoverwrite <Sip::TestAccessBusyOverWrite>] [no accessontimeout | accessontimeout <Sip::TestAccessonTimeout>] [no provider-name | provider-name <Sip::TermUserProviderName>] [no signalling-type | signalling-type <Sip::TermSignallingType>] [no cpn-screen | cpn-screen <Sip::TermCallingScreen>] [no default-cpc | default-cpc <Sip::TermDefaultCPC>] [no restrict-acc | restrict-acc <Sip::TermRestrictACC>] [no hunting-way | hunting-way <Sip::TermHuntingWay>] [no sip-rule-set | sip-rule-set <Sip::IsdnCasManipulationRuleSet>] [no cas-exch-type | cas-exch-type <Sip::TermExchangeType>] [no q-value | q-value <Sip::IsdnCasTerminationQValue>] [no reg-mode | reg-mode <Sip::IsdnCasTerminationRegMode>] [no e1-hunting-way | e1-hunting-way <Sip::IsdnCasTerminationE1HuntingWay>] [no e1-cluster-itf | e1-cluster-itf <Sip::IsdnCasTerminationE1ClusterItf>] [no e1-cluster-name | e1-cluster-name <Sip::IsdnCasTerminationE1ClusterName>] [no isdn-nsm-profile | isdn-nsm-profile <Sip::NsmProfile>] [no cas-nsm-profile | cas-nsm-profile <Sip::CasNsmProfPointer>])

Command Parameters

Table 40.21-1 "Voice Sip Isdn/Cas Termination Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(if-index)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number	The unique internal identifier of the termination port

Resource Identifier	Type	Description
	Field type <Eqpt::PortId> - the port number	

Table 40.21-2 "Voice Sip Isdn/Cas Termination Configuration Commands" Command Parameters

Parameter	Type	Description
[no] directory-number	Parameter type: <Sip::TermDnumber> Format: - identify the DN of the subscriber line - length: x<=32	*optional parameter with default value: ""* Identify the directory (telephone) number of the subscriber line
[no] user-name	Parameter type: <Sip::TermUserName> Format: - identify the user name for this port - length: x<=64	*optional parameter with default value: ""* Identify the user name for this termination port
[no] display-name	Parameter type: <Sip::TermUserName> Format: - identify the user name for this port - length: x<=64	*optional parameter with default value: ""* Display name for this termination port
[no] uri	Parameter type: <Sip::TermUri> Format: - SIP uri by which the user agent identifies subscriber line - length: x<=80	*optional parameter with default value: ""* SIP URI by which user agent identifies subscriber line
[no] md5-realm	Parameter type: <Sip::MD5Realm> Format: - the realm identifier (Due to legacy reasons, the MD5 character string was included in the object name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication realm identifier.) - length: x<=64	*optional parameter with default value: ""* Realm identifier corresponding to voice service gateway domain
[no] md5-password	Parameter type: <Security::Password4> Format: (prompt | plain : <Security::PlainPassword4>) Possible values: - prompt : prompts the operator for a password - plain : the password in plain text, may contain any printable character defined by the US-ASCII character set and that has a decimal code in the range 33 to 126, and the length should be less than or equal to 64 (Due to legacy reasons, the MD5 character string was included in the parameter name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication password.) Field type <Security::PlainPassword4> - the password, may contain any printable character defined by the US-ASCII character set and that has a decimal code in the range 33 to 126, and the length should be less than or equal to 64 (Due to legacy reasons, the MD5 character string was included in the parameter name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication password.)	*optional parameter with default value: "plain : "* The password associated with the user

Parameter	Type	Description
	- length: x<=64	
[no] aka-secret-key	Parameter type: <Security::Password8> Format: (prompt | plain : <Security::PlainPassword8>) Possible values: - prompt : prompts the operator for AKA shared secret key - plain : the AKA shared secret key in plain text, each character should be 0-f, and the length can only be 0 or 32 Field type <Security::PlainPassword8> - the AKA shared secret key, each character should be 0-f, and the length can only be 0 or 32 - length: x<=32	*optional parameter with default value: "plain : "* The AKA shared secret key
[no] admin-status	Parameter type: <Sip::IsdnTermAdminStatus> Format: (up | down) Possible values: - up : unlock the sip isdn/cas termination - down : lock the sip isdn/cas termination	*optional parameter with default value: "down"* Identify the status of this port administration
[no] line-id	Parameter type: <Sip::ISDNTermLineId> Format: - Prior to ISR6.0.01:\nThe Line ID of this PRA ISDN or R2 CAS SIP termination. It is contained in the Pani header.\n The line id configuration is not bounded to any format. It is the responsibility of the customer to define the desired line ID format.\n There is no restriction on the character set being used to configure the line ID/line ID format.\n From ISR6.0.01 onwards:\n The object allows the configuration, together or separately, for the pani header line id and/or the E1 interface CRC4 framing mode.\n In case the E1 interface CRC4 framing mode is not configured, the setting in the CDE profile applies. Otherwise(the E1 interface CRC4 framing mode is configured via this object), the input must be given via the following syntax: crc4:(enabled/disabled), 'crc4' is a fixed case-sensitive key-word immediately followed by a colon which in turn is immediately followed by the case-sensitive key-words 'enabled' or 'disabled'.\n 'enabled' means: Transmit and receive framing mode is set to CRC-multi frame.\n 'disabled' means: Transmit and receive framing mode is set to double frame.\n In case, it is required to configure both the CRC4 framing mode and the pani header line id, the CRC4 configuration input must be given prior to the pani header configuration input.\n In such case the syntax of the CRC4 configuration input is as described above, immediately followed by a semicolon and the semicolon immediately followed by the pani header line id in the same free format as described above: crc4:[enabled/disabled];[pani header line id]. - length: x<=63	*optional parameter with default value: ""* Identify the line id for this termination port
[no] testaccessstate	Parameter type: <Sip::TestAccessState> Format: (on | off) Possible values: - on : turn on TestAccessState - off : turn off TestAccessState	*optional parameter with default value: "off"* Test access status

Parameter	Type	Description
[no] busyoverwrite	Parameter type: <Sip::TestAccessBusyOverWrite> Format: (true | false) Possible values: - true : enable TestAccessBusyOverWrite parameter - false : disable TestAccessBusyOverWrite parameter	*optional parameter with default value: "false"* Test access busyoverwrite
[no] accessontimeout	Parameter type: <Sip::TestAccessonTimeout> Format: - value of test access on expires. - unit: second - range: [0...900]	*optional parameter with default value: "900"* Time to go until test access timeout. In case test access state is off, the value will be 0. In case test access state is on, this parameter is optional with default initial value 900. The value will start counting down, until value 0 and test access state will be off again
[no] provider-name	Parameter type: <Sip::TermUserProviderName> Format: - identify the user VSG name for this port - length: x<=32	*optional parameter with default value: "vsp1"* Identify the user VSG name for this termination port
[no] signalling-type	Parameter type: <Sip::TermSignallingType> Format: (isdn-pra | cas-r2) Possible values: - isdn-pra : set signalling type to ISDN PRA - cas-r2 : set signalling type to CAS R2	*optional parameter with default value: "isdn-pra"* Identify the signalling type for this termination port. It can not be modified after term created.
[no] cpn-screen	Parameter type: <Sip::TermCallingScreen> Format: (disable | enable) Possible values: - disable : disable calling party number screening - enable : enable calling party number screening	*optional parameter with default value: "enable"* Identify whether to support the screening of calling party number(CPN) for outgoing call of CAS R2 termination port. When enabled, the ISAMV shall request the calling party number from PBX and screen the CPN. When disabled, the ISAMV shall not request the calling party number and the default number assigned for the given termination shall be sent
[no] default-cpc	Parameter type: <Sip::TermDefaultCPC> Format: (passthrough | ordinary | specchg | test | payphone | operator | data | payphoneiu | atme) Possible values: - passthrough : bypass the cpc from PBX - ordinary : replace the cpc from PBX by ordinary	*optional parameter with default value: "passthrough"* Identify the default calling party's category for outgoing call of CAS R2 termination port. Value of passthrough means to bypass the cpc from PBX, other value means to replace the cpc from PBX by this value

Parameter	Type	Description
	- specchg : replace the cpc from PBX by specchg - test : replace the cpc from PBX by test - payphone : replace the cpc from PBX by payphone - operator : replace the cpc from PBX by operator - data : replace the cpc from PBX by data. data is not supported right now. - payphoneiu : replace the cpc from PBX by payphoneIU - atme : replace the cpc from PBX by atme	
[no] restrict-acc	Parameter type: <Sip::TermRestrictACC> Format: (disable | enable) Possible values: - disable : no restriction for terminated ACC - enable : restriction for terminated ACC	*optional parameter with default value: "disable"* Identify whether the PBX is restricted for terminated automatic collect call for incoming call of CAS R2 termination port
[no] hunting-way	Parameter type: <Sip::TermHuntingWay> Format: (cyclic-forward | sequential-forward | sequential-reverse | random-forward | cyclic-reverse | random-reverse) Possible values: - cyclic-forward : Initially the lowest B-channel is selected i.e. B-channel 1; afterwards, always again the next B-channel following the one used last is selected. - sequential-forward : The selection of the B-channel is based on an increasing port number, starting with the B-channel with the lowest port number of the E1/T1 interface and the next B-channel in the ordered series of B-channel is selected if the B-channel last used is busy. Again select the B-channel with the lowest port number. - sequential-reverse : The selection of the B-channel is based on an decreasing port number, starting with the B-channel with the highest port number of the E1/T1 interface and the next B-channel in the ordered series of B-channels is selected if the B-channel last used is busy. Again select the B-channel with the highest port number. - random-forward : allocation by random order. if the randomly selected B-channel is busy, the selection of a new candidate B-channel starts at the next in the ordered series of B-channels following the one been randomly selected, and follows increasing order. - cyclic-reverse : Initially the highest B-channel is selected i.e. B-channel 30; afterwards, always again the next B-channel following the one used last is selected. - random-reverse : allocation by random order. if the randomly selected B-channel is busy, the selection of a new candidate B-channel starts at the next in the ordered series of B-channels following the one been randomly selected, and follows decreasing order.	*optional parameter with default value: "cyclic-forward"* Identify the channel allocation method. For ISDN PRA, used for both outgoing and incoming calls. For CAS R2, only used for incoming call
[no] sip-rule-set	Parameter type: <Sip::IsdnCasManipulationRuleSet> Format: - the ISDN-PRA/CAS-R2 manipulation rule set, at most 4 rule ids can be configured for ISDN-PRA term and 3 rule ids for CAS-R2 term	*optional parameter with default value: ""* Identify the rules this termination can apply

Parameter	Type	Description
	- length: x<=4	
[no] cas-exch-type	Parameter type: <Sip::TermExchangeType> Format: (normal | compelled-a3-and-b1 | pulsed-a3-and-b1) Possible values: - normal : the exchange type is normal - compelled-a3-and-b1 : the exchange can only handle B-1 and compelled A3 when digit completed - pulsed-a3-and-b1 : the exchange can only handle B-1 and pulsed A3 when digit completed	*optional parameter with default value: "normal"* Only used for cas termination to identify the PBX type. It is only used for special legacy PBX in special nations and should be set to normal for normal PBX, otherwise it will break
[no] q-value	Parameter type: <Sip::IsdnCasTerminationQValue> Format: - The q value is a number specified as parameter in the Contact header field, denotes an order of preference - range: [0...100]	*optional parameter with default value: "100"* The q value is a number specified as parameter in the Contact header field, denotes an order of preference
[no] reg-mode	Parameter type: <Sip::IsdnCasTerminationRegMode> Format: (explicit | implicit | no-reg) Possible values: - explicit : The Explicit register (ER) mode is only relevant in case a list of individual DDI-EXTs has been manually configured and associated with this SIP termination - implicit : The Implicit register (IR) mode, the list of implicit registered DDI-EXTs is returned by the register server in the 200 OK response to the SIP register request - no-reg : The no-registeration mode is only relevant in case the sip UA does not have to send any sip REGISTER / re-REGISTER request to the sipserver nor for the GDN, nor for the optional manually configured DDI-EXT.	*optional parameter with default value: "implicit"* The sipIsdnCasTerminationRegMode object allows to configure whether the implicit or explicit register mode applies
[no] e1-hunting-way	Parameter type: <Sip::IsdnCasTerminationE1HuntingWay> Format: (cyclic-forward | sequential-forward-fl | sequential-reverse-fl | random-forward | cyclic-reverse | random-reverse | sequential-forward-ud | sequential-reverse-ud) Possible values: - cyclic-forward : The selection of the E1 interfaces is a cyclic process in which the different E1 interface are alternately selected.At the first selection attempt the first E1 in the configuration input will be selected and If no free channel on this interface the selection proceeds with the succeeding interface (i.e. in forward order) in the configuration order.For any subsequent selection attempt, the selection process will start at the interface which is succeeding the one in the configuration order that was selected last for the previous attempt (i.e. starting point proceeds in forward order). - sequential-forward-fl : Each time always start from the first	*optional parameter with default value: "cyclic-forward"* The selection of the E1 interface ways upon the receipt of an incoming call attempt (received from the IMS core side)

Parameter	Type	Description
	E1 in the configuration input, will select the next E1 in the order configured only when all B channels of current E1 are busy. - sequential-reverse-fl : Each time always start from the last E1 in the configuration input, will select the preceding E1 in the order configured only when all B channels of current E1 are busy. - random-forward : The selection of the E1 interfaces happens randomly. If selected E1 interface has no idle channel, then take the next E1 interface(forward) - cyclic-reverse : The selection of the E1 interfaces is a cyclic process in which the different E1 interface are alternately selected.At the first selection attempt the last E1 in the configuration input will be selected and If no free channel on this interface the selection proceeds with the preceding interface (i.e in reverse order) in the configuration order.For any subsequent selection attempt, the selection process will start at the interface which is preceding the one in the configuration order that was selected last for the previous attempt (i.e starting point proceeds in reverse order) - random-reverse : The selection of the E1 interfaces happens randomly. If selected E1 interface has no idle channel, then take the next E1 interface(reverse) - sequential-forward-ud : Aiming at having a equal load (more or less equal number of busy B-channels for each of the E1 interfaces in the E1 cluster. MIB readiness only. Unsupport now - sequential-reverse-ud : Aiming at having a equal load (more or less equal number of busy B-channels for each of the E1 interfaces in the E1 cluster. MIB readiness only. Unsupport now	
[no] e1-cluster-itf	Parameter type: <Sip::IsdnCasTerminationE1ClusterItf> Format: - The ifindex of interfaces.At maximum 8 ifIndexs can be configured.The ifindex used as index of the ISDN PRI/CAS R2 SIP termination must be configured in IsdnCasTerminationE1ClusterItf object.The order of the E1 interface ifindices in the configuration input for the sipIsdnCasTerminationE1ClusterItf object defines the priority of the E1 interface selection.	*optional parameter with default value: ""* A SIP termination is created on top of an E1 cluster that embraces multiple E1 interfaces
[no] e1-cluster-name	Parameter type: <Sip::IsdnCasTerminationE1ClusterName> Format: - Name of the E1 cluster - length: x<=32	*optional parameter with default value: ""* Name of the E1 cluster
[no] isdn-nsm-profile	Parameter type: <Sip::NsmProfile> Format: (none | <Sip::NsmProfilePointer> | name : <PrintableString>) Possible values: - none : no ISDN NSM profile assigned. - name : ISDN NSM profile name Field type <Sip::NsmProfilePointer> - the index of the ISDN NSM profile. - range: [1...32] Field type <PrintableString>	*optional parameter with default value: "none"* The nsm profile which should be associated with the ISDN PRA termination. For CAS R2 termination, it must be none.

Parameter	Type	Description
	- printable string	
[no] cas-nsm-profile	Parameter type: <Sip::CasNsmProfPointer> Format: (none | <Sip::CasNsmProfPointer> | name : <PrintableString>) Possible values: - none : no CAS NSM profile assigned. - name : CAS NSM profile name Field type <Sip::CasNsmProfPointer> - the index of the CAS NSM profile. - range: [1...32] Field type <PrintableString> - printable string	*optional parameter with default value: "none"* The cas nsm profile which should be associated with the CAS R2 termination. For ISDN PRA termination, it must be none.

40.22 Voice Sip Isdn/Cas Termination Configuration Commands

Command Description

This command allows the operator to manage the Isdn Pri /Cas SIP termination created on top of an E1/T1 trunk (sub-)group.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no pri-cas-tksg-term (if-index) tksg-id <Sip::TrunkSubGroupId>) | (pri-cas-tksg-term (if-index) tksg-id <Sip::TrunkSubGroupId> [no directory-number | directory-number <Sip::TermDnumber>] [no user-name | user-name <Sip::TermUserName>] [no display-name | display-name <Sip::TermDisplayName>] [no uri | uri <Sip::TermUri>] [no realm | realm <Sip::MD5Realm>] [no password | password <Security::Password4>] [no aka-secret-key | aka-secret-key <Security::Password8>] [no admin-status | admin-status <Sip::IsdnTermAdminStatus>] [no line-id | line-id <Sip::PriTksgTermLineId>] [no provider-name | provider-name <Sip::TermUserProviderName>] [no ds0-hunting-way | ds0-hunting-way <Sip::TermHuntingWay>] [no itf-hunting-way | itf-hunting-way <Sip::IsdnCasTerminationItfHuntingWay>] [no sip-rule-set | sip-rule-set <Sip::IsdnCasManipulationTksgRuleSet>] [no q-value | q-value <Sip::IsdnCasTerminationQValue>] [no reg-mode | reg-mode <Sip::IsdnCasTerminationRegMode>] [no extension | extension <Sip::SipPriCasTksgTermExtension>] trunk-group-name <Sip::SipPriCasTksgTermTrunkGroupName> [no signalling-type | signalling-type <Sip::TksgTermSignallingType>] [no cas-cpn-screen | cas-cpn-screen <Sip::TermCallingScreen>] [no cas-default-cpc | cas-default-cpc <Sip::TermDefaultCPC>] [no cas-restrict-acc | cas-restrict-acc <Sip::TermRestrictACC>] [no cas-exch-type | cas-exch-type <Sip::TermExchangeType>] [no isdn-nsm-profile | isdn-nsm-profile <Sip::NsmProfile>] [no cas-nsm-profile | cas-nsm-profile <Sip::CasNsmProfPointer>] [no uus-profile | uus-profile <Sip::UserToUserProfile>] [no cas-timer-profile | cas-timer-profile <Sip::CasTimerProfPointer>])

Command Parameters

Table 40.22-1 "Voice Sip Isdn/Cas Termination Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(if-index)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId>	The ifindex of one of E1/T1 interfaces of the trunk (sub-)group on top of which the ISDN PRI / CAS SIP termination is created

Resource Identifier	Type	Description
	- the LT slot number Field type <Eqpt::PortId> - the port number	
tksg-id	Parameter type: <Sip::TrunkSubGroupId> Format: - The index of E1/T1 interface trunk sub-group. - range: [0...256]	The E1/T1 Trunk sub-group ID.

Table 40.22-2 "Voice Sip Isdn/Cas Termination Configuration Commands" Command Parameters

Parameter	Type	Description
[no] directory-number	Parameter type: <Sip::TermDnumber> Format: - identify the DN of the subscriber line - length: x<=32	*optional parameter with default value: ""* The global directory number (GDN)
[no] user-name	Parameter type: <Sip::TermUserName> Format: - identify the user name for this port - length: x<=64	*optional parameter with default value: ""* The user name for this ISDN PRI / CAS SIP termination
[no] display-name	Parameter type: <Sip::TermDisplayName> Format: - identify the display name for this port - length: x<=64	*optional parameter with default value: ""* The display name for this ISDN PRI / CAS SIP termination
[no] uri	Parameter type: <Sip::TermUri> Format: - SIP uri by which the user agent identifies subscriber line - length: x<=80	*optional parameter with default value: ""* The sip uri by which the SIP user agent identifies the SIP termination
[no] realm	Parameter type: <Sip::MD5Realm> Format: - the realm identifier (Due to legacy reasons, the MD5 character string was included in the object name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication realm identifier.) - length: x<=64	*optional parameter with default value: ""* The Realm of the voice service gateway domain
[no] password	Parameter type: <Security::Password4> Format: (prompt | plain : <Security::PlainPassword4>) Possible values: - prompt : prompts the operator for a password - plain : the password in plain text, may contain any printable character defined by the US-ASCII character set and that has a decimal code in the range 33 to 126, and the length should be less than or equal to 64 (Due to legacy reasons, the MD5 character string was included in the parameter name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication password.) Field type <Security::PlainPassword4> - the password, may contain any printable character defined by the US-ASCII character set and that has a decimal code in the range 33 to 126, and the length should be less than or	*optional parameter with default value: "plain : "* The password of the ISDN PRI / CAS SIP termination

Parameter	Type	Description
	equal to 64 (Due to legacy reasons, the MD5 character string was included in the parameter name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication password.) - length: x<=64	
[no] aka-secret-key	Parameter type: <Security::Password8> Format: (prompt | plain : <Security::PlainPassword8>) Possible values: - prompt : prompts the operator for AKA shared secret key - plain : the AKA shared secret key in plain text, each character should be 0-f, and the length can only be 0 or 32 Field type <Security::PlainPassword8> - the AKA shared secret key, each character should be 0-f, and the length can only be 0 or 32 - length: x<=32	*optional parameter with default value: "plain : "* The AKA shared secret key of the ISDN PRI / CAS SIP termination
[no] admin-status	Parameter type: <Sip::IsdnTermAdminStatus> Format: (up | down) Possible values: - up : unlock the sip isdn/cas termination - down : lock the sip isdn/cas termination	*optional parameter with default value: "down"* The administrative status of the ISDN PRI / CAS SIP termination.
[no] line-id	Parameter type: <Sip::PriTksgTermLineId> Format: - The line-id allows to configure the Fax Mode(fm) and/or the Voice Activity Detection(vad) and/or the processing procedure for the dtmf event tones(2833) and/or the Vbd-mode (vm) and/or the UDP port usage for T38 to voice switching (udp) and/or the Pani header line ID.\n Overall syntax: \n [fm:(value(1:3))][;][vad:e|vad:d][;][2833:(value(1:4))][;][vm:(value(1:4))][;][udp:e|udp:d][;][(Pani header line-id value)]\n A semi-colon must be inserted between the configuration input strings. The configuration input must be given in lower case characters. No space allowed anywhere; no leading zero(es) allowed anywhere.\n In case the configuration input for one or multiple input strings must be changed, the configuration input for all other input strings must be repeated too i.e. the new configuration input for the line id parameter fully overwrites the previous configuration input.\n - Keyword : fm\n Values : range [1:3]\n 1 : g711vbd \n 2 : t.38-with-g711vbd-fallback-with-signaling \n 3 : t.38-with-g711vbd-fallback-without-signaling\n In case the keyword is not found in the configuration input of lind-id parameter, the system wide pre-provisioned value in the CDE profile file applies.\n - Keyword : vad\n Values : e | d\n e : vad is enabled (sending comfort noise packets during silent periods)\n d : vad is disabled (do not send comfort noise packets during silent periods)\n In case the keyword is not found in the configuration input of the line-id parameter, the system wide pre-provisioned value in the CDE profile file applies.\n - Keyword : 2833\n Values : range [1:4]\n 1 : the processing procedure is audio\n 2 : the processing	*optional parameter with default value: ""* The format of the line ID to be included in the PANI-HEADER.

Parameter	Type	Description
	procedure is rfc2833\n 3 : the processing procedure is both, audio and rfc2833\n 4 : audio has priority in the processing procedure\n In case the keyword is not found in the configuration input of the line-id parameter, the system wide configured value in the sipVspTable applies.\n - Keyword : vm\n Values : range [1:4]\n 1 : auto switch without negotiation\n 2 : renegotiation\n 3 : If V152 negotiation fails, autoswitch applies(not support now)\n 4 : If V152 negotiation fails, renegotiation applies(not support now)\n In case the keyword is not found in the configuration input of the line-id parameter, the system wide configured value in the sipVspTable applies.\n - Keyword : udp\n Values : e | d\n e : the same UDP port is used for Voice and T38\n d : a different UDP port is used for Voice and T38\n In case the keyword is not found in the configuration input of the line-id parameter, the system wide configured value in the sipVspTable applies.\n - Pani header line id :\n Must always appear at the end of the configuration string.\n The configuration input is not bounded to any format i.e. It is the responsibility of the customer to define the desired line ID format; There is no restriction on the character set being used. - length: x<=63	
[no] provider-name	Parameter type: <Sip::TermUserProviderName> Format: - identify the user VSG name for this port - length: x<=32	*optional parameter with default value: "vsp1"* The name of the VSG the ISDN PRI /CAS SIP termination is associated with.
[no] ds0-hunting-way	Parameter type: <Sip::TermHuntingWay> Format: (cyclic-forward | sequential-forward | sequential-reverse | random-forward | cyclic-reverse | random-reverse) Possible values: - cyclic-forward : Initially the lowest B-channel is selected i.e. B-channel 1; afterwards, always again the next B-channel following the one used last is selected. - sequential-forward : The selection of the B-channel is based on an increasing port number, starting with the B-channel with the lowest port number of the E1/T1 interface and the next B-channel in the ordered series of B-channel is selected if the B-channel last used is busy. Again select the B-channel with the lowest port number. - sequential-reverse : The selection of the B-channel is based on an decreasing port number, starting with the B-channel with the highest port number of the E1/T1 interface and the next B-channel in the ordered series of B-channels is selected if the B-channel last used is busy. Again select the B-channel with the highest port number. - random-forward : allocation by random order. if the randomly selected B-channel is busy, the selection of a new candidate B-channel starts at the next in the ordered series of B-channels following the one been randomly selected, and follows increasing order.	*optional parameter with default value: "cyclic-forward"* The channel selection method. For ISDN PRA, it applies both outgoing and incoming calls. For CAS, it applies to incoming calls only.

Parameter	Type	Description
	- cyclic-reverse : Initially the highest B-channel is selected i.e. B-channel 30; afterwards, always again the next B-channel following the one used last is selected. - random-reverse : allocation by random order. if the randomly selected B-channel is busy, the selection of a new candidate B-channel starts at the next in the ordered series of B-channels following the one been randomly selected, and follows decreasing order.	
[no] itf-hunting-way	Parameter type: <Sip::IsdnCasTerminationItfHuntingWay> Format: (cyclic-forward | sequential-forward-fl | sequential-reverse-fl | random-forward | cyclic-reverse | random-reverse | sequential-forward-ud | sequential-reverse-ud) Possible values: - cyclic-forward : The selection of the E1/T1 interfaces is a cyclic process in which the different E1/T1 interface are alternately selected.At the first selection attempt the first E1/T1 in the configuration input will be selected and If no free channel on this interface the selection proceeds with the succeeding interface (i.e. in forward order) in the configuration order.For any subsequent selection attempt, the selection process will start at the interface which is succeeding the one in the configuration order that was selected last for the previous attempt (i.e. starting point proceeds in forward order) - sequential-forward-fl : Each time always start from the first E1/T1 in the configuration input, will select the next E1/T1 in the order configured only when all B channels of current E1/T1 are busy. - sequential-reverse-fl : Each time always start from the last E1/T1 in the configuration input, will select the preceding E1/T1 in the order configured only when all B channels of current E1/T1 are busy. - random-forward : The selection of the E1/T1 interfaces happens randomly. If selected E1/T1 interface has no idle channel, then take the next E1/T1 interface(forward) - cyclic-reverse : The selection of the E1/T1 interfaces is a cyclic process in which the different E1/T1 interface are alternately selected.At the first selection attempt the last E1/T1 in the configuration input will be selected and If no free channel on this interface the selection proceeds with the preceding interface (i.e in reverse order) in the configuration order.For any subsequent selection attempt, the selection process will start at the interface which is preceding the one in the configuration order that was selected last for the previous attempt (i.e starting point proceeds in reverse order) - random-reverse : The selection of the E1/T1 interfaces happens randomly. If selected E1/T1 interface has no idle channel, then take the next E1/T1 interface(reverse) - sequential-forward-ud : Aiming at having a equal load (more or less equal number of busy B-channels for each of	*optional parameter with default value: "cyclic-forward"* The E1/T1 interface selection method (for incoming call attempt)

Parameter	Type	Description
	the E1/T1 interfaces in the E1/T1 cluster. MIB readiness only. Unsupport now - sequential-reverse-ud : Aiming at having a equal load (more or less equal number of busy B-channels for each of the E1/T1 interfaces in the E1/T1 cluster. MIB readiness only. Unsupport now	
[no] sip-rule-set	Parameter type: <Sip::IsdnCasManipulationTksgRuleSet> Format: - the ISDN-PRA/CAS-R2 manipulation rule set, at most 32 rule ids can be configured for ISDN-PRA term and CAS-R2 term - length: x<=32	*optional parameter with default value: ""* The SIP termination specific rule set that applies to this ISDN PRI / CAS SIP termination.
[no] q-value	Parameter type: <Sip::IsdnCasTerminationQValue> Format: - The q value is a number specified as parameter in the Contact header field, denotes an order of preference - range: [0...100]	*optional parameter with default value: "100"* The q value for in case SIP Forking applies
[no] reg-mode	Parameter type: <Sip::IsdnCasTerminationRegMode> Format: (explicit | implicit | no-reg) Possible values: - explicit : The Explicit register (ER) mode is only relevant in case a list of individual DDI-EXTs has been manually configured and associated with this SIP termination - implicit : The Implicit register (IR) mode, the list of implicit registered DDI-EXTs is returned by the register server in the 200 OK response to the SIP register request - no-reg : The no-registeration mode is only relevant in case the sip UA does not have to send any sip REGISTER / re-REGISTER request to the sipserver nor for the GDN, nor for the optional manually configured DDI-EXT.	*optional parameter with default value: "implicit"* The register mode : implicit registration, explicit registration, no-registration
[no] extension	Parameter type: <Sip::SipPriCasTksgTermExtension> Format: - The ifindex of interfaces.At maximum 32 ifIndexs can be configured.The ifindex used as index of the ISDN PRI/CAS SIP termination must be configured in SipPriCasTksgExtension object.The order of the E1/T1 interface ifindices in the configuration input for the SipPriCasTksgExtension object defines the priority of the E1/T1 interface selection.	*optional parameter with default value: ""* The list of E1/T1 interfaces that composes the Trunk (sub-)group.
trunk-group-name	Parameter type: <Sip::SipPriCasTksgTermTrunkGroupName> Format: - Name of the E1/T1 Trunk Group - length: x<=32	*mandatory parameter* The name of the E1/T1 Trunk Group
[no] signalling-type	Parameter type: <Sip::TksgTermSignallingType> Format: (isdn-pra | cas-2bit-mfc-gr-fw-bw | cas-2bit-dtmf | cas-1bit-dtmf | cas-1bit-mfc-gr-fw-bw | cas-1bit-decadic | cas-1bit-mfc-fw) Possible values:	*optional parameter with default value: "isdn-pra"* The signaling type for this SIP termination.

Parameter	Type	Description
	- isdn-pra : Q931/Q921 protocol based communication between the CPE side and the NIAT-A board - cas-2bit-mfc-gr-fw-bw : 2 bit line signaling and MFC register signaling in compliance with ITU-T Q.400-490 CAS R2 specification (supporting groups, forward and backward signals) - cas-2bit-dtmf : 2 bit line signaling and DTMF register signaling - cas-1bit-dtmf : 1 bit line signaling and DTMF register signaling - cas-1bit-mfc-gr-fw-bw : 1 bit line signaling and MFC register signaling (supporting groups, forward and backward signals). This ENUM value is actually NOT supported. - cas-1bit-decadic : 1 bit line signaling and DECADIC register signaling. This ENUM value is actually NOT supported. - cas-1bit-mfc-fw : 1 bit line signaling and MFC register signaling in compliance with ITU-T Q.310-Q322 CAS R1 specification (supporting only forward signals). This ENUM value is actually NOT supported.	
[no] cas-cpn-screen	Parameter type: <Sip::TermCallingScreen> Format: (disable | enable) Possible values: - disable : disable calling party number screening - enable : enable calling party number screening	*optional parameter with default value: "enable"* To enable / disable the screening of the calling party number for calls originating at the CPE side.It applies to the CAS SIP termination only.
[no] cas-default-cpc	Parameter type: <Sip::TermDefaultCPC> Format: (passthrough | ordinary | specchg | test | payphone | operator | data | payphoneiu | atme) Possible values: - passthrough : bypass the cpc from PBX - ordinary : replace the cpc from PBX by ordinary - specchg : replace the cpc from PBX by specchg - test : replace the cpc from PBX by test - payphone : replace the cpc from PBX by payphone - operator : replace the cpc from PBX by operator - data : replace the cpc from PBX by data. data is not supported right now. - payphoneiu : replace the cpc from PBX by payphoneIU - atme : replace the cpc from PBX by atme	*optional parameter with default value: "passthrough"* The calling party's category for this SIP termination. In case a CAS SIP termination is expected to request the calling party number and category from the CPE, the received calling party's category shall be replaced by the configured category. It applies to the CAS SIP termination only.
[no] cas-restrict-acc	Parameter type: <Sip::TermRestrictACC> Format: (disable | enable) Possible values: - disable : no restriction for terminated ACC - enable : restriction for terminated ACC	*optional parameter with default value: "disable"* To enable/disable whether the CPE is restricted for terminated Automatic Collect Call. It applies to the CAS SIP termination only.
[no] cas-exch-type	Parameter type: <Sip::TermExchangeType>	*optional parameter with default*

Parameter	Type	Description
	Format: (normal | compelled-a3-and-b1 | pulsed-a3-and-b1) Possible values: - normal : the exchange type is normal - compelled-a3-and-b1 : the exchange can only handle B-1 and compelled A3 when digit completed - pulsed-a3-and-b1 : the exchange can only handle B-1 and pulsed A3 when digit completed	*value: "normal"* To configure the exchange type : - the CPE supports all, the group A and the group B signals - compelled-a3-and-b1 - the CPE supports only the B-1 signal. Any other B-x signal sent to the CPE is ignored. - pulsed-a3-and-b1 - the CPE supports only the B-1 signal. A pulsed A-3 signal is sent to the CPE after DML analysis has completed and a B-1 signal is sent to the CPE after a SIP INVITE has been sent to the SIP server. It applies to the CAS SIP termination only.
[no] isdn-nsm-profile	Parameter type: <Sip::NsmProfile> Format: (none | <Sip::NsmProfilePointer> | name : <PrintableString>) Possible values: - none : no ISDN NSM profile assigned. - name : ISDN NSM profile name Field type <Sip::NsmProfilePointer> - the index of the ISDN NSM profile. - range: [1...32] Field type <PrintableString> - printable string	*optional parameter with default value: "none"* The nsm profile assigned to the ISDN PRI SIP termination.
[no] cas-nsm-profile	Parameter type: <Sip::CasNsmProfPointer> Format: (none | <Sip::CasNsmProfPointer> | name : <PrintableString>) Possible values: - none : no CAS NSM profile assigned. - name : CAS NSM profile name Field type <Sip::CasNsmProfPointer> - the index of the CAS NSM profile. - range: [1...32] Field type <PrintableString> - printable string	*optional parameter with default value: "none"* The cas nsm profile assigned to the CAS SIP termination.
[no] uus-profile	Parameter type: <Sip::UserToUserProfile> Format: (none | <Sip::UserToUserProfilePointer> | name : <PrintableString>) Possible values: - none : no ISDN user-to-user profile assigned. - name : ISDN user-to-user profile name Field type <Sip::UserToUserProfilePointer> - the index of the user-to-user profile. - range: [1...32] Field type <PrintableString> - printable string	*optional parameter with default value: "none"* The user-to-user service profile assigned to the ISDN PRI SIP termination.
[no] cas-timer-profile	Parameter type: <Sip::CasTimerProfPointer> Format:	*optional parameter with default value: "none"*

Parameter	Type	Description
	(none | <Sip::CasTimerProfPointer> | name : <PrintableString>) Possible values: - none : no CAS timer profile assigned. - name : CAS timer profile name Field type <Sip::CasTimerProfPointer> - the index of the CAS timer profile. - range: [1...32] Field type <PrintableString> - printable string	The cas timer profile assigned to the CAS SIP termination.

40.23 Voice Sip Manipulation Rule Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip manipulation rule profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no manipulate-rule (rule-id)) | (manipulate-rule (rule-id) [no admin-status | admin-status <Sip::RuleAdminStatus>] [no rule-name | rule-name <Sip::RuleName>] rule <Sip::ManipulateRule> [no rule-type | rule-type <Sip::ManipulateRuleType>])

Command Parameters

Table 40.23-1 "Voice Sip Manipulation Rule Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(rule-id)	Format: - the index of one manipulation rule - range: [1...255]	uniquely identify of this sip manipulation rule

Table 40.23-2 "Voice Sip Manipulation Rule Configuration Commands" Command Parameters

Parameter	Type	Description
[no] admin-status	Parameter type: <Sip::RuleAdminStatus> Format: (enabled | disabled) Possible values: - enabled : the rule can be executed - disabled : the rule can't be executed	*optional parameter with default value: "disabled"* administrative status of this sip manipulation rule
[no] rule-name	Parameter type: <Sip::RuleName> Format: - The operator friendly name of the rule - length: x<=32	*optional parameter with default value: ""* The operator friendly name of the rule
rule	Parameter type: <Sip::ManipulateRule> Format: - The supported rule key phrases are listed below. Only 1 rule key phrase can be configured per rule.\n-[ISDN]outg-use-default-pbx-number:[value]\n-[ISDN]inc-add-phone-context-to-number\n-[ISDN]outg-add-country-area-code-in-phone-context:[countrycode]-[areacode]\n-	*mandatory parameter* The manipulation rule

Parameter	Type	Description
	[ISDN/CAS]inc-short-number-len:[shortnumberlength]\n- [ISDN]outg-add-domain-name-in-phone-context\n - [POTS]out-dialedprefix-outprefix:[dialedprefix]-[outprefix]\n- [POTS/ISDN/CAS]outg-call-duration-limitation:[Prefix(es)]-[MaxCallDuration]-[MaxToneDurationTi [POTS/ISDN/CAS]inc-call-duration-limitation:[MaxCallDuration]-[MaxToneDurationTimer]\n- [POTS/ISDN]emergency-mode\n- [POTS/ISDN]priority-user\n- [CAS]outg-add-country-area-code-in-cpn:[countrycode]-[areacode]\n- [CAS]inc-del-local-country-code-in-cpn:[countrycode]\n For further details about the value ranges and meaning of the different values, please have a look at the description of the sipRuleTable in the SIP-MIB. - length: 1<=x<=64	
[no] rule-type	Parameter type: <Sip::ManipulateRuleType> Format: (pots | isdn-pra | pots-all | isdn-pra-all | cas-r2 | cas-r2-all) Possible values: - pots : this rule is applied to POTS term - isdn-pra : this rule is applied to ISDN PRA term - pots-all : this rule is applied to all POTS terms - isdn-pra-all : this rule is applied to all ISDN PRA terms - cas-r2 : this rule is applied to CAS term - cas-r2-all : this rule is applied to all CAS terms	*optional parameter with default value: "isdn-pra"* The manipulation rule type

40.24 Voice Sip Cas R2 Timer Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip Cas R2 timer profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip cas-r2-timer (if-index) [no seizure-ack | seizure-ack <Sip::CasR2TimerSeizureAck>] [no disconnect-ack | disconnect-ack <Sip::CasR2TimerDisconnectAck>] [no force-rel-ack | force-rel-ack <Sip::CasR2TimerForceReleaseAck>] [no pres-fwd-sig | pres-fwd-sig <Sip::CasR2TimerPreFwdSig>] [no recept-fwd-sig | recept-fwd-sig <Sip::CasR2TimerRecpFwdSig>] [no re-answer | re-answer <Sip::CasR2TimerReanswer>] [no call-process | call-process <Sip::CasR2TimerCallProcess>]

Command Parameters

Table 40.24-1 "Voice Sip Cas R2 Timer Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(if-index)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::PortId> - the port number	The unique internal identifier of the termination port

Table 40.24-2 "Voice Sip Cas R2 Timer Configuration Commands" Command Parameters

Parameter	Type	Description
[no] seizure-ack	Parameter type: <Sip::CasR2TimerSeizureAck> Format: - Timer for the reception of the SEIZURE ACK Signal to the TDM R2 side. - unit: millisecond - range: [100...1000]	*optional parameter with default value: "200"* Timer for the reception of the SEIZURE ACK Signal to the TDM R2 side
[no] disconnect-ack	Parameter type: <Sip::CasR2TimerDisconnectAck> Format:	*optional parameter with default value: "90"*

Parameter	Type	Description
	- Timer for the reception of the DISCONNECTION ACK Signal to the TDM R2 side. - unit: second - range: [1...360]	Timer for the reception of the DISCONNECTION ACK Signal to the TDM R2 side.
[no] force-rel-ack	Parameter type: <Sip::CasR2TimerForceReleaseAck> Format: - Timer for the reception of the FORCED RELEASE ACK / CLEAR BACK ACK Signal to the TDM R2 side. - unit: second - range: [1...360]	*optional parameter with default value: "90"* Timer for the reception of the FORCED RELEASE ACK / CLEAR BACK ACK Signal to the TDM R2 side.
[no] pres-fwd-sig	Parameter type: <Sip::CasR2TimerPreFwdSig> Format: - Timer that defines the interval for the presence of the FORWARDING Signal to the TDM R2 side. - unit: second - range: [1...36]	*optional parameter with default value: "15"* Timer that defines the interval for the presence of the FORWARDING Signal to the TDM R2 side.
[no] recept-fwd-sig	Parameter type: <Sip::CasR2TimerRecpFwdSig> Format: - Timer that defines the interval for the reception of the FORWARDING Signal to the TDM R2 side. - unit: second - range: [1...20]	*optional parameter with default value: "7"* Timer that defines the interval for the reception of the FORWARDING Signal to the TDM R2 side.
[no] re-answer	Parameter type: <Sip::CasR2TimerReanswer> Format: - The re-answer timer on terminated calls to the TDM R2 side. A value '0' configured for the re-answer timer means that the re-answer timer shall not be started. - unit: second - range: [0...360]	*optional parameter with default value: "90"* The re-answer timer on terminated calls to the TDM R2 side.
[no] call-process	Parameter type: <Sip::CasR2TimerCallProcess> Format: - Timer that defines the interval between digit collection completed and called party answering the call request. The timer is started for incoming call requests to the TDM R2 side. A value '0' configured for the call-process timer means that the call-process timer shall not be started. - unit: second - range: [0...360]	*optional parameter with default value: "0"* Timer that defines the interval between digit collection completed and called party answering the call request. The timer is started for incoming call requests to the TDM R2 side. A value '0' configured for the call-process timer means that the call-process timer shall not be started.

40.25 Voice Sip PUID Configuration Commands

Command Description

This command allows the operator to manage the PUID term.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no isdn-cas-cls-puid (name) directory-number <Sip::TermDnumber>) | (isdn-cas-cls-puid (name) directory-number <Sip::TermDnumber> [no user-name | user-name <Sip::TermUserName>] [no uri | uri <Sip::TermUri>] [no md5-password | md5-password <Security::Password4>] [no aka-secret-key | aka-secret-key <Security::Password8>] [no display-name | display-name <Sip::TermUserName>])

Command Parameters

Table 40.25-1 "Voice Sip PUID Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(name)	Format: - Name of the E1 cluster - length: x<=32	Name of E1 or T1 cluster
directory-number	Parameter type: <Sip::TermDnumber> Format: - identify the DN of the subscriber line - length: x<=32	The directory number of this PUID

Table 40.25-2 "Voice Sip PUID Configuration Commands" Command Parameters

Parameter	Type	Description
[no] user-name	Parameter type: <Sip::TermUserName> Format: - identify the user name for this port - length: x<=64	*optional parameter with default value: ""* Identify the user name for this PUID
[no] uri	Parameter type: <Sip::TermUri> Format: - SIP uri by which the user agent identifies subscriber line - length: x<=80	*optional parameter with default value: ""* SIP URI by which user agent identifies PUID
[no] md5-password	Parameter type: <Security::Password4> Format: (prompt | plain : <Security::PlainPassword4>) Possible values: - prompt : prompts the operator for a password - plain : the password in plain text, may contain any printable character defined by the US-ASCII character set	*optional parameter with default value: "plain : "* The password associated with the user

Parameter	Type	Description
	and that has a decimal code in the range 33 to 126, and the length should be less than or equal to 64 (Due to legacy reasons, the MD5 character string was included in the parameter name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication password.) Field type <Security::PlainPassword4> - the password, may contain any printable character defined by the US-ASCII character set and that has a decimal code in the range 33 to 126, and the length should be less than or equal to 64 (Due to legacy reasons, the MD5 character string was included in the parameter name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication password.) - length: x<=64	
[no] aka-secret-key	Parameter type: <Security::Password8> Format: (prompt | plain : <Security::PlainPassword8>) Possible values: - prompt : prompts the operator for AKA shared secret key - plain : the AKA shared secret key in plain text, each character should be 0-f, and the length can only be 0 or 32 Field type <Security::PlainPassword8> - the AKA shared secret key, each character should be 0-f, and the length can only be 0 or 32 - length: x<=32	*optional parameter with default value: "plain : "* The AKA shared secret key
[no] display-name	Parameter type: <Sip::TermUserName> Format: - identify the user name for this port - length: x<=64	*optional parameter with default value: ""* Display name for this PUID

40.26 Voice Sip PUID Configuration Commands

Command Description

This command allows the operator to manage the DDI-EXTs of the ISDN PRI / CAS SIP termination

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no pri-cas-tksg-puid (if-index) tksg-id <Sip::TrunkSubGroupId> directory-number <Sip::TermDnumber>) | (pri-cas-tksg-puid (if-index) tksg-id <Sip::TrunkSubGroupId> directory-number <Sip::TermDnumber> [no user-name | user-name <Sip::TermUserName>] [no uri | uri <Sip::TermUri>] [no password | password <Security::Password4>] [no aka-secret-key | aka-secret-key <Security::Password8>] [no display-name | display-name <Sip::TermUserName>])

Command Parameters

Table 40.26-1 "Voice Sip PUID Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(if-index)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::PortId> - the port number	The ifindex of the related ISDN PRI / CAS SIP termination
tksg-id	Parameter type: <Sip::TrunkSubGroupId> Format: - The index of E1/T1 interface trunk sub-group. - range: [0...256]	The E1/T1 Trunk sub-group ID
directory-number	Parameter type: <Sip::TermDnumber> Format: - identify the DN of the subscriber line - length: x<=32	The directory number of the DDI-EXT

Table 40.26-2 "Voice Sip PUID Configuration Commands" Command Parameters

Parameter	Type	Description
[no] user-name	Parameter type: <Sip::TermUserName> Format: - identify the user name for this port	*optional parameter with default value: ""* The user name of the DDI-EXT

Parameter	Type	Description
	- length: x<=64	
[no] uri	Parameter type: <Sip::TermUri> Format: - SIP uri by which the user agent identifies subscriber line - length: x<=80	*optional parameter with default value: ""* The SIP URI by which the SIP user agent identifies the DDI-EXT
[no] password	Parameter type: <Security::Password4> Format: (prompt | plain : <Security::PlainPassword4>) Possible values: - prompt : prompts the operator for a password - plain : the password in plain text, may contain any printable character defined by the US-ASCII character set and that has a decimal code in the range 33 to 126, and the length should be less than or equal to 64 (Due to legacy reasons, the MD5 character string was included in the parameter name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication password.) Field type <Security::PlainPassword4> - the password, may contain any printable character defined by the US-ASCII character set and that has a decimal code in the range 33 to 126, and the length should be less than or equal to 64 (Due to legacy reasons, the MD5 character string was included in the parameter name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication password.) - length: x<=64	*optional parameter with default value: "plain : "* The password of the DDI-EXT
[no] aka-secret-key	Parameter type: <Security::Password8> Format: (prompt | plain : <Security::PlainPassword8>) Possible values: - prompt : prompts the operator for AKA shared secret key - plain : the AKA shared secret key in plain text, each character should be 0-f, and the length can only be 0 or 32 Field type <Security::PlainPassword8> - the AKA shared secret key, each character should be 0-f, and the length can only be 0 or 32 - length: x<=32	*optional parameter with default value: "plain : "* The AKA shared secret key of the DDI-EXT
[no] display-name	Parameter type: <Sip::TermUserName> Format: - identify the user name for this port - length: x<=64	*optional parameter with default value: ""* Display name of this DDI-EXT

40.27 Voice Sip Isdn Nsm Profile Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip Isdn nsm Profile profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no isdn-nsm-prof (prof-name)) | (isdn-nsm-prof (prof-name) [no version-nbr | version-nbr <Sip::NsmProfileVersionNbr>] [no outg-from-no-cgpn | outg-from-no-cgpn <Sip::NsmOutFromNoCgpn>] [no outg-privacy | outg-privacy <Sip::NsmOutPricy>] [no outg-from-pi | outg-from-pi <Sip::NsmOutFromPi>] [no inc-from-noclip | inc-from-noclip <Sip::NsmIncFromNoClip>] [no inc-from-clip-pri | inc-from-clip-pri <Sip::NsmIncFromClipPri>] [no inc-from-clip-ua | inc-from-clip-ua <Sip::NsmIncFromClipUa>] [no inc-npi-to-cpn | inc-npi-to-cpn <Sip::NsmIncNpiToCpn>] [no inc-ton-to-cpn | inc-ton-to-cpn <Sip::NsmIncTonToCpn>] international-prefix <Sip::NsmProfInPrefix> [no national-prefix | national-prefix <Sip::NsmProfNPrefix>] country-code <Sip::NsmProfCC> outg-cpn-length <Sip::NsmOutCpnLength> [no inc-cgpn-add-ac | inc-cgpn-add-ac <Sip::NsmIncCgpnAddAC>] [no inc-ton-to-cdpn | inc-ton-to-cdpn <Sip::NsmIncTonToCdPn>])

Command Parameters

Table 40.27-1 "Voice Sip Isdn Nsm Profile Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(prof-name)	Format: - the name of a nsm profile - length: x<=32	The unique name identifier of an row in the sipIsdnNsmProfileTable.

Table 40.27-2 "Voice Sip Isdn Nsm Profile Configuration Commands" Command Parameters

Parameter	Type	Description
[no] version-nbr	Parameter type: <Sip::NsmProfileVersionNbr> Format: - a version number of the profile - range: [1...65535]	*optional parameter with default value: 1* The version number to the profile.
[no] outg-from-no-cgpn	Parameter type: <Sip::NsmOutFromNoCgpn> Format: (impu-gdn | unavailable) Possible values: - impu-gdn : The From header is to be sent with the default group number for outgoing call	*optional parameter with default value: "impu-gdn"* This object allows indicating whether the From-header user part of an outgoing SIP Invite request needs to be set to 'Unavailable' or to include the

Parameter	Type	Description
	- unavailable : The From header is to be sent as unavailable@unknown.invalid for outgoing call	default group number
[no] outg-privacy	Parameter type: <Sip::NsmOutPricy> Format: (privacy-id-hdr-usr | privacy-none) Possible values: - privacy-id-hdr-usr : Privacy header included with value id, header, user - privacy-none : Privacy header included with value none	*optional parameter with default value: "privacy-id-hdr-usr"* this object allows to indicate whether a privacy header field with value 'id, header, user' or a privacy header with value 'none' is to be included when CgPN IE received with NPI other than 'ISDN/telephony numbering plan' or 'unknown' or CgPN IE received with NPI is 'ISDN/telephony numbering plan' or 'unknown' and PI=absent.
[no] outg-from-pi	Parameter type: <Sip::NsmOutFromPi> Format: (anonymous | not-anonymous) Possible values: - anonymous : the From header shall be set to: From: Anonymous sip:anonymous@anonymous.invalid - not-anonymous : the setting of the sipIsdnNsmProfileOutFromNoInvNbr object applies.	*optional parameter with default value: "not-anonymous"* This object allows to indicate whether the 'From' header is to be anonymized or not, when PI = Presentation restricted is received.
[no] inc-from-noclip	Parameter type: <Sip::NsmIncFromNoClip> Format: - the string value that IMS will send in the 'From' header user info in case the user does not have the CLIP feature assigned - length: x<=32	*optional parameter with default value: "unsubscribed"* This object allows to configure the string value the IMS will send in the 'From' header user info in case the terminating user does not have the CLIP feature assigned.
[no] inc-from-clip-pri	Parameter type: <Sip::NsmIncFromClipPri> Format: - the string value that IMS will send in the 'From' header user info in case the CLI has to be kept private - length: x<=32	*optional parameter with default value: "anonymous"* This object allows to configure the string value the IMS will send in the 'From' header user info in case the CLI has to be kept private.
[no] inc-from-clip-ua	Parameter type: <Sip::NsmIncFromClipUa> Format: - the string value that IMS will send in the 'From' header user info in case there's no CLI that can be passed to the MSAN - length: x<=32	*optional parameter with default value: "unavailable"* This object allows to configure the string value the IMS will send in the 'From' header user info in case no CLI is available.
[no] inc-npi-to-cpn	Parameter type: <Sip::NsmIncNpiToCpn> Format: (itu-t-e164 | unknown) Possible values: - itu-t-e164 : the NPI is set to ISDN/telephony numbering plan (ITU-T Recommendation E.164) for incoming call - unknown : the NPI is set to unknown for incoming call	*optional parameter with default value: "unknown"* This object allows to configure how the numbering plan identifier field must be coded towards the terminating ISDN PRI user.
[no] inc-ton-to-cpn	Parameter type: <Sip::NsmIncTonToCpn> Format: (international | unknown) Possible values:	*optional parameter with default value: "unknown"* This object allows to configure the preferred value of the TON (Type Of Number) parameter of

Parameter	Type	Description
	- international : the TON is set to international for incoming call - unknown : the TON is set to unknown for incoming call	CgPN IE that is sent to the terminating ISDN PRI user.
international-prefix	Parameter type: <Sip::NsmProfInPrefix> Format: - the string value of the international prefix - length: 1<=x<=8	*mandatory parameter* The international prefix. Mandatory parameter
[no] national-prefix	Parameter type: <Sip::NsmProfNPrefix> Format: - the string value of the national prefix - length: x<=4	*optional parameter with default value: ""* The national prefix.
country-code	Parameter type: <Sip::NsmProfCC> Format: - the string value of the country code - length: 1<=x<=8	*mandatory parameter* The country code. Mandatory parameter
outg-cpn-length	Parameter type: <Sip::NsmOutCpnLength> Format: - The length of digits - range: [0...15]	*mandatory parameter* The number of digits that allows the MSAN to distinguish whether the phone number received in the CgPN is a shortened phone number (a DDI extension) or a phone number with a national or international format. Mandatory parameter.
[no] inc-cgpn-add-ac	Parameter type: <Sip::NsmIncCgpnAddAC> Format: (disable | enable) Possible values: - disable : Do not add area code to CgPN when intra-area call received - enable : Add area code to CgPN when intra-area call received	*optional parameter with default value: "disable"* This object allows to configure whether to add national prefix and area code to CgPN IE when receive an intra-area call.
[no] inc-ton-to-cdpn	Parameter type: <Sip::NsmIncTonToCdPn> Format: (unknown | international | national) Possible values: - unknown : the TON of CdPN is set to unknown for incoming call - international : the TON of CdPN is set to international for incoming call - national : the TON of CdPN is set to national for incoming call	*optional parameter with default value: "unknown"* This object allows to configure the value of the TON (Type Of Number) parameter of CdPN IE that is sent to the terminating ISDN PRI user.

40.28 Voice Sip Cas Nsm Profile Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip Cas nsm Profile profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no cas-nsm-prof (prof-name)) | (cas-nsm-prof (prof-name) [no version-nbr | version-nbr <Sip::CasNsmProfVersionNbr>] [no outg-from-no-cgpn | outg-from-no-cgpn <Sip::CasNsmProfOutFromNoInvNbr>] international-prefix <Sip::CasNsmProfInPrefix> [no national-prefix | national-prefix <Sip::CasNsmProfNPrefix>] country-code <Sip::CasNsmProfCC> outg-cpn-length <Sip::CasNsmProfCgPnNbrLength>)

Command Parameters

Table 40.28-1 "Voice Sip Cas Nsm Profile Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(prof-name)	Format: - the name of a nsm profile - length: 1<=x<=32	The unique name identifier of an row in the sipCasNsmProfileTable.

Table 40.28-2 "Voice Sip Cas Nsm Profile Configuration Commands" Command Parameters

Parameter	Type	Description
[no] version-nbr	Parameter type: <Sip::CasNsmProfVersionNbr> Format: - a version number of the profile - range: [1...65535]	*optional parameter with default value: 1* The version number to the profile.
[no] outg-from-no-cgpn	Parameter type: <Sip::CasNsmProfOutFromNoInvNbr> Format: (impu-gdn | unavailable) Possible values: - impu-gdn : The From header is to be sent with the default group number for outgoing call - unavailable : The From header is to be sent as unavailable@unknown.invalid for outgoing call	*optional parameter with default value: "impu-gdn"* This object allows indicating whether the From-header user part of an outgoing SIP Invite request needs to be set to 'Unavailable' or to include the default group number
international-prefix	Parameter type: <Sip::CasNsmProfInPrefix> Format: - the string value of the international prefix	*mandatory parameter* The international prefix. Mandatory parameter

Parameter	Type	Description
	- length: 1<=x<=8	
[no] national-prefix	Parameter type: <Sip::CasNsmProfNPrefix> Format: - the string value of the national prefix - length: x<=4	*optional parameter with default value: ""* The national prefix.
country-code	Parameter type: <Sip::CasNsmProfCC> Format: - the string value of the country code - length: 1<=x<=8	*mandatory parameter* The country code. Mandatory parameter
outg-cpn-length	Parameter type: <Sip::CasNsmProfCgPnNbrLength> Format: - the length of digits - range: [0...15]	*mandatory parameter* The number of digits that allows the MSAN to distinguish whether the phone number received in the CgPN is a shortened phone number (a DDI extension) or a phone number with a national or international format. Mandatory parameter.

40.29 Voice Sip Pri User To User Profile Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip Pri User To User Profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no user-to-user-prof (prof-name)) | (user-to-user-prof (prof-name) [no version-nbr | version-nbr <Sip::UserToUserProfVersionNbr>] [no service-type | service-type <Sip::UserToUserProfServiceType>] [no act-estab-call | act-estab-call <Sip::UserToUserProfActivateEstablishCall>] [no act-stable-call | act-stable-call <Sip::UserToUserProfActivateStableCall>] [no uus3-t1-timer | uus3-t1-timer <Sip::UserToUserProfUUS3T1Timer>])

Command Parameters

Table 40.29-1 "Voice Sip Pri User To User Profile Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(prof-name)	Format: - The unique name identifier of a row in the sipPriUserToUserProfileTable - length: 1<=x<=32	The unique name identifier of a row in the sipPriUserToUserProfileTable.

Table 40.29-2 "Voice Sip Pri User To User Profile Configuration Commands" Command Parameters

Parameter	Type	Description
[no] version-nbr	Parameter type: <Sip::UserToUserProfVersionNbr> Format: - The version number of the pri user to user profile - range: [1...65535]	*optional parameter with default value: 1* The version number of the profile.
[no] service-type	Parameter type: <Sip::UserToUserProfServiceType> Format: uus3 Possible values: - uus3 : User-to-user information exchanged while a call is in the Active state	*optional parameter with default value: "uus3"* This object allows the operator to configure the UUS service type (UUS service 1, UUS service 2, UUS service 3). Only uus3 is supported.
[no] act-estab-call	Parameter type: <Sip::UserToUserProfActivateEstablishCall>	*optional parameter with default value: "false"*

Parameter	Type	Description
	Format: (true | false) Possible values: - true : the UUS service will be active during call establishment - false : the UUS service will not be active during call establishment	If this object is set as true(1), the UUS service will be active during call establishment; the SETUP message shall contain an independent UUS service request. The value true(1) can only be set when the act-stable-call is set to false(2).
[no] act-stable-call	Parameter type: <Sip::UserToUserProfActivateStableCall> Format: (true | false) Possible values: - true : the UUS service will be active during the active state of a call - false : the UUS service will not be active during the active state of a call	*optional parameter with default value: "false"* If this object is set as true(1), the UUS service will be active during the active state of a call; a FACILITY message indicating the UUS service request will be sent out for UUS service activation. The value true(1) can only be set when the act-estab-call is set to false(2).
[no] uus3-t1-timer	Parameter type: <Sip::UserToUserProfUUS3T1Timer> Format: - Value of UUS3-T1 timer - unit: second - range: [1...60]	*optional parameter with default value: 10* This object allows the operator to configure the UUS3-T1 timer. The timer is started when a FACILITY message indicating a User-To-user service 3 request is sent out and will be stopped when a response FACILITY message is received.

40.30 Voice Sip Isdn/Cas trunk sub-group Configuration Commands

Command Description

This command allows the operator to manage the E1/T1 trunk sub-group.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no tksg (if-index) tksg-id <Sip::TksgId> interface <Itf::IsdnLine>) | (tksg (if-index) tksg-id <Sip::TksgId> interface <Itf::IsdnLine> tksg-name <Sip::TksgName> ds0-bitmap <Sip::Ds0BitSet> [no admin-status | admin-status <Sip::IsdnTermAdminStatus>] [no ds0-hunting-way | ds0-hunting-way <Sip::Ds0HuntingWay>])

Command Parameters

Table 40.30-1 "Voice Sip Isdn/Cas trunk sub-group Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(if-index)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::PortId> - the port number	The ifIndex of the ISDN PRI / CAS SIP termination that has been created on top of the E1/T1 trunk sub-group.
tksg-id	Parameter type: <Sip::TksgId> Format: - The index of trunk sub-group - range: [1...256]	The unique Identifier of the trunk sub-group (chosen by the operator in the range [1..256]).
interface	Parameter type: <Itf::IsdnLine> Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId>	The Ifindex of the E1/T1 interface at which the trunk sub-group is created.

Resource Identifier	Type	Description
	- the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::PortId> - the port number	

Table 40.30-2 "Voice Sip Isdn/Cas trunk sub-group Configuration Commands" Command Parameters

Parameter	Type	Description
tksg-name	Parameter type: <Sip::TksgName> Format: - The operator friendly name of the trunk sub-group. - length: 1<=x<=32	*mandatory parameter* The operator friendly name of the trunk sub-group.
ds0-bitmap	Parameter type: <Sip::Ds0BitSet> Format: - The bitmap indicating the DS0 channels that belong to the trunk sub-group.	*mandatory parameter* The bitmap indicating the DS0 channels that belong to the trunk sub-group.
[no] admin-status	Parameter type: <Sip::IsdnTermAdminStatus> Format: (up | down) Possible values: - up : unlock the sip isdn/cas termination - down : lock the sip isdn/cas termination	*optional parameter with default value: "down"* The administrative status of the trunk sub-group.
[no] ds0-hunting-way	Parameter type: <Sip::Ds0HuntingWay> Format: (cyclic-forward | sequential-forward | sequential-reverse | random-forward | cyclic-reverse | random-reverse | inherit) Possible values: - cyclic-forward : DS0 channels are allocated cyclically.The preference is for the allocation of the subsequent idle channel in a cyclical manner.Initially the lowest DS0 channel is selected. i.e. DS0-channel 1;afterwards, always again the next DS0-channel following the one used last is selected. - sequential-forward : DS0 channels are allocated increasingly.The preference is for the allocation of the idle channel with the lowest number. - sequential-reverse : DS0 channels are allocated decreasingly.The preference is for the allocation of the idle channel with the highest number. - random-forward : DS0 channels are allocated randomly.The preference is for the allocation of the idle channel with the random number.If selected channel is busy, then Take the next free B-channel (forward) - cyclic-reverse : DS0 channels are allocated cyclically.Initially the highest DS0-channel is selected i.e. DS0-channel 30;afterwards, always again the preceding B-channel of the one used last is selected. - random-reverse : DS0 channels are allocated randomly.The preference is for the allocation of the idle channel with the random number.If selected channel is busy, then Take the	*optional parameter with default value: "inherit"* The hunting method for the DS0 channels of the trunk sub-group.

Parameter	Type	Description
	next free B-channel (reverse) - inherit : DS0 channels are allocated as configured in the corresponding ISDN PRI/CAS R2 SIP termination entry in the sipPriCasTksgTerminationDs0HuntingWay object of the sipPriCasTksgTerminationTable.	

40.31 Voice Sip Physical Interface Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip physical interface profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no phys-itf (if-index)) | (phys-itf (if-index) [no admin-status | admin-status <Sip::PhysItfAdminStatus>] [no interface-type | interface-type <Sip::PhysItfType>] [no signalling-type | signalling-type <Sip::PhysItfSignalingType>] [no e1-crc4frame | e1-crc4frame <Sip::PhysItfE1Crc4Framing>] [no t1-framemode | t1-framemode <Sip::PhysItfT1FrameMode>] [no t1-tx-linecode | t1-tx-linecode <Sip::PhysItfT1LineCode>] [no t1-rx-linecode | t1-rx-linecode <Sip::PhysItfT1LineCode>] [no t1-lbo-value | t1-lbo-value <Sip::PhysItfT1LBOValue>] [no blocking-bitmap | blocking-bitmap <Sip::PhysItfBlockingBitmap>] [no cap-name | cap-name <Sip::PhysItfCapName>] [no cas-sig-prof | cas-sig-prof <Sip::CasSignalProfPointer>])

Command Parameters

Table 40.31-1 "Voice Sip Physical Interface Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(if-index)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::PortId> - the port number	The unique internal identifier of the physical E1/T1 interface

Table 40.31-2 "Voice Sip Physical Interface Configuration Commands" Command Parameters

Parameter	Type	Description
[no] admin-status	Parameter type: <Sip::PhysItfAdminStatus> Format: (up | down | testing)	*optional parameter with default value: "down"* The administrative status of the physical E1/T1 interface.

Parameter	Type	Description
	Possible values: - up : unlock the E1/T1 physical interface - down : lock the E1/T1 physical interface - testing : the operator is going to run or is running loop-back testing on this interface	
[no] interface-type	Parameter type: <Sip::PhysItfType> Format: (e1-itf-type | t1-itf-type) Possible values: - e1-itf-type : E1 interface with 30 B channels. - t1-itf-type : T1 interface with 23 B channels.	*optional parameter with default value: "e1-itf-type"* The type of physical interface
[no] signalling-type	Parameter type: <Sip::PhysItfSignalingType> Format: (isdn-pra | cas-2bit-mfc-gr-fw-bw | cas-2bit-dtmf | cas-1bit-dtmf | cas-1bit-mfc-gr-fw-bw | cas-1bit-decadic | cas-1bit-mfc-fw) Possible values: - isdn-pra : Q931/Q921 protocol based communication between the CPE side and the NIAT-A board - cas-2bit-mfc-gr-fw-bw : 2 bit line signaling and MFC register signaling in compliance with ITU-T Q.400-490 CAS R2 specification (supporting groups, forward and backward signals) - cas-2bit-dtmf : 2 bit line signaling and DTMF register signaling - cas-1bit-dtmf : 1 bit line signaling and DTMF register signaling - cas-1bit-mfc-gr-fw-bw : 1 bit line signaling and MFC register signaling (supporting groups, forward and backward signals). This ENUM value is actually NOT supported. - cas-1bit-decadic : 1 bit line signaling and DECADIC register signaling. This ENUM value is actually NOT supported. - cas-1bit-mfc-fw : 1 bit line signaling and MFC register signaling in compliance with ITU-T Q.310-Q322 CAS R1 specification (supporting only forward signals). This ENUM value is actually NOT supported.	*optional parameter with default value: "isdn-pra"* The signaling type between the end-user (PBX) side and the NIAT-A board
[no] e1-crc4frame	Parameter type: <Sip::PhysItfE1Crc4Framing> Format: (cde-profile | enabled | disabled) Possible values: - cde-profile : default value defined in CDE. - enabled : CRC4 multiframe. - disabled : double frame.	*optional parameter with default value: "cde-profile"* The E1 interface CRC4 framing mode
[no] t1-framemode	Parameter type: <Sip::PhysItfT1FrameMode> Format: (cde-value | f4 | f12 | f24	*optional parameter with default value: "cde-value"* The Frame Mode of the T1 interface

Parameter	Type	Description
	| f72) Possible values: - cde-value : default value defined in CDE. - f4 : 4-frame multiframe. - f12 : 12-frame multiframe (D4). - f24 : 24-frame Superframe (ESF). - f72 : 72-frame multiframe (SLC96).	
[no] t1-tx-linecode	Parameter type: <Sip::PhysItfT1LineCode> Format: (cde-value | ami | ami-with-zcs | b8zs) Possible values: - cde-value : default value defined in CDE. - ami : Alternate Mark Inversion. - ami-with-zcs : Alternate Mark Inversion with Zero-Code Suppression. - b8zs : Bipolar with 8-zero Substitution.	*optional parameter with default value: "cde-value"* The Tx line code of the T1 interface
[no] t1-rx-linecode	Parameter type: <Sip::PhysItfT1LineCode> Format: (cde-value | ami | ami-with-zcs | b8zs) Possible values: - cde-value : default value defined in CDE. - ami : Alternate Mark Inversion. - ami-with-zcs : Alternate Mark Inversion with Zero-Code Suppression. - b8zs : Bipolar with 8-zero Substitution.	*optional parameter with default value: "cde-value"* The Rx line code of the T1 interface
[no] t1-lbo-value	Parameter type: <Sip::PhysItfT1LBOValue> Format: (lbo-0db | lbo-7-5db | lbo-15db) Possible values: - lbo-0db : LBO = 0dB, the resulting range of CI attenuation shall be 0 to 5.5 dB. - lbo-7-5db : LBO = 7.5dB, the resulting range of CI attenuation shall be 7.5 to 13.0 dB. - lbo-15db : LBO = 15dB, the resulting range of CI attenuation shall be 15 to 20.5 dB.	*optional parameter with default value: "lbo-0db"* The line build-out of the interface
[no] blocking-bitmap	Parameter type: <Sip::PhysItfBlockingBitmap> Format: - The bitmap indicating the TS channels that belong to the physical interface.	*optional parameter with default value: ""* The blocking/unblocking of a physical interface or TS channel (range)
[no] cap-name	Parameter type: <Sip::PhysItfCapName> Format: - The name of the Central Access Point that manages the E1/T1 interface - length: x<=31	*optional parameter with default value: ""* The name of the CAP
[no] cas-sig-prof	Parameter type: <Sip::CasSignalProfPointer> Format: (none | <Sip::CasSignalProfPointer>	*optional parameter with default value: "none"* The cas timer profile assigned to the CAS SIP termination.

Parameter	Type	Description
	| name : <PrintableString>) Possible values: - none : no CAS signaling profile assigned. - name : CAS signaling profile name Field type <Sip::CasSignalProfPointer> - the index of the CAS signaling profile. - range: [1...32] Field type <PrintableString> - printable string	

40.32 Voice Sip Cas Signaling Profile Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip Cas Signaling Profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no cas-sig-prof (prof-name)) | (cas-sig-prof (prof-name) [no version-nbr | version-nbr <Sip::CasSignalProfVersionNbr>] [no idle | idle <Sip::CasSignalProfIdle>] [no seize | seize <Sip::CasSignalProfSeize>] [no seize-ack | seize-ack <Sip::CasSignalProfSeizeAck>] [no answer | answer <Sip::CasSignalProfAnswer>] [no fw-disconnect | fw-disconnect <Sip::CasSignalProfForwardDisconnect>] [no fw-disconnect-ack | fw-disconnect-ack <Sip::CasSignalProfForwardDisconnectAck>] [no bw-disconnect | bw-disconnect <Sip::CasSignalProfBackwardDisconnect>] [no bw-disconnect-ack | bw-disconnect-ack <Sip::CasSignalProfBackwardDisconnectAck>] [no fault | fault <Sip::CasSignalProfFault>] [no block | block <Sip::CasSignalProfBlock>] [no meter | meter <Sip::CasSignalProfMeter>] [no reject-call | reject-call <Sip::CasSignalProfRejectCall>] [no bw-disc-reanswer | bw-disc-reanswer <Sip::CasSignalProfBackwardDisconnectWithReAnswer>])

Command Parameters

Table 40.32-1 "Voice Sip Cas Signaling Profile Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(prof-name)	Format: - name of the signal profile - length: 1<=x<=32	The unique name identifier of an row in the sipCasSignalingProfileTable.

Table 40.32-2 "Voice Sip Cas Signaling Profile Configuration Commands" Command Parameters

Parameter	Type	Description
[no] version-nbr	Parameter type: <Sip::CasSignalProfVersionNbr> Format: - version number of the signaling profile - range: [1...65535]	*optional parameter with default value: 1* The version number to the profile.
[no] idle	Parameter type: <Sip::CasSignalProfIdle> Format: - four signal bits(A/B/C/D) for line signal IDLE of CAS R2 or CAS R1 - range: [0...15]	*optional parameter with default value: "9"* Four signaling bits(A/B/C/D) for line signal IDLE of CAS R2 or CAS R1.
[no] seize	Parameter type: <Sip::CasSignalProfSeize> Format:	*optional parameter with default value: "1"*

Parameter	Type	Description
	- four signal bits(A/B/C/D) for line signal Seize of CAS R2 or CAS R1 - range: [0...15]	Four signaling bits(A/B/C/D) for line signal Seizure of CAS R2 or CAS R1.
[no] seize-ack	Parameter type: <Sip::CasSignalProfSeizeAck> Format: - four signal bits(A/B/C/D) for line signal Seize Ack of CAS R2 or CAS R1 - range: [0...15]	*optional parameter with default value: "13"* Four signaling bits(A/B/C/D) for line signal Seizure Ack of CAS R2 or CAS R1.
[no] answer	Parameter type: <Sip::CasSignalProfAnswer> Format: - four signal bits(A/B/C/D) for line signal Answer of CAS R2 or CAS R1 - range: [0...15]	*optional parameter with default value: "5"* Four signaling bits(A/B/C/D) for line signal Answer of CAS R2 or CAS R1.
[no] fw-disconnect	Parameter type: <Sip::CasSignalProfForwardDisconnect> Format: - four signal bits(A/B/C/D) for line signal Forward disconnect of CAS R2 or CAS R1 - range: [0...15]	*optional parameter with default value: "9"* Four signaling bits(A/B/C/D) for line signal Forward disconnect of CAS R2 or CAS R1.
[no] fw-disconnect-ack	Parameter type: <Sip::CasSignalProfForwardDisconnectAck> Format: - four signal bits(A/B/C/D) for line signal Forward disconnect Ack of CAS R2 or CAS R1 - range: [0...15]	*optional parameter with default value: "9"* Four signaling bits(A/B/C/D) for line signal Forward disconnect Ack of CAS R2 or CAS R1.
[no] bw-disconnect	Parameter type: <Sip::CasSignalProfBackwardDisconnect> Format: - four signal bits(A/B/C/D) for line signal Backward disconnect of CAS R2 or CAS R1 - range: [0...15]	*optional parameter with default value: "1"* Four signaling bits(A/B/C/D) for line signal Backward disconnect of CAS R2 or CAS R1.
[no] bw-disconnect-ack	Parameter type: <Sip::CasSignalProfBackwardDisconnectAck> Format: - four signal bits(A/B/C/D) for line signal Backward disconnect Ack of CAS R2 or CAS R1 - range: [0...15]	*optional parameter with default value: "9"* Four signaling bits(A/B/C/D) for line signal Backward disconnect Ack of CAS R2 or CAS R1.
[no] fault	Parameter type: <Sip::CasSignalProfFault> Format: - four signal bits(A/B/C/D) for line signal Fault of CAS R2 or CAS R1 - range: [0...15]	*optional parameter with default value: "13"* Four signaling bits(A/B/C/D) for line signal Fault of CAS R2 or CAS R1.
[no] block	Parameter type: <Sip::CasSignalProfBlock> Format: - four signal bits(A/B/C/D) for line signal Block of CAS R2 or CAS R1 - range: [0...15]	*optional parameter with default value: "13"* Four signaling bits(A/B/C/D) for line signal Block of CAS R2 or CAS R1.
[no] meter	Parameter type: <Sip::CasSignalProfMeter> Format: - four signal bits(A/B/C/D) for line signal Meter of CAS R2 or CAS R1 - range: [0...15]	*optional parameter with default value: "13"* Four signaling bits(A/B/C/D) for line signal Meter of CAS R2 or CAS R1.
[no] reject-call	Parameter type: <Sip::CasSignalProfRejectCall> Format: - four signal bits(A/B/C/D) for line signal Reject Call of CAS R2 or CAS R1 - range: [0...15]	*optional parameter with default value: "1"* Four signaling bits(A/B/C/D) for line signal Reject call of CAS R2 or CAS R1.
[no] bw-disc-reanswer	Parameter type: <Sip::CasSignalProfBackwardDisconnectWithReAnswer>	*optional parameter with default value: "13"*

Parameter	Type	Description
	Format: - four signal bits(A/B/C/D) for line signal backward_disconnect with_reanswer of CAS R2 or CAS R1 - range: [0...15]	Four signaling bits(A/B/C/D) for line signal backward_disconnect with_reanswer of CAS R2 or CAS R1.

40.33 Central Access Point Table Configuration Commands

Command Description

This command allows the operator to configure Central Access Point Table.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no cap (cap)) | (cap (cap) [no cap-name | cap-name <Sip::CapName>] [no ne-name | ne-name <Sip::NeName>] [no mode | mode <Sip::CapMode>] [no location | location <Sip::CapLocation>] [no admin-status | admin-status <Sip::CapAdminStatus>] [no ip | ip <SIP::IpAddressAndMask>] [no udp-port | udp-port <Sip::CapUDPPort>] [no gateway-ip | gateway-ip <Sip::CapGatewayIpAddr>] [no provider-name | provider-name <Sip::CapProviderName>])

Command Parameters

Table 40.33-1 "Central Access Point Table Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(cap)	Format: - the central access point id - range: [1...16]	The unique identifier of this central access point

Table 40.33-2 "Central Access Point Table Configuration Commands" Command Parameters

Parameter	Type	Description
[no] cap-name	Parameter type: <Sip::CapName> Format: - The name of the Central Access Point - length: x<=32	*optional parameter with default value: ""* The operator friendly name of this central access point
[no] ne-name	Parameter type: <Sip::NeName> Format: - the name of the network element - length: x<=32	*optional parameter with default value: ""* The operator friendly name of NE that hosts the central access point
[no] mode	Parameter type: <Sip::CapMode> Format: (enabled | disabled) Possible values: - enabled : cap mode enable for the ISDN PRI/CAS VoIP service	*optional parameter with default value: "disabled"* allows to configure whether the central access point mode is enabled or disabled

Parameter	Type	Description
	- disabled : cap mode disable for the ISDN PRI/CAS VoIP service	
[no] location	Parameter type: <Sip::CapLocation> Format: (local | remote) Possible values: - local : the value is set to local in cap ne - remote : the value is set to remote in remote ne	*optional parameter with default value: "local"* allows to configure whether the central access point is hosted in the local NE or hosted
[no] admin-status	Parameter type: <Sip::CapAdminStatus> Format: (up | down) Possible values: - up : the cap mode as configured becomes effective - down : the cap mode becomes not effective when cap mode enabled;non cap mode run in ISAMV NE when cap mode disabled	*optional parameter with default value: "down"* administrative status of this central access point
[no] ip	Parameter type: <SIP::IpAddressAndMask> Format: <Ip::V4Address> / <Sip::PrefixLength> Field type <Ip::V4Address> - IPv4-address Field type <Sip::PrefixLength> - prefix length of the subnet - range: [0...128]	*optional parameter with default value: "0.0.0.0/0"* the IPv4 address and prefix length of the Central Access Point
[no] udp-port	Parameter type: <Sip::CapUDPPort> Format: - The UDP port by which the cap can be addressed at the NVPS-C - range: [49152...65534]	*optional parameter with default value: "49152"* the UDP port by which the Central Access Point can be addressed via the proprietary communication protocol
[no] gateway-ip	Parameter type: <Sip::CapGatewayIpAddr> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* the Gateway IP address that allows the CAP to send internal communication packets to the remote NE
[no] provider-name	Parameter type: <Sip::CapProviderName> Format: - provider name - length: x<=32	*optional parameter with default value: "common"* The operator friendly name of the Voice Service Provider

40.34 Central Access Point Table Configuration Commands

Command Description

This command allows the operator to configure Central Access Point Table.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no ne-board cap-id <Sip::CapId> ne-id <Sip::NeId> board <Equipm::LtSlotIndex>) | (ne-board cap-id <Sip::CapId> ne-id <Sip::NeId> board <Equipm::LtSlotIndex> [no admin-status | admin-status <Sip::NeBoardAdminStatus>] xhub-port <Sip::xHubPort>)

Command Parameters

Table 40.34-1 "Central Access Point Table Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
cap-id	Parameter type: <Sip::CapId> Format: - the central access point id - range: [1...16]	The unique identifier of this central access point
ne-id	Parameter type: <Sip::NeId> Format: - the network element id - range: [1...8]	The unique identifier of the Network element.
board	Parameter type: <Equipm::LtSlotIndex> Format: (<Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> | <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId>) Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number	The slot identifier of the NIAT-A LT board.

Table 40.34-2 "Central Access Point Table Configuration Commands" Command Parameters

Parameter	Type	Description
[no] admin-status	Parameter type: <Sip::NeBoardAdminStatus> Format: (up	*optional parameter with default value: "down"* administrative status of this

Parameter	Type	Description
	| down) Possible values: - up : the NIAT-A LT board has been planned and has been physically pulled in - down : the NIAT-A LT board has not been planned yet or the NIAT-A LT board has been planned but is not physical plugged in.	NIAT-A LT board.
xhub-port	Parameter type: <Sip::xHubPort> Format: - The xHUB port to which the NIAT-A LT board is connected. - range: [1...32]	*mandatory parameter* The xHUB port to which the NIAT-A LT board is connected

40.35 Voice Sip Cas Timer Profile Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip Cas Timer Profile profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no cas-timer-prof (prof-name)) | (cas-timer-prof (prof-name) [no version-nbr | version-nbr <Sip::CasTimerProfVersionNbr>] appl-bitmap <Sip::CasTimerProfAppliBitmap> [no seize-ack | seize-ack <Sip::CasTimerProfSeizeAck>] [no clear-back-ack | clear-back-ack <Sip::CasTimerProfClearBwAck>] [no clear-fw-ack | clear-fw-ack <Sip::CasTimerProfClearFwAck>] [no presence-fw-sig | presence-fw-sig <Sip::CasTimerProfPresenceFwSig>] [no receipt-fw-sig | receipt-fw-sig <Sip::CasTimerProfReceiptFwSig>] [no re-answer | re-answer <Sip::CasTimerProfReAnswer>] [no call-process | call-process <Sip::CasTimerProfCallProcess>] [no delay-dial-min | delay-dial-min <Sip::CasTimerProfDelayDialingMinDur>] [no delay-dial-max | delay-dial-max <Sip::CasTimerProfDelayDialingMaxDur>] [no seize-state-time | seize-state-time <Sip::CasTimerProfSeizeStateTime>] [no reg-recall-min | reg-recall-min <Sip::CasTimerProfRegRecallMinDur>] [no reg-recall-max | reg-recall-max <Sip::CasTimerProfRegRecallMaxDur>] [no decadic-pulse-dur | decadic-pulse-dur <Sip::CasTimerProfDpDur>] [no decadic-break-dur | decadic-break-dur <Sip::CasTimerProfDpBreakDur>] [no inter-digit-peri | inter-digit-peri <Sip::CasTimerProfInterDigitPeriod>] [no ring-fw | ring-fw <Sip::CasTimerProfRingForward>] [no receipt-nbr-rcvd | receipt-nbr-rcvd <Sip::CasTimerProfReceipNbrRecv>] [no meter-pulse-dur | meter-pulse-dur <Sip::CasTimerProfMeterPulseDur>])

Command Parameters

Table 40.35-1 "Voice Sip Cas Timer Profile Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(prof-name)	Format: - the name of a timer profile - length: 1<=x<=32	The unique name identifier of an row in the sipCasTimerProfileTable.

Table 40.35-2 "Voice Sip Cas Timer Profile Configuration Commands" Command Parameters

Parameter	Type	Description
[no] version-nbr	Parameter type: <Sip::CasTimerProfVersionNbr> Format: - a version number of the profile - range: [1...65535]	*optional parameter with default value: 1* The version number to the profile.
appl-bitmap	Parameter type: <Sip::CasTimerProfAppliBitmap> Format:	*mandatory parameter* A bitmap that allows to indicate

Parameter	Type	Description
	(cas-2bit-dtmf | cas-1bit-dtmf | cas-1bit-decade | cas-1bit-mfc-fw | cas-1bit-mfc-gr-fw-bw | cas-2bit-mfc-gr-fw-bw) Possible values: - cas-2bit-dtmf : Indicate the timer objects of sipCasTimerProfileSeizeAck, sipCasTimerProfileClearBackAck, sipCasTimerProfileClearForwardAck, sipCasTimerProfileCallProcess, sipCasTimerProfileSeizureStateTime - cas-1bit-dtmf : Indicate the timer objects of sipCasTimerProfileClearBackAck, sipCasTimerProfileClearForwardAck, sipCasTimerProfileCallProcess, sipCasTimerProfileDelayDialingMinDuration, sipcasTimerProfileDelayDialingMaxDuration, sipCasTimerProfileSeizureStateTime - cas-1bit-decade : Indicate the timer objects of sipCasTimerProfileClearBackAck, sipCasTimerProfileClearForwardAck, sipCasTimerProfileCallProcess, sipCasTimerProfileDelayDialingMinDuration, sipCasTimerProfileDelayDialingMaxDuration, sipCasTimerProfileSeizureStateTime, sipCasTimerProfileDecadicAFPulseDuration, sipCasTimerProfileDecadicAFBreakDuration, sipCasTimerProfileDpInterdigitPeriod - cas-1bit-mfc-fw : Indicate the timer objects of sipCasTimerProfileClearBackAck, sipCasTimerProfileClearForwardAck, sipCasTimerProfileCallProcess, sipCasTimerProfileDelayDialingMinDuration, sipcasTimerProfileDelayDialingMaxDuration, sipCasTimerProfileRingForward - cas-1bit-mfc-gr-fw-bw : Indicate the timer objects of sipCasTimerProfileClearBackAck, sipCasTimerProfileClearForwardAck, sipCasTimerProfilePresenceForwardSig, sipCasTimerProfileReceiptForwardSig, sipCasTimerProfileCallProcess, sipCasTimerProfileSeizureStateTime, sipCasTimerProfileRegisterRecallMinDuration, sipCasTimerProfileRegisterRecallMaxDuration, sipCasTimerProfileReceiptNumberReceived - cas-2bit-mfc-gr-fw-bw : Indicate the timer objects of sipCasTimerProfileSeizeAck, sipCasTimerProfileClearBackAck, sipCasTimerProfileClearForwardAck, sipCasTimerProfilePresenceForwardSig, sipCasTimerProfileReceiptForwardSig, sipCasTimerProfileReanswer, sipCasTimerProfileCallProcess, sipCasTimerProfileRegisterRecallMinDuration, sipCasTimerProfileRegisterRecallMaxDuration,	the timer objects that are applicable to the profile.

Parameter	Type	Description
	sipCasTimerProfileMeteringPulseDuration	
[no] seize-ack	Parameter type: <Sip::CasTimerProfSeizeAck> Format: - Timer for the receipt of the SEIZURE ACK Signal. - unit: millisecond - range: [100...1000]	*optional parameter with default value: "200"* Timer for the receipt of the SEIZURE ACK Signal. The unit is millisecond.
[no] clear-back-ack	Parameter type: <Sip::CasTimerProfClearBwAck> Format: - Timer for the receipt of the CLEAR BACK ACK Signal. - unit: second - range: [1...360]	*optional parameter with default value: "90"* Timer for the receipt of the CLEAR BACK ACK Signal. The unit is second.
[no] clear-fw-ack	Parameter type: <Sip::CasTimerProfClearFwAck> Format: - Timer for the receipt of the CLEAR FORWARD ACK Signal. - unit: second - range: [1...360]	*optional parameter with default value: "90"* Timer for the receipt of the CLEAR FORWARD ACK Signal. The unit is second.
[no] presence-fw-sig	Parameter type: <Sip::CasTimerProfPresenceFwSig> Format: - Timer that defines the interval for the presence of the FORWARDING Signal. - unit: second - range: [1...36]	*optional parameter with default value: "15"* Timer that defines the interval for the presence of the FORWARDING Signal. The unit is second.
[no] receipt-fw-sig	Parameter type: <Sip::CasTimerProfReceiptFwSig> Format: - Timer that defines the interval for the receipt of the FORWARDING Signal. - unit: second - range: [1...20]	*optional parameter with default value: "7"* Timer that defines the interval for the receipt of the FORWARDING Signal. The unit is second.
[no] re-answer	Parameter type: <Sip::CasTimerProfReAnswer> Format: - The re-answer timer on terminated calls to the TDM R2 side. - unit: second - range: [0...360]	*optional parameter with default value: "90"* The re-answer timer on terminated calls to the TDM R2 side. The unit is second.
[no] call-process	Parameter type: <Sip::CasTimerProfCallProcess> Format: - Timer for the call process. The timer is started after digit collection completed. - unit: second - range: [0...360]	*optional parameter with default value: "0"* Timer for the call process. The timer is started after digit collection completed. The unit is second.
[no] delay-dial-min	Parameter type: <Sip::CasTimerProfDelayDialingMinDur> Format: - Minimum time of the signal duration to recognize the delay-dialing signal (defined by ITU Q.311). - unit: millisecond - range: [140...500]	*optional parameter with default value: "220"* Minimum time of the signal duration to recognize the delay-dialing signal (defined by ITU Q.311). The unit is millisecond.
[no] delay-dial-max	Parameter type: <Sip::CasTimerProfDelayDialingMaxDur> Format: - Maximum time of the signal duration to recognize the delay-dialing signal (defined by ITU Q.311). - unit: millisecond - range: [140...500]	*optional parameter with default value: "250"* Maximum time of the signal duration to recognize the delay-dialing signal (defined by ITU Q.311). The unit is millisecond.
[no] seize-state-time	Parameter type: <Sip::CasTimerProfSeizeStateTime> Format:	*optional parameter with default value: "600"*

Parameter	Type	Description
	- The timer is used to indicate how long after the seizure signal. The value is set as 0 means that SVG will not start this timer and value 0 only apply to cas-2bit-dtmf signalling type termination. The actual valid range is from 200 ms to 1500 ms. - unit: millisecond - range: [0,200...1500]	The timer is used to indicate how long after the seizure signal, the originating side will start sending the CdPN, for those cases where the line signaling has no indication from the calling side that seizure is accepted. The value is set as 0 means that SVG will not start this timer and value 0 only apply to cas-2bit-dtmf signalling type termination. The actual valid range is from 200 ms to 1500 ms. The unit is millisecond.
[no] reg-recall-min	Parameter type: <Sip::CasTimerProfRegRecallMinDur> Format: - Minimum time of the signal duration to recognize the Register-recall signal. - unit: millisecond - range: [30...2000]	*optional parameter with default value: "40"* Minimum time of the signal duration to recognize the Register-recall signal. The unit is millisecond.
[no] reg-recall-max	Parameter type: <Sip::CasTimerProfRegRecallMaxDur> Format: - Maximum time of the signal duration to recognize the Register-recall signal. - unit: millisecond - range: [30...2000]	*optional parameter with default value: "890"* Maximum time of the signal duration to recognize the Register-recall signal. The unit is millisecond.
[no] decadic-pulse-dur	Parameter type: <Sip::CasTimerProfDpDur> Format: - Length of time for the generation of the Decadic Pulse signal. - unit: millisecond - range: [20...100]	*optional parameter with default value: "60"* Length of time for the generation of the Decadic Pulse signal. The unit is millisecond.
[no] decadic-break-dur	Parameter type: <Sip::CasTimerProfDpBreakDur> Format: - Length of time for the generation of the Decadic Pulse signal. - unit: millisecond - range: [20...100]	*optional parameter with default value: "40"* Length of time for the generation of the Decadic Pulse break. The unit is millisecond.
[no] inter-digit-peri	Parameter type: <Sip::CasTimerProfInterDigitPeriod> Format: - Timer for the inter-digit pauses. - unit: millisecond - range: [200...2000]	*optional parameter with default value: "600"* Timer for the inter-digit pauses. The unit is millisecond.
[no] ring-fw	Parameter type: <Sip::CasTimerProfRingForward> Format: - Length of time for the generation of the ring-forward signal (Q.311). - unit: second - range: [65...135]	*optional parameter with default value: "100"* Length of time for the generation of the ring-forward signal (Q.311). The unit is second.
[no] receipt-nbr-rcvd	Parameter type: <Sip::CasTimerProfReceipNbrRecv> Format: - Timer for the receipt of the Number-received signal. - unit: second - range: [15...100]	*optional parameter with default value: "30"* Timer for the receipt of the Number-received signal. The unit is second.
[no] meter-pulse-dur	Parameter type: <Sip::CasTimerProfMeterPulseDur> Format: - Length of time for the generation of the metering pulse.	*optional parameter with default value: "150"* Length of time for the generation

Parameter	Type	Description
	- unit: millisecond - range: [50...500]	of the metering pulse. The unit is millisecond.

40.36 Network Element Table Configuration Commands

Command Description

This command allows the operator to configure Network Element Table.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no network-element cap-id <Sip::CapId> ne-id <Sip::NeId>) | (network-element cap-id <Sip::CapId> ne-id <Sip::NeId> ne-name <Sip::NeName> [no role | role <Sip::NeRole>] [no admin-status | admin-status <Sip::NeAdminStatus>] [no ip | ip <SIP::IpAddressAndMask>] [no vlan-id | vlan-id <Sip::NeVlanId>])

Command Parameters

Table 40.36-1 "Network Element Table Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
cap-id	Parameter type: <Sip::CapId> Format: - the central access point id - range: [1...16]	The unique identifier of this central access point
ne-id	Parameter type: <Sip::NeId> Format: - the network element id - range: [1...8]	The unique identifier of this network element

Table 40.36-2 "Network Element Table Configuration Commands" Command Parameters

Parameter	Type	Description
ne-name	Parameter type: <Sip::NeName> Format: - the name of the network element - length: x<=32	*mandatory parameter* The operator friendly name of this network element
[no] role	Parameter type: <Sip::NeRole> Format: (local-ne | remote-ne | cap-ne) Possible values: - local-ne : the entry of a REMOTE NE is configured in that same REMOTE NE	*optional parameter with default value: "cap-ne"* The CAP related role of the network element

Parameter	Type	Description
	- remote-ne : the entry of a REMOTE NE is configured in the CAP NE or another REMOTE NE - cap-ne : the entry of the CAP NE is configured in the CAP NE or a remote NE	
[no] admin-status	Parameter type: <Sip::NeAdminStatus> Format: (up | down) Possible values: - up : the ISAMv NE can be addressed - down : the ISAMv NE cannot be addressed	*optional parameter with default value: "down"* administrative status of this network element
[no] ip	Parameter type: <SIP::IpAddressAndMask> Format: <Ip::V4Address> / <Sip::PrefixLength> Field type <Ip::V4Address> - IPv4-address Field type <Sip::PrefixLength> - prefix length of the subnet - range: [0...128]	*optional parameter with default value: "0.0.0.0/0"* The internal communication IP address that applies to the NIAT-A LT boards
[no] vlan-id	Parameter type: <Sip::NeVlanId> Format: - The VLAN ID used by the internal communication between the CAP and the NIAT-A LT board - range: [0...4093]	*optional parameter with default value: "0"* The VLAN ID used by the internal communication between the CAP and the NIAT-A LT board

40.37 Voice Sip Call Query Configuration Commands

Command Description

This command allows the operator to create and start a call query for active call.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice sip (no call-query (id)) | (call-query (id) [no timeout-period | timeout-period <Sip::TimeoutPeriod>] target-dn <Sip::CallQueryTargetDN> [no remote-dn | remote-dn <Sip::CallQueryRemoteDN>] [no type | type <Sip::CallQueryType>])

Command Parameters

Table 40.37-1 "Voice Sip Call Query Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(id)	Format: - the index of call query session - range: [1]	The unique identifier of this call query session

Table 40.37-2 "Voice Sip Call Query Configuration Commands" Command Parameters

Parameter	Type	Description
[no] timeout-period	Parameter type: <Sip::TimeoutPeriod> Format: - Allows to set the whole timer of sending call query request to all configured LTs and receiving query results of these LTs. - unit: second - range: [1...255]	*optional parameter with default value: 10* Allows to set the whole timer of sending call query request to all configured LTs and receiving query results of these LTs.
target-dn	Parameter type: <Sip::CallQueryTargetDN> Format: - The Target DN is a number that is registered on that specific SVG shelf. It can be a GDN number or Puid number (for ISDN and CAS), and it can be either the calling or called number. If uri is configured, the Target DN should be consistent with user part in configured uri. If uri is not configured, the Target DN should be consistent with directory number. - length: 1<=x<=32	*mandatory parameter* The Target DN is a number that is registered on that specific SVG shelf. It can be a GDN number or Puid number (for ISDN and CAS), and it can be either the calling or called number. If uri is configured, the Target DN should be consistent with user part in configured uri. If uri is not configured, the Target DN should

Parameter	Type	Description
		be consistent with directory number.
[no] remote-dn	Parameter type: <Sip::CallQueryRemoteDN> Format: - The Remote DN can be any number, this is the number with which the Target DN is expected to be in a call. - length: x<=32	*optional parameter with default value: ""* The Remote DN can be any number, this is the number with which the Target DN is expected to be in a call.
[no] type	Parameter type: <Sip::CallQueryType> Format: (pots | isdn-pri | cas) Possible values: - pots : only transfer request data to NPOT-B/NPOT-C and receive POTS call query results from them. - isdn-pri : only transfer request data to NIAT-A and receive ISDN-PRI call query results from them. - cas : only transfer request data to NIAT-A and receive CAS call query results from them.	*optional parameter with default value: "pots"* Allows to configure which type of call will be queried on Target DN.

40.38 Voice Cluster Configuration Commands

Command Description

This command allows the operator to manage the Voice Cluster.

User Level

The command can be accessed by operators with megaco privileges, and executed by operators with megaco privileges.

Command Syntax

The command has the following syntax:

> configure voice cluster (cluster-id)

Command Parameters

Table 40.38-1 "Voice Cluster Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(cluster-id)	Format: - the xvps cluster id - range: [1...8,11]	uniquely identify of this xvps cluster

40.39 Voice Megaco IP Configuration Commands

Command Description

This command allows the operator to manage the Voice Megaco xvps ip.

User Level

The command can be accessed by operators with megaco privileges, and executed by operators with megaco privileges.

Command Syntax

The command has the following syntax:

> configure voice cluster (cluster-id) ip [no ivps-ip | ivps-ip <Ip::V4Address>] [no netmask | netmask <Ip::V4Address>] [no router-ip | router-ip <Ip::V4Address>] [vlan-id <MEGACO::ivpsXLESVLAN>] [no ip-mode | ip-mode <MEGACO::voiceIPConfigMode>] [no dhcpoption60 | dhcpoption60 <MEGACO::voiceIPDhcpv4Option60>] [no private-ip | private-ip <Ip::V4Address>] [no private-netmask | private-netmask <Ip::V4Address>] [no private-vlan-id | private-vlan-id <MEGACO::ivpsPrivateVLAN>]

Command Parameters

Table 40.39-1 "Voice Megaco IP Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(cluster-id)	Format: - the xvps cluster id - range: [1...8,11]	uniquely identify of this xvps cluster

Table 40.39-2 "Voice Megaco IP Configuration Commands" Command Parameters

Parameter	Type	Description
[no] ivps-ip	Parameter type: <Ip::V4Address> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* xles ip address of this xvps cluster.When operator configure Ip parameters first time ,operator should provide ivps-ip,netmask and vlan-id at the same time.
[no] netmask	Parameter type: <Ip::V4Address> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* xles ip subnet mask address of this xvps cluster.When operator configure Ip parameters first time ,operator should provide ivps-ip,netmask and vlan-id at the same time.
[no] router-ip	Parameter type: <Ip::V4Address> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* the next hop ip address of this

Parameter	Type	Description
		xvps cluster.When operator configure Ip parameters first time ,operator should provide ivps-ip,netmask and vlan-id at the same time.
vlan-id	Parameter type: <MEGACO::ivpsXLESVLAN> Format: - the vlan id of xvps - range: [0...4093]	*optional parameter* vlan id of voice xvps cluster
[no] ip-mode	Parameter type: <MEGACO::voiceIPConfigMode> Format: (dhcp | manual) Possible values: - dhcp : The ip mode is dhcp - manual : The ip mode is manual	*optional parameter with default value: "manual"* the configure mode of the IP addresses
[no] dhcpoption60	Parameter type: <MEGACO::voiceIPDhcpv4Option60> Format: - DHCP option 60 for the DHCP client - length: x<=64	*optional parameter with default value: ""* the DHCP option 60 for the DHCP client
[no] private-ip	Parameter type: <Ip::V4Address> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* the private ip address of voice xvps cluster.When operator configure Ip parameters first time ,operator should provide ivps-ip,netmask and vlan-id at the same time.
[no] private-netmask	Parameter type: <Ip::V4Address> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* the private ip subnet mask address of voice xvps cluster.When operator configure Ip parameters first time ,operator should provide ivps-ip,netmask and vlan-id at the same time.
[no] private-vlan-id	Parameter type: <MEGACO::ivpsPrivateVLAN> Format: - the private vlan id of xvps - range: [0...4093]	*optional parameter with default value: 0* the private vlan of voice xvps cluster.When operator configure Ip parameters first time ,operator should provide ivps-ip,netmask and vlan-id at the same time.

40.40 Voice Megaco Equipment Configuration Commands

Command Description

This command allows the operator to manage the Voice equipment.One ivps can manage at most 256 equipments,so the id of equipment is from 1 to 256.

If you want to configure a board or a termination ,you must configure a equipment first.

User Level

The command can be accessed by operators with megaco privileges, and executed by operators with megaco privileges.

Command Syntax

The command has the following syntax:

> configure voice cluster (cluster-id) (no equipment (equip-id)) | (equipment (equip-id) asam-id <MEGACO::accessEquipmentAsamId> ip-address <Ip::V4Address> [no next-hop | next-hop <Ip::V4Address>])

Command Parameters

Table 40.40-1 "Voice Megaco Equipment Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(cluster-id)	Format: - the xvps cluster id - range: [1...8,11]	uniquely identify of this xvps cluster
(equip-id)	Format: - id of equipment - range: [1...32]	uniquely identify of this equipment

Table 40.40-2 "Voice Megaco Equipment Configuration Commands" Command Parameters

Parameter	Type	Description
asam-id	Parameter type: <MEGACO::accessEquipmentAsamId> Format: - the equipment asam id of ne - range: [a-zA-Z0-9-_./] - length: x<=64	*mandatory parameter* asam identify of this ne
ip-address	Parameter type: <Ip::V4Address> Format: - IPv4-address	*mandatory parameter* *The parameter is not visible during modification.* the voice ip address of this isam-v ne.
[no] next-hop	Parameter type: <Ip::V4Address> Format:	*optional parameter with default value: "0.0.0.0"*

Parameter	Type	Description
	- IPv4-address	the ip address of the next hop for this isam-v ne

40.41 Voice Megaco Equipment Board Configuration Commands

Command Description

This command allows the operator to manage the Voice board.Before you confiure a board,you must configure a equipment first.

User Level

The command can be accessed by operators with megaco privileges, and executed by operators with megaco privileges.

Command Syntax

The command has the following syntax:

> configure voice cluster (cluster-id) equipment (equip-id) (no board (board-id)) | (board (board-id) planned-type <Equipm::BoardFuncType> lanx-port <MEGACO::accessBoardLanxPort>)

Command Parameters

Table 40.41-1 "Voice Megaco Equipment Board Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(cluster-id)	Format: - the xvps cluster id - range: [1...8,11]	uniquely identify of this xvps cluster
(equip-id)	Format: - id of equipment - range: [1...32]	uniquely identify of this equipment
(board-id)	Format: (<Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::NewSlotId> | <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::NewSlotId>) Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::NewSlotId> - the LT slot number	uniquely identify of this board. the CLI slot numbering method is based on logical concept.for XD shelf, CLI logic slot-id(1 to 16) is mapped to physical slot-id from 4 to 19. for FD shelf, CLI logic slot-id(1 to 16) is mapped to physical slot-id from 1 to 8 and 12 to 19. In extend-lt mode, the slot-id range shall be 1 to 18.For XD shelf, CLI logic slot-id(17 to 18) is mapped to physical slot-id from 2 to 3. for FD shelf, CLI logic slot-id(17 to 18) is mapped to physical slot-id from 10 to 11

Table 40.41-2 "Voice Megaco Equipment Board Configuration Commands" Command Parameters

Parameter	Type	Description
planned-type	Parameter type: <Equipm::BoardFuncType> Format: (nbat-b | nbat-a | npot-a | nvps-a | nvps-c | polt-b | npot-b | npot-c | ivps-b | balt-a | ivps-a | polt-a | isdn-24l | pots-48l) Possible values: - nbat-b : ISDN Basic Access line termination board for FD (4B3T) - nbat-a : ISDN Basic Access line termination board for FD - npot-a : 48 Pots Only LT board for FD - nvps-a : Isam Voice Packet Server for FD - nvps-c : Isam Voice Packet Server for FD - polt-b : 48 Pots Only LT board for XD - npot-b : 72 Pots Only LT board for FD - npot-c : 48 Pots Only LT board for FD - ivps-b : Isam Voice Packet Server for XD - balt-a : 48 ISDN Only LT board for XD - ivps-a : Isam Voice Packet Server for XD - polt-a : 48 Pots Only LT board for XD - isdn-24l : 24 ISDN LT board for XD/FD, only for migration, can not be configured - pots-48l : 48 Pots LT board for XD/FD, only for migration, can not be configured	*mandatory parameter* the type of user board
lanx-port	Parameter type: <MEGACO::accessBoardLanxPort> Format: - the logic slot of board - range: [1...32]	*mandatory parameter* *The parameter is not visible during modification.* the logic slot of this board

40.42 Voice Megaco Equipment Termination Configuration Commands

Command Description

This command allows the operator to manage the Voice termination.Before you configure a termination ,you must configure a equipment first.

User Level

The command can be accessed by operators with megaco privileges, and executed by operators with megaco privileges.

Command Syntax

The command has the following syntax:

> configure voice cluster (cluster-id) equipment (equip-id) (no termination (port-id)) | (termination (port-id) [no type | type <MEGACO::accTerminationUserPortType>] [no isdn-codec | isdn-codec <MEGACO::IsdnCodecType>] [no switch-type | switch-type <MEGACO::accTerminationPacketSwitchType>] [no activate-type | activate-type <MEGACO::accTerminationL1ActivateType>] [termination-id <MEGACO::accessTerminationId>] media-gateway-id <MEGACO::MediaGatewayId> [no admin-status | admin-status <MEGACO::accTerminationAdminStatus>] [no line-feed | line-feed <MEGACO::accTerminationLineCharact>] [no rx-gain | rx-gain <MEGACO::accTerminationRxGain>] [no tx-gain | tx-gain <MEGACO::accTerminationTxGain>] [no impedance | impedance <MEGACO::accTerminationImpedance>] [no rtp-dscp | rtp-dscp <MEGACO::accTerminationVoiceDscp>] [no rtp-pbits | rtp-pbits <MEGACO::accTerminationVoiceDot1P>] [no clip-mode | clip-mode <MEGACO::accTerminationETSIClipDataMode>] [no metering-type | metering-type <MEGACO::accTerminationMeteringPulseType>] [no directory-number | directory-number <MEGACO::accTerminationDirectoryNumber>] [no voice-service | voice-service <MEGACO::accTermVoiceServAtMgc>] [no reserved | reserved <MEGACO::accTerminationReserved>])

Command Parameters

Table 40.42-1 "Voice Megaco Equipment Termination Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(cluster-id)	Format: - the xvps cluster id - range: [1...8,11]	uniquely identify of this xvps cluster
(equip-id)	Format: - id of equipment - range: [1...32]	uniquely identify of this equipment
(port-id)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::NewSlotId> / <Eqpt::MegacoPortId> Field type <Eqpt::RackId>	unique identifier of this termination port

Resource Identifier	Type	Description
	- the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::NewSlotId> - the LT slot number Field type <Eqpt::MegacoPortId> - the port number of Megaco	

Table 40.42-2 "Voice Megaco Equipment Termination Configuration Commands" Command Parameters

Parameter	Type	Description
[no] type	Parameter type: <MEGACO::accTerminationUserPortType> Format: (pstn | isdn) Possible values: - pstn : the termination type is pstn - isdn : the termination type is isdn	*optional parameter with default value: "pstn"* The type of termination user port
[no] isdn-codec	Parameter type: <MEGACO::IsdnCodecType> Format: (alaw | ulaw) Possible values: - alaw : the codec type on TDM side of isdn is a_law - ulaw : the codec type on TDM side of isdn is u_law	*optional parameter with default value: "alaw"* the type of isdn codec on TDM side
[no] switch-type	Parameter type: <MEGACO::accTerminationPacketSwitchType> Format: (enable | disable) Possible values: - enable : the packet switch type is enable - disable : the packet switch type is disable	*optional parameter with default value: "disable"* The type of the packet switch type, only for type isdn
[no] activate-type	Parameter type: <MEGACO::accTerminationL1ActivateType> Format: (permanent | percall) Possible values: - permanent : the layer1 activate type is permanent - percall : the layer1 activate type is percall	*optional parameter with default value: "permanent"* The type of the layer1 activate type, only for type isdn
termination-id	Parameter type: <MEGACO::accessTerminationId> Format: - the termination id, if it is not specified, system will automatically assign a free one in range - range: [0...32767]	*optional parameter* The id of termination, if not entered during creating, it will be assigned by system
media-gateway-id	Parameter type: <MEGACO::MediaGatewayId> Format: - the media gateway table index - range: [1]	*mandatory parameter* The id of media gateway
[no] admin-status	Parameter type: <MEGACO::accTerminationAdminStatus> Format: (locked | unlocked) Possible values: - locked : the admin status of termination is locked	*optional parameter with default value: "locked"* The administrative status of termination

Parameter	Type	Description
	- unlocked : the admin status of termination is unlocked	
[no] line-feed	Parameter type: <MEGACO::accTerminationLineCharact> Format: (25 | 40) Possible values: - 25 : the line character is 25 mA - 40 : the line character is 40 mA	*optional parameter with default value: "25"* The line character of this termination, only for type pstn
[no] rx-gain	Parameter type: <MEGACO::accTerminationRxGain> Format: - the termination rx-gain - range: [-14...6]	*optional parameter with default value: "0"* The rx-gain of this termination, only for type pstn
[no] tx-gain	Parameter type: <MEGACO::accTerminationTxGain> Format: - the termination tx-gain - range: [-14...6]	*optional parameter with default value: "0"* The tx-gain of this termination, only for type pstn
[no] impedance	Parameter type: <MEGACO::accTerminationImpedance> Format: (default | 200 | 220 | 220minisplitter | 270 | 300 | 370 | 600 | 900 | 370minisplitter | 600splitter | 370nvlsasplitter | 370skinnysplitter | 270duratelsplitter | 2703msplitter | 100 | 150 | 220splitter | reserv19 | reserv20 | reserv21) Possible values: - default : the impedance is default vaule - 200 : the line impedance is 200 ohm - 220 : the line impedance is 220 ohm - 220minisplitter : the line impedance is 220 ohm+ miniSplitter - 270 : the line impedance is 270 ohm - 300 : the line impedance is 300 ohm - 370 : the line impedance is 370 ohm - 600 : the line impedance is 600 ohm - 900 : the line impedance is 900 ohm - 370minisplitter : the line impedance is 370 ohm+ miniSplitter - 600splitter : the line impedance is 600 ohm+ Splitter - 370nvlsasplitter : the line impedance is 370 ohm+ NVLSASplitter - 370skinnysplitter : the line impedance is 370 ohm+ SkinnySplitter	*optional parameter with default value: "default"* The impedance of this termination, only for type pstn

Parameter	Type	Description
	- 270duratelsplitter : the line impedance is 270 ohm+ DuratelSplitter - 2703msplitter : the line impedance is 270 ohm+ 3MDplitter - 100 : the line impedance is 100 ohm - 150 : the line impedance is 150 ohm - 220splitter : the line impedance is 220 ohm+ Splitter - reserv19 : reserved for future use 19 - reserv20 : reserved for future use 20 - reserv21 : reserved for future use 21	
[no] rtp-dscp	Parameter type: <MEGACO::accTerminationVoiceDscp> Format: - the termination voice dscp - range: [-1...63]	*optional parameter with default value: "-1"* the voice dscp of termination
[no] rtp-pbits	Parameter type: <MEGACO::accTerminationVoiceDot1P> Format: - the termination voice p-bit - range: [-1...7]	*optional parameter with default value: "-1"* the voice p-bit of termination
[no] clip-mode	Parameter type: <MEGACO::accTerminationETSIClipDataMode> Format: (cdevalue | fsk | dtmf) Possible values: - cdevalue : cde configuration will be applied for etsi clip - fsk : fsk will be applied for etsi clip - dtmf : dtmf will be applied for etsi clip	*optional parameter with default value: "cdevalue"* the etsi clip data transmission protocol of this access termination,it can only be configured in pstn line cards
[no] metering-type	Parameter type: <MEGACO::accTerminationMeteringPulseType> Format: (pulse | polarityreverse) Possible values: - pulse : 12/16 KHz sine waveform pulse will be applied for metering pulse - polarityreverse : line polarity reverse pulse will be applied for metering pulse	*optional parameter with default value: "pulse"* the metering pulse type of this access termination,it can only be configured on pstn line
[no] directory-number	Parameter type: <MEGACO::accTerminationDirectoryNumber> Format: - the termination directory number.'#' is invalid character - length: 1<=x<=16	*optional parameter with default value: ""* the directory number of this access termination,it can be configured on pstn and ISDN BA line
[no] voice-service	Parameter type: <MEGACO::accTermVoiceServAtMgc> Format: (enabled | disabled) Possible values: - enabled : The voice service of termination is enabled - disabled : The voice service of termination is disabled	*optional parameter with default value: "enabled"* voice service provision of termination
[no] reserved	Parameter type: <MEGACO::accTerminationReserved> Format: - reserved for future usage - length: x<=32	*optional parameter with default value: ""* reserved for future usage

40.43 Voice Megaco Termination Configuration Commands

Command Description

This command allows the operator to enable/disable tca or configure the threshold for rtp packetloss, jitter and delay of the Voice termination.

User Level

The command can be accessed by operators with megaco privileges, and executed by operators with megaco privileges.

Command Syntax

The command has the following syntax:

> configure voice cluster (cluster-id) equipment (equip-id) termination (port-id) tca [no tca-enable | tca-enable <MEGACO::accTerminationTCAEnable>] [no rtp-pktloss-thres | rtp-pktloss-thres <MEGACO::accTerminationRtpPacketLossTCAThreshold>] [no rtp-jitter-thres | rtp-jitter-thres <MEGACO::accTerminationRtpJitterTCAThreshold>] [no rtp-delay-thres | rtp-delay-thres <MEGACO::accTerminationRtpDelayTCAThreshold>]

Command Parameters

Table 40.43-1 "Voice Megaco Termination Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(cluster-id)	Format: - the xvps cluster id - range: [1...8,11]	uniquely identify of this xvps cluster
(equip-id)	Format: - id of equipment - range: [1...32]	uniquely identify of this equipment
(port-id)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::NewSlotId> / <Eqpt::MegacoPortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::NewSlotId> - the LT slot number Field type <Eqpt::MegacoPortId> - the port number of Megaco	unique identifier of this termination port

Table 40.43-2 "Voice Megaco Termination Configuration Commands" Command Parameters

Parameter	Type	Description
[no] tca-enable	Parameter type: <MEGACO::accTerminationTCAEnable> Format: (enabled | disabled) Possible values: - enabled : enable tca alarm report - disabled : disable tca alarm report	*optional parameter with default value: "disabled"* Allows to enable/disable the Threshold Crossing Alarm feature for this termination
[no] rtp-pktloss-thres	Parameter type: <MEGACO::accTerminationRtpPacketLossTCAThreshold> Format: - the packet loss threshold for the termination - range: [0...100]	*optional parameter with default value: 1* the rtp packet loss threshold of this megaco termination,a value 0 means TCA_packetloss will be disabled
[no] rtp-jitter-thres	Parameter type: <MEGACO::accTerminationRtpJitterTCAThreshold> Format: - the interarrival jitter threshold for the termination - unit: millisec - range: [0...2147483647]	*optional parameter with default value: 60* the rtp interarrival jitter threshold of this megaco termination,a value 0 means TCA_jitter will be disabled
[no] rtp-delay-thres	Parameter type: <MEGACO::accTerminationRtpDelayTCAThreshold> Format: - the round trip delay threshold for the termination - unit: millisec - range: [0...2147483647]	*optional parameter with default value: 400* the rtp round trip delay threshold of this megaco termination,a value 0 means TCA_delay will be disabled

40.44 Voice Megaco Termination Lawful Intercept Configuration Commands

Command Description

This command allows the operator to enable/disable lawful intercept and configure the src-port,dest-ip,dest-port for li stream. the src ip is same as rtp ip.

User Level

The command can be accessed by operators with li_voice privileges, and executed by operators with li_voice privileges.

Command Syntax

The command has the following syntax:

> configure voice cluster (cluster-id) equipment (equip-id) termination (port-id) (no li) | (li [no state | state <MEGACO::accTerminationLI>] src-port <MEGACO::accTerminationLISrcUdpPort> dest-ip <Ip::V4Address> dest-port <MEGACO::accTerminationLIDestUdpPort>)

Command Parameters

Table 40.44-1 "Voice Megaco Termination Lawful Intercept Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(cluster-id)	Format: - the xvps cluster id - range: [1...8,11]	uniquely identify of this xvps cluster
(equip-id)	Format: - id of equipment - range: [1...32]	uniquely identify of this equipment
(port-id)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::NewSlotId> / <Eqpt::MegacoPortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::NewSlotId> - the LT slot number Field type <Eqpt::MegacoPortId> - the port number of Megaco	unique identifier of this termination port

Table 40.44-2 "Voice Megaco Termination Lawful Intercept Configuration Commands" Command Parameters

Parameter	Type	Description
[no] state	Parameter type: <MEGACO::accTerminationLI> Format: (enabled | disabled) Possible values: - enabled : enable lawful intercept - disabled : disable lawful intercept	*optional parameter with default value: "disabled"* allows to enable/disable the lawful intercept feature for this termination
src-port	Parameter type: <MEGACO::accTerminationLISrcUdpPort> Format: - the source udp port for lawful intercept - range: [49153...65535]	*mandatory parameter* the source udp port for lawful intercept. source ip address is same as voice ip
dest-ip	Parameter type: <Ip::V4Address> Format: - IPv4-address	*mandatory parameter* the destination ip address for lawful intercept
dest-port	Parameter type: <MEGACO::accTerminationLIDestUdpPort> Format: - the destination udp port for lawful intercept - range: [0...65535]	*mandatory parameter* the destination udp port for lawful intercept

40.45 Voice Megaco Equipment Secondary Termination Configuration Commands

Command Description

This command allows the operator to manage the Voice secondary termination.Before you configure the secondary termination ,you must configure an equipment and prime termination first.

User Level

The command can be accessed by operators with megaco privileges, and executed by operators with megaco privileges.

Command Syntax

The command has the following syntax:

> configure voice cluster (cluster-id) equipment (equip-id) (no sharedlineterm (port-id)) | (sharedlineterm (port-id) [sharedlineterm-id <MEGACO::accessTerminationId>] [no admin-status | admin-status <MEGACO::accTerminationAdminStatus>] [no directory-number | directory-number <MEGACO::accTerminationDirectoryNumber>] [no voice-service | voice-service <MEGACO::accTermVoiceServAtMgc>])

Command Parameters

Table 40.45-1 "Voice Megaco Equipment Secondary Termination Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(cluster-id)	Format: - the xvps cluster id - range: [1...8,11]	uniquely identify of this xvps cluster
(equip-id)	Format: - id of equipment - range: [1...32]	uniquely identify of this equipment
(port-id)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::NewSlotId> / <Eqpt::MegacoPortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::NewSlotId> - the LT slot number Field type <Eqpt::MegacoPortId> - the port number of Megaco	unique identifier of the secondary termination port

Table 40.45-2 "Voice Megaco Equipment Secondary Termination Configuration Commands"

Command Parameters

Parameter	Type	Description
sharedlineterm-id	Parameter type: <MEGACO::accessTerminationId> Format: - the termination id, if it is not specified, system will automatically assign a free one in range - range: [0...32767]	*optional parameter* The id of the secondary termination, if not set manually during creating, it will be assigned by system
[no] admin-status	Parameter type: <MEGACO::accTerminationAdminStatus> Format: (locked | unlocked) Possible values: - locked : the admin status of termination is locked - unlocked : the admin status of termination is unlocked	*optional parameter with default value: "locked"* The administrative status of the secondary termination
[no] directory-number	Parameter type: <MEGACO::accTerminationDirectoryNumber> Format: - the termination directory number.'#' is invalid character - length: 1<=x<=16	*optional parameter with default value: ""* the directory number of this access secondary termination
[no] voice-service	Parameter type: <MEGACO::accTermVoiceServAtMgc> Format: (enabled | disabled) Possible values: - enabled : The voice service of termination is enabled - disabled : The voice service of termination is disabled	*optional parameter with default value: "enabled"* voice service provision of the secondary termination

40.46 Voice Megaco Media Gateway Configuration Commands

Command Description

This command allows the operator to manage the Voice Megaco media gateway.
- pstn-term-format: the termination id format pattern of pstn user.
If termid-type is hierarchy, It should be a string constructed according to the following rule:
[prefix][['Dslam_Id'][deli1]][['rack'][formater][deli2]]'shelf'[formater][deli3]'slot'[formater][deli4]'port'[formater]
and thereinto:
'Dslam_Id' is key to indicate whether the Dslam id should be filled into hierarchy termination id. It is optional.
'rack' is key to indicate whether the rack id should be filled into hierarchy termination id. It is optional.
'shelf' is key to indicate the shelf id. It is mandatory.
'slot' is key to indicate the slot id. It is mandatory.
'port' is key to indicate the port id. It is mandatory.
Each key should present only once and the order of the keys should keep as in the rule.
deli can be zero or several characters, be note that the char must be valid for MEGACO termination id.
formater is a format string to indicate how ISAMV should format the id right before it. It must be constructed by zero or at most five digits. The number of digits indicate the minimum width of the id in the termination id, so if the width of the id is less than number of digits in formater, '0' is filling at the beginning. The value that converted by the digits into a integer indicates the value of the first device NO. in the termination id. If the no digit existing, it indicates that no format need to be applied.
prefix is a string constructed by any character that is valid for MEGACO termination id.
At least one of the deli and the formater must be presenting after the key(exclude 'Dslam_Id').
for example:
format string port position termination id
(dslamid/rack/shelf/slot/port)

AL/Dslam_Id/rack/shelf/slot/port nod01/1/1/1/1 AL/nod01/1/1/1/1
AL/Dslam_Id/shelf/slot/port nod01/1/1/1/1 AL/nod01/1/1/1
AL/rack/shelf/slot/port nod01/1/1/1/1 AL/1/1/1/1
AL/rack0/shelf0/slot0/port00 nod01/1/1/1/1 AL/0/0/0/00
ALDslam_Idrack0shelf0slot0port00 nod01/1/1/1/1 ALnod0100000
ALDslam_Idrack0shelf0slot0port0 nod01/1/1/1/12 ALnod0100012

If the termid-type is flat,It should be a string constructed according to the following rule:

[prefix]['<tid'[formater]'>']

and thereinto:

prefix is a string construct by any character that is valid for MEGACO termination id.

formater is a format string to indicate how ISAMV should format the id right before it.

It must be constructed by zero or at most five digits. The number of digits indicat the

minimum width of the id in the termination id,If contains no digit, it indicat that no

format need to be applied. for example:

format string port position termination id

(termination-id)

AL/ 0 AL/0

AL 0 AL0

AL<tid000> 0 AL000

AL<tid000> 888 AL888

- isdn-term-format: the termination id format pattern of isdn user.

If termid-type is hierarchy, It should be a string constructed according to the following rule:

[prefix][['Dslam_Id'][deli1]][['rack'][formater][deli2]]'shelf'[formater][deli3]'slot'[formater]

[deli4]'port'[formater]][deli4]'channel'

and thereinto:

'Dslam_Id' is key to indicate whether the Dslam id should be filled into hierarchy termination id.

It is optional.

'rack' is key to indicate whether the rack id should be filled into hierarchy termination id.

It is optional.

'shelf' is key to indicate the shelf id. It is mandatory.

'slot' is key to indicate the slot id. It is mandatory.

'port' is key to indicate the port id. It is mandatory.

'channel' is key to indicate channel id. It is mandatory.

Each key should present only once and the order of the keys should keep as in the rule.

deli can be zero or several characters, be note that the char must be valid for MEGACO termination id.

formater is a format string to indicate how ISAMV should format the id right before it. It must be

constructed by zero or at most five digits. The number of digits indicate the minimum width of the id

in the termination id, so if the width of the id is less than number of digits in formater, '0' is

filling at the beginning. The value that converted by the digits into a integer indicates the value of

the first device NO. in the termination id. If the no digit existing, it indicates that no format need to

be applied. suffix already includes delimiter, so deli4 is will be replaced by suffix.

prefix is a string constructed by any character that is valid for MEGACO termination id.

At least one of the deli and the formater must be presenting after the key(exclude 'Dslam_Id').

for example:

format string port position termination id

(dslamid/rack/shelf/slot/port suffix)

BA/Dslam_Id/rack/shelf/slot/port/channel nod01/1/1/1/1 /B1 BA/nod01/1/1/1/1/B1

BA/Dslam_Id/shelf/slot/port/channel nod01/1/1/1/1 /B1 BA/nod01/1/1/1/B1

BA/rack/shelf/slot/port/channel nod01/1/1/1/1 /B1 BA/1/1/1/1/B1

BA/rack0/shelf0/slot0/port00/channel/channel nod01/1/1/1/1 /B1 BA/0/0/0/00/B1

BADslam_Idrack0shelf0slot0port00 nod01/1/1/1/1 /B1 BAnod0100000/B1

BADslam_Idrack0shelf0slot0port0/channel nod01/1/1/1/12 B1 BAnod0100012B1

--

If the termid-type is flat,It should be a string constructed according to the following rule:

[prefix]['<tid'[formater]'>']

and thereinto:

prefix is a string construct by any character that is valid for MEGACO termination id.

formater is a format string to indicate how ISAMV should format the id right before it.It must be constructed by zero or at most five digits. The number of digits indicat the minimum width of the id in the termination id,If contains no digit, it indicat that no format need to be applied. ISDN termination id should not be zero because zero is reserved for special use of IID.

for example:

format string port position termination id

(termination-id sufix)

--

BA/ 1 /B1 BA/1/B1

BA 1 /B1 BA1/B1

BA<tid000> 1 /B1 BA001/B1

BA<tid000> 888 /B1 BA888/B1

--

User Level

The command can be accessed by operators with megaco privileges, and executed by operators with megaco privileges.

Command Syntax

The command has the following syntax:

> configure voice cluster (cluster-id) (no media-gateway (media-gateway-id)) | (media-gateway

(media-gateway-id) [no name | name <MEGACO::mediaGatewayName>] [no ip-mode | ip-mode <MEGACO::voiceIPConfigMode>] [no dhcpoption60 | dhcpoption60 <MEGACO::voiceIPDhcpv4Option60>] [no ip-address | ip-address <Ip::V4Address>] [no netmask | netmask <Ip::V4Address>] [no udp-port | udp-port <MEGACO::mediaGatewayUDPPort>] [no router-ip | router-ip <Ip::V4Address>] vlan-id <MEGACO::mediaGatewayVLAN> [no mgc-type | mgc-type <MEGACO::mediaGatewayMgcType>] prim-mgc-ip <Ip::V4Address> [no mgc-id | mgc-id <MEGACO::medGwyCtrlrCallServerId>] [no prim-mgc-udp | prim-mgc-udp <MEGACO::medGwyCtrlrPrimaryUDPPort>] [no sec-mgc-ip | sec-mgc-ip <Ip::V4Address>] [no sec-mgc-udp | sec-mgc-udp <MEGACO::medGwyCtrlrSecondaryUDPPort>] [no tert-mgc-ip | tert-mgc-ip <Ip::V4Address>] [no tert-mgc-udp | tert-mgc-udp <MEGACO::medGwyCtrlrTertiaryUDPPort>] [no quat-mgc-ip | quat-mgc-ip <Ip::V4Address>] [no quat-mgc-udp | quat-mgc-udp <MEGACO::medGwyCtrlrQuaternaryUDPPort>] [no esa-mgc-service | esa-mgc-service <MEGACO::mediaGatewayESAMgcService>] [no mg-mid-type | mg-mid-type <MEGACO::mediaGatewayMidType>] [no mg-domain-name | mg-domain-name <MEGACO::mediaGatewayDomainName>] [no svcreason-format | svcreason-format <MEGACO::mediaGatewaySVCReasonFormat>] [no mg-profile-name | mg-profile-name <MEGACO::mediaGatewayProfileName>] [no admin-status | admin-status <MEGACO::mediaGatewayadminStatus>] [no termid-type | termid-type <MEGACO::mediaGwyTermFormat>] [no pstn-term-format | pstn-term-format <MEGACO::mediaGwyPstnTermFormat>] [isdn-term-format <MEGACO::mediaGwyIsdnTermFormat>] [isdn-suffix1 <MEGACO::mediaGwyIsdnSuffix>] [isdn-suffix2 <MEGACO::mediaGwyIsdnSuffix>] [no max-transhandling | max-transhandling <MEGACO::mediaGwyTMax>] [no max-network-delay | max-network-delay <MEGACO::mediaGwyMaxNetworkDelay>] [no max-retrans | max-retrans <MEGACO::mediaGwyMaxRetrans>] [no red-bat-delay | red-bat-delay <MEGACO::mediaGwyRedBatteryDelay>] [no release-delay | release-delay <MEGACO::mediaGwyReleaseDelay>] [no release-type | release-type <MEGACO::mediaGwyReleaseType>] [no wt-rls-delay | wt-rls-delay <MEGACO::mediaGwyWaitingReleaseDelay>] [no active-heartbeat | active-heartbeat <MEGACO::mediaGwyHeartBeat>] [no passive-heartbeat | passive-heartbeat <MEGACO::mediaGwyHeartBeat>] [no retrans | retrans <MEGACO::mediaGwyRetrans>] [no max-waiting-delay | max-waiting-delay <MEGACO::mediaGwyMaxWaitingDelay>] [no prov-rpl-time | prov-rpl-time <MEGACO::mediaGwyProvResp>] [no signal-dscp | signal-dscp <MEGACO::mediaGatewaySignDscp>] [no signal-pbits | signal-pbits <MEGACO::mediaGatewaySignDot1P>] [no rtp-dscp | rtp-dscp <MEGACO::mediaGatewayVoiceDscp>] [no rtp-pbits | rtp-pbits <MEGACO::mediaGatewayVoiceDot1P>] [no event-req-id | event-req-id <MEGACO::medGwyEventRequestId>] [[no] stml-stdsg-evt] [[no] al-of-evt] [[no] al-on-evt] [[no] al-of-strict-evt] [[no] al-on-strict-evt] [[no] mg-overload-evt] [[no] mg-dummy-evt] [no rfc2833-pl-type | rfc2833-pl-type <MEGACO::mediaGatewayRfc2833PayloadType>] [no rfc2833-process | rfc2833-process <MEGACO::mediaGatewaytRfc2833Process>] [no dial-start-timer | dial-start-timer <MEGACO::mediaGatewayDialStartTimer>] [no dial-long-timer | dial-long-timer <MEGACO::mediaGatewayDialLongTimer>] [no dial-short-timer | dial-short-timer <MEGACO::mediaGatewayDialShortTimer>] [no min-data-jitter | min-data-jitter <MEGACO::mediaGatewayDataJitter>] [no init-data-jitter | init-data-jitter <MEGACO::mediaGatewayDataJitter>] [no max-data-jitter | max-data-jitter <MEGACO::mediaGatewayDataJitter>] [no ephe-term-prefix | ephe-term-prefix <MEGACO::mediaGatewayEpheTermPrefix>] [no ephe-term-min | ephe-term-min <MEGACO::mediaGatewayEpheTerm>] [no ephe-term-max | ephe-term-max <MEGACO::mediaGatewayEpheTerm>])

Command Parameters

Table 40.46-1 "Voice Megaco Media Gateway Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(cluster-id)	Format: - the xvps cluster id - range: [1...8,11]	uniquely identify of this xvps cluster
(media-gateway-id)	Format: - the media gateway table index - range: [1]	uniquely identify of this media gateway

Table 40.46-2 "Voice Megaco Media Gateway Configuration Commands" Command Parameters

Parameter	Type	Description
[no] name	Parameter type: <MEGACO::mediaGatewayName> Format: - uniquely name of this media gateway - length: 1<=x<=64	*optional parameter with default value: "AG"* device name of media gateway, can be modified when mgi is locked, this parameter will be used for mg mid when mg-mid-type select device-name. only '*', '/', '_', '$', '@', 'ALPHA', 'DIGIT', '-', '.' are valid for device-name.
[no] ip-mode	Parameter type: <MEGACO::voiceIPConfigMode> Format: (dhcp | manual) Possible values: - dhcp : The ip mode is dhcp - manual : The ip mode is manual	*optional parameter with default value: "manual"* the ip mode of this media gateway
[no] dhcpoption60	Parameter type: <MEGACO::voiceIPDhcpv4Option60> Format: - DHCP option 60 for the DHCP client - length: x<=64	*optional parameter with default value: ""* the DHCP option 60 for the DHCP client
[no] ip-address	Parameter type: <Ip::V4Address> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* the ip address of this media gateway
[no] netmask	Parameter type: <Ip::V4Address> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* the netmask of this media gateway
[no] udp-port	Parameter type: <MEGACO::mediaGatewayUDPPort> Format: - the udp port of media gateway - range: [1025...65534]	*optional parameter with default value: 2944* the udp port of mgi
[no] router-ip	Parameter type: <Ip::V4Address> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* the next hop ip address of media gateway
vlan-id	Parameter type: <MEGACO::mediaGatewayVLAN> Format: - the vlan id of media gateway - range: [1...4093]	*mandatory parameter* the vlan id of this xvps cluster
[no] mgc-type	Parameter type: <MEGACO::mediaGatewayMgcType> Format: (general | lucent-fs5000 | alcatel-a5020 | alcatel-e10 | zte-zxss | hw-soft3000 | siemens | nortel | ericsson | meta-softswitch | italtel | comverse	*optional parameter with default value: "general"* the type of media gateway controller

Parameter	Type	Description
	| g6-gr303 | other-vendor3 | other-vendor4 | other-vendor16 | other-vendor17) Possible values: - general : The type of mgc is general - lucent-fs5000 : The type of mgc is lucent-fs5000 - alcatel-a5020 : The type of mgc is alcatel-a5020 - alcatel-e10 : The type of mgc is alcatel-e10 - zte-zxss : The type of mgc is zte-zxss - hw-soft3000 : The type of mgc is hw-soft3000 - siemens : The type of mgc is siemens - nortel : The type of mgc is nortel - ericsson : The type of mgc is ericsson - meta-softswitch : The type of mgc is meta-softswitch - italtel : The type of mgc is italtel - comverse : The type of mgc is comverse - g6-gr303 : The type of mgc is g6-gr303 - other-vendor3 : The type of mgc is other-vendor3 - other-vendor4 : The type of mgc is other-vendor4 - other-vendor16 : The type of mgc is other-vendor16 - other-vendor17 : The type of mgc is other-vendor17	
prim-mgc-ip	Parameter type: <Ip::V4Address> Format: - IPv4-address	*mandatory parameter* the ip address of the primary media gateway controller
[no] mgc-id	Parameter type: <MEGACO::medGwyCtrlrCallServerId> Format: - a signed integer	*optional parameter with default value: 0* the identifier of the peer call server
[no] prim-mgc-udp	Parameter type: <MEGACO::medGwyCtrlrPrimaryUDPPort> Format: - the primary udp port of media gateway controller - range: [1...65534]	*optional parameter with default value: 2944* the udp port of primary mgc
[no] sec-mgc-ip	Parameter type: <Ip::V4Address> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* the ip address of the secondary media gateway controller
[no] sec-mgc-udp	Parameter type: <MEGACO::medGwyCtrlrSecondaryUDPPort> Format: - the secondary udp port of media gateway controller - range: [1...65534]	*optional parameter with default value: 2944* the udp port of secondary mgc
[no] tert-mgc-ip	Parameter type: <Ip::V4Address> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* the ip address of the tertiary media gateway controller
[no] tert-mgc-udp	Parameter type: <MEGACO::medGwyCtrlrTertiaryUDPPort> Format: - the tertiary udp port of media gateway controller - range: [1...65534]	*optional parameter with default value: 2944* the udp port of tertiary mgc
[no] quat-mgc-ip	Parameter type: <Ip::V4Address> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* the ip address of the quaternary media gateway controller

Parameter	Type	Description
[no] quat-mgc-udp	Parameter type: <MEGACO::medGwyCtrlrQuaternaryUDPPort> Format: - the quaternary udp port of media gateway controller - range: [1...65534]	*optional parameter with default value: 2944* the udp port of quaternary mgc
[no] esa-mgc-service	Parameter type: <MEGACO::mediaGatewayESAMgcService> Format: (enable | disable) Possible values: - enable : enable local ESA mgc service - disable : disable local ESA mgc service	*optional parameter with default value: "disable"* the state of local ESA mgc service
[no] mg-mid-type	Parameter type: <MEGACO::mediaGatewayMidType> Format: (ipv4 | ipv4-port | domain-name | domain-name-port | device-name) Possible values: - ipv4 : The mg mid type is ipv4 - ipv4-port : The mg mid type is ipv4+port - domain-name : The mg mid type is domain-name - domain-name-port : The mg mid type is domain-name+port - device-name : The mg mid type is device-name	*optional parameter with default value: "ipv4-port"* the mid type of this media gateway: if ipv4 is selected, the MG ip-address will be used; if ipv4-port is selected, the MG ip-address + udp-port will be used; if domain-name is selected, the mg-domain-name will be used; if domain-name-port is selected, the mg-domain-name + udp-port will be used; if device-name is selected, the mg name will be used for mg mid.
[no] mg-domain-name	Parameter type: <MEGACO::mediaGatewayDomainName> Format: - domain name of this media gateway - length: 1<=x<=255	*optional parameter with default value: "isamv.alcatel-lucent.com"* the domain name of media gateway, used when mg-addr-type is domain-name or domain-name-port. only 'ALPHA','DIGIT', '-', '.' ,'@' are valid for domainName; max length is 255.
[no] svcreason-format	Parameter type: <MEGACO::mediaGatewaySVCReasonFormat> Format: (with-quotation | without-quotation) Possible values: - with-quotation : the mg service change reason format is with-quotation - without-quotation : the mg service change reason format is without-quotation	*optional parameter with default value: "with-quotation"* The format of service change reason, which used in H248 service change.
[no] mg-profile-name	Parameter type: <MEGACO::mediaGatewayProfileName> Format: - profile name of this media gateway - length: 1<=x<=64	*optional parameter with default value: "AGW"* profile name of this media gateway. which used in h248 message service change. the max number of the string is 64.
[no] admin-status	Parameter type: <MEGACO::mediaGatewayadminStatus> Format: (locked | unlocked) Possible values:	*optional parameter with default value: "locked"* the administrative status of this media gateway

Parameter	Type	Description
	- locked : The admin status is locked - unlocked : The admin status is unlocked	
[no] termid-type	Parameter type: <MEGACO::mediaGwyTermFormat> Format: (hierarchy | flat) Possible values: - hierarchy : The termination id format is hierarchy - flat : The termination id format is flat	*optional parameter with default value: "flat"* termination id format
[no] pstn-term-format	Parameter type: <MEGACO::mediaGwyPstnTermFormat> Format: - the termination id format pattern of pstn user, the termination id format type has two types:flat and hierarchy.Flat type: example AL; AALN;length: [1...10].Hierarchy type: example AL/Dslam_Id/rack/shelf/slot/port (Dslam_Id/rack optional) length: [1...72]. - length: 1<=x<=72	*optional parameter with default value: "AL"* the termination id format pattern of pstn user
isdn-term-format	Parameter type: <MEGACO::mediaGwyIsdnTermFormat> Format: - the termination id format pattern of isdn user, the termination id format type has two types:flat and hierarchy.Flat type: example BA;length: [1...10]. Hierarchy type: example BA/Dslam_Id/rack/shelf/slot/port/channel(Dslam_Id/rack optional) length:[1...72]. - length: 1<=x<=72	*optional parameter* the termination id format pattern of isdn user
isdn-suffix1	Parameter type: <MEGACO::mediaGwyIsdnSuffix> Format: - the suffix1 of the isdn user termination format - length: 1<=x<=8	*optional parameter* the suffix1 of the isdn user termination format
isdn-suffix2	Parameter type: <MEGACO::mediaGwyIsdnSuffix> Format: - the suffix1 of the isdn user termination format - length: 1<=x<=8	*optional parameter* the suffix2 of the isdn user termination format
[no] max-transhandling	Parameter type: <MEGACO::mediaGwyTMax> Format: - the max time of mgc handling the transaction - range: [1000...30000]	*optional parameter with default value: 16000* the max time of mgc handling the transaction, the unit is millisecond
[no] max-network-delay	Parameter type: <MEGACO::mediaGwyMaxNetworkDelay> Format: - the max network delay time, the unit is millisecond - range: [100...1000]	*optional parameter with default value: 500* the max network delay time, the unit is millisecond
[no] max-retrans	Parameter type: <MEGACO::mediaGwyMaxRetrans> Format: - the max retransimit times of the transaction, the unit is times - range: [7...11]	*optional parameter with default value: 7* the max retransmit times of the transaction
[no] red-bat-delay	Parameter type: <MEGACO::mediaGwyRedBatteryDelay> Format: - the delay before MGI coming into reduced battery state - range: [0...120000]	*optional parameter with default value: 70000* the delay before mgi coming into reduced battery state
[no] release-delay	Parameter type: <MEGACO::mediaGwyReleaseDelay> Format: - the delay before mgi releasing all sessions	*optional parameter with default value: 600000* the delay before mgi releasing all

Parameter	Type	Description
	- unit: millisecond - range: [0...900000]	sessions
[no] release-type	Parameter type: <MEGACO::mediaGwyReleaseType> Format: (normal | never) Possible values: - normal : release type is normal - never : release type is never	*optional parameter with default value: "normal"* the type of releasing the active call
[no] wt-rls-delay	Parameter type: <MEGACO::mediaGwyWaitingReleaseDelay> Format: - the delay after the graceful lock delay timer expired - range: [0...3600000]	*optional parameter with default value: 0* the delay after the graceful lock delay timer expired
[no] active-heartbeat	Parameter type: <MEGACO::mediaGwyHeartBeat> Format: (fix : <MEGACO::mediaGwyHeartBeatInterval> | dynamic) Possible values: - fix : The heart beat mode is fixed - dynamic : The heart beat mode is dynamic Field type <MEGACO::mediaGwyHeartBeatInterval> - the heart beat interval value, if input less than 1s, it will be set to 1s - unit: 10 msec - range: [0...65535]	*optional parameter with default value: "fix : 0"* the active heart beat mode and interval
[no] passive-heartbeat	Parameter type: <MEGACO::mediaGwyHeartBeat> Format: (fix : <MEGACO::mediaGwyHeartBeatInterval> | dynamic) Possible values: - fix : The heart beat mode is fixed - dynamic : The heart beat mode is dynamic Field type <MEGACO::mediaGwyHeartBeatInterval> - the heart beat interval value, if input less than 1s, it will be set to 1s - unit: 10 msec - range: [0...65535]	*optional parameter with default value: "dynamic"* the passive heart beat mode and interval
[no] retrans	Parameter type: <MEGACO::mediaGwyRetrans> Format: (fix : <MEGACO::mediaGwyRetransInterval> | dynamic) Possible values: - fix : The re-transmission mode is fixed - dynamic : The re-transmission mode is dynamic Field type <MEGACO::mediaGwyRetransInterval> - the heart beat interval value - range: [100...4000]	*optional parameter with default value: "fix : 4000"* the retransmission mode and interval
[no] max-waiting-delay	Parameter type: <MEGACO::mediaGwyMaxWaitingDelay> Format: - max time to wait after a restart before contacting to the call server. - range: [0...120]	*optional parameter with default value: 5* max time to wait after a restart before contacting to the call server
[no] prov-rpl-time	Parameter type: <MEGACO::mediaGwyProvResp> Format: - the time need to send a transaction pending - range: [0...10000]	*optional parameter with default value: 1000* the time need to send a transaction pending since the

Parameter	Type	Description
		reception of the transaction
[no] signal-dscp	Parameter type: <MEGACO::mediaGatewaySignDscp> Format: - the signaling dscp of media gateway - range: [0...63]	*optional parameter with default value: "0"* the signaling dscp of this media gateway
[no] signal-pbits	Parameter type: <MEGACO::mediaGatewaySignDot1P> Format: - the signaling p-bit of media gateway - range: [0...7]	*optional parameter with default value: "0"* the signaling p-bit of this media gateway
[no] rtp-dscp	Parameter type: <MEGACO::mediaGatewayVoiceDscp> Format: - the voice dscp of media gateway - range: [0...63]	*optional parameter with default value: "0"* the voice dscp of this media gateway
[no] rtp-pbits	Parameter type: <MEGACO::mediaGatewayVoiceDot1P> Format: - the voice p-bit of media gateway - range: [0...7]	*optional parameter with default value: "0"* the voice p-bit of this media gateway
[no] event-req-id	Parameter type: <MEGACO::medGwyEventRequestId> Format: - the event request id of this media gateway - range: [0...65535]	*optional parameter with default value: 0* the event request id of this media gateway
[no] stml-stdsg-evt	Parameter type: boolean	*optional parameter* permit notify of stimal/stedsig event
[no] al-of-evt	Parameter type: boolean	*optional parameter* permit notify of al/of event
[no] al-on-evt	Parameter type: boolean	*optional parameter* permit notify of al/on event
[no] al-of-strict-evt	Parameter type: boolean	*optional parameter* permit notify of al/of strict=state event
[no] al-on-strict-evt	Parameter type: boolean	*optional parameter* permit notify of al/on strict=state event
[no] mg-overload-evt	Parameter type: boolean	*optional parameter* permit notify of ocp/mg_overload event
[no] mg-dummy-evt	Parameter type: boolean	*optional parameter* dummy parameter for reserved bit
[no] rfc2833-pl-type	Parameter type: <MEGACO::mediaGatewayRfc2833PayloadType> Format: - payload type according to rfc2833 - range: [96...127]	*optional parameter with default value: 96* payload type according to rfc2833
[no] rfc2833-process	Parameter type: <MEGACO::mediaGatewaytRfc2833Process> Format: (audio | rfc2833 | both) Possible values: - audio : processing procedure is audio - rfc2833 : processing procedure is rfc2833 - both : processing procedure is both	*optional parameter with default value: "audio"* processing procedure for the dtmf event tones
[no] dial-start-timer	Parameter type: <MEGACO::mediaGatewayDialStartTimer>	*optional parameter with default value: 10*

Parameter	Type	Description
	Format: - dialing start timer in second - unit: second - range: [0...99]	maximum waiting time for dialing the first digit
[no] dial-long-timer	Parameter type: <MEGACO::mediaGatewayDialLongTimer> Format: - dialing start long in second - unit: second - range: [1...60]	*optional parameter with default value: 20* maximum waiting time for dialing when no matching found in the digitmap
[no] dial-short-timer	Parameter type: <MEGACO::mediaGatewayDialShortTimer> Format: - dialing short timer in second - unit: second - range: [1...60]	*optional parameter with default value: 5* maximum waiting time for dialing when matching found in the digitmap
[no] min-data-jitter	Parameter type: <MEGACO::mediaGatewayDataJitter> Format: - jitter buffer for data calls working in rtp mode - unit: millisecond - range: [0...200]	*optional parameter with default value: 50* minimum jitter buffer for data calls working in rtp mode
[no] init-data-jitter	Parameter type: <MEGACO::mediaGatewayDataJitter> Format: - jitter buffer for data calls working in rtp mode - unit: millisecond - range: [0...200]	*optional parameter with default value: 50* initial jitter buffer for data calls working in rtp mode
[no] max-data-jitter	Parameter type: <MEGACO::mediaGatewayDataJitter> Format: - jitter buffer for data calls working in rtp mode - unit: millisecond - range: [0...200]	*optional parameter with default value: 50* maximum jitter buffer for data calls working in rtp mode
[no] ephe-term-prefix	Parameter type: <MEGACO::mediaGatewayEpheTermPrefix> Format: - prefix to compose the ephemeral termination id - length: 1<=x<=10	*optional parameter with default value: "E"* prefix to compose the ephemeral termination id
[no] ephe-term-min	Parameter type: <MEGACO::mediaGatewayEpheTerm> Format: - ephemeral termination id - range: [0...4294967295]	*optional parameter with default value: 65536* minimum ephemeral termination id
[no] ephe-term-max	Parameter type: <MEGACO::mediaGatewayEpheTerm> Format: - ephemeral termination id - range: [0...4294967295]	*optional parameter with default value: 72366* maximum ephemeral termination id

40.47 Voice Megaco Signal Gateway Configuration Commands

Command Description

This command allows the operator to manage the Voice Megaco signal gateway.

User Level

The command can be accessed by operators with megaco privileges, and executed by operators with megaco privileges.

Command Syntax

The command has the following syntax:

> configure voice cluster (cluster-id) (no signal-gateway (signal-gateway-id)) | (signal-gateway (signal-gateway-id) prim-asp-ip <Ip::V4Address> prim-sctp-port <MEGACO::signallingGatewayInterfacePrimarySCTPPort> [no sec-asp-ip | sec-asp-ip <Ip::V4Address>] [no sec-sctp-port | sec-sctp-port <MEGACO::signallingGatewayInterfaceSecondarySCTPPort>] [no tert-asp-ip | tert-asp-ip <Ip::V4Address>] [no tert-sctp-port | tert-sctp-port <MEGACO::signallingGatewayInterfaceTertiarySCTPPort>] [no quat-asp-ip | quat-asp-ip <Ip::V4Address>] [no quat-sctp-port | quat-sctp-port <MEGACO::signallingGatewayInterfaceQuaternarySCTPPort>] ip-address <Ip::V4Address> sgi-user-label <MEGACO::signallingGatewayInterfaceUserLabel> sgi-mgi <MEGACO::MediaGatewayId> [no admin-status | admin-status <MEGACO::signallingGatewayInterfaceAdminStatus>])

Command Parameters

Table 40.47-1 "Voice Megaco Signal Gateway Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(cluster-id)	Format: - the xvps cluster id - range: [1...8,11]	uniquely identify of this xvps cluster
(signal-gateway-id)	Format: - the signal gateway table index - range: [1]	uniquely identify of this signal gateway

Table 40.47-2 "Voice Megaco Signal Gateway Configuration Commands" Command Parameters

Parameter	Type	Description
prim-asp-ip	Parameter type: <Ip::V4Address> Format: - IPv4-address	*mandatory parameter* the primary asp ip
prim-sctp-port	Parameter type: <MEGACO::signallingGatewayInterfacePrimarySCTPPort> Format: - the sctp port of the primary asp	*mandatory parameter* the sctp port of the primary asp

Parameter	Type	Description
	- range: [1...65534]	
[no] sec-asp-ip	Parameter type: <Ip::V4Address> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* the secondary asp ip
[no] sec-sctp-port	Parameter type: <MEGACO::signallingGatewayInterfaceSecondarySCTPPort> Format: - the sctp port of the secondary asp - range: [1...65534]	*optional parameter with default value: 9900* the sctp port of the secondary asp
[no] tert-asp-ip	Parameter type: <Ip::V4Address> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* the tertiary asp ip
[no] tert-sctp-port	Parameter type: <MEGACO::signallingGatewayInterfaceTertiarySCTPPort> Format: - the sctp port of the tertiary asp - range: [1...65534]	*optional parameter with default value: 9900* the sctp port of the tertiary asp
[no] quat-asp-ip	Parameter type: <Ip::V4Address> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* the quaternary asp ip
[no] quat-sctp-port	Parameter type: <MEGACO::signallingGatewayInterfaceQuaternarySCTPPort> Format: - the sctp port of the quaternary asp - range: [1...65534]	*optional parameter with default value: 9900* the sctp port of the quaternary asp
ip-address	Parameter type: <Ip::V4Address> Format: - IPv4-address	*mandatory parameter* *The parameter is not visible during modification.* the ip address of this signal gateway
sgi-user-label	Parameter type: <MEGACO::signallingGatewayInterfaceUserLabel> Format: - the user label of the signal gateway interface - length: 1<=x<=64	*mandatory parameter* the user label of the signal gateway interface
sgi-mgi	Parameter type: <MEGACO::MediaGatewayId> Format: - the media gateway table index - range: [1]	*mandatory parameter* *The parameter is not visible during modification.* The id of media gateway associated with signal gateway
[no] admin-status	Parameter type: <MEGACO::signallingGatewayInterfaceAdminStatus> Format: (locked | unlocked) Possible values: - locked : the admin status of signal gateway is locked - unlocked : the admin status of signal gateway is unlocked	*optional parameter with default value: "locked"* The administrative status of signal gateway

40.48 Epon Ont Voice SIP UserAgent Configuration

Command Description

This command allows the operator to configure EPON ONT voice sip user agent parameters

User Level

The command can be accessed by operators with sip,megaco privileges, and executed by operators with sip,megaco privileges.

Command Syntax

The command has the following syntax:

> configure voice eont (no sip-useragent (ont-idx)) | (sip-useragent (ont-idx) [no udp-port | udp-port <Epon::UdpPort>] proxy-pri-ip <Epon::PxyPrimaryIp> [no proxy-pri-port | proxy-pri-port <Epon::PxyPrimaryPort>] [proxy-sec-ip <Epon::PxySecondaryIp>] [no proxy-sec-port | proxy-sec-port <Epon::PxySecondaryPort>] reg-pri-ip <Epon::RegPrimaryIp> [no reg-pri-port | reg-pri-port <Epon::RegPrimaryPort>] [reg-sec-ip <Epon::RegSecondaryIp>] [no reg-sec-port | reg-sec-port <Epon::RegSecondaryPort>] [outbound-ip <Epon::OutboundIp>] [no outbound-port | outbound-port <Epon::OutboundPort>] [no reg-intval | reg-intval <Epon::RegInterval>] [no heartbeat-mode | heartbeat-mode <Epon::HeartBeatMode>] [no heartbeat-cycle | heartbeat-cycle <Epon::HeartBeatCycle>] [no heartbeat-count | heartbeat-count <Epon::HeartBeatCount>] digit-map-prof <Epon::DigitMapProfile>)

Command Parameters

Table 40.48-1 "Epon Ont Voice SIP UserAgent Configuration" Resource Parameters

Resource Identifier	Type	Description
(ont-idx)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PonId> / <Eqpt::OntId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::PonId> - the PON identifier Field type <Eqpt::OntId> - the ONT identifier	ONT index

Table 40.48-2 "Epon Ont Voice SIP UserAgent Configuration" Command Parameters

Parameter	Type	Description
[no] udp-port	Parameter type: <Epon::UdpPort> Format: - sip udp port number - range: [0...65535]	*optional parameter with default value: "5060"* signal udp port of sip user agent

Parameter	Type	Description
proxy-pri-ip	Parameter type: <Epon::PxyPrimaryIp> Format: - IPv4-address	*mandatory parameter* sip proxy primary ip address
[no] proxy-pri-port	Parameter type: <Epon::PxyPrimaryPort> Format: - sip udp port number - range: [0...65535]	*optional parameter with default value: "5060"* sip proxy primary udp port
proxy-sec-ip	Parameter type: <Epon::PxySecondaryIp> Format: - IPv4-address	*optional parameter* sip proxy secondary ip address
[no] proxy-sec-port	Parameter type: <Epon::PxySecondaryPort> Format: - sip udp port number - range: [0...65535]	*optional parameter with default value: "5060"* sip proxy secondary udp port
reg-pri-ip	Parameter type: <Epon::RegPrimaryIp> Format: - IPv4-address	*mandatory parameter* sip register server primary ip address
[no] reg-pri-port	Parameter type: <Epon::RegPrimaryPort> Format: - sip udp port number - range: [0...65535]	*optional parameter with default value: "5060"* sip register primary udp port
reg-sec-ip	Parameter type: <Epon::RegSecondaryIp> Format: - IPv4-address	*optional parameter* sip register server secondary ip address
[no] reg-sec-port	Parameter type: <Epon::RegSecondaryPort> Format: - sip udp port number - range: [0...65535]	*optional parameter with default value: "5060"* sip register secondary udp port
outbound-ip	Parameter type: <Epon::OutboundIp> Format: - IPv4-address	*optional parameter* sip out bound proxy ip address
[no] outbound-port	Parameter type: <Epon::OutboundPort> Format: - sip udp port number - range: [0...65535]	*optional parameter with default value: "5060"* sip out bound proxy udp port
[no] reg-intval	Parameter type: <Epon::RegInterval> Format: - reg refresh period - range: [0...4294967295]	*optional parameter with default value: "3600"* registering refresh period
[no] heartbeat-mode	Parameter type: <Epon::HeartBeatMode> Format: (enable | disable) Possible values: - enable : enable heart beat mode - disable : disable heart beat mode	*optional parameter with default value: "enable"* heart beat mode
[no] heartbeat-cycle	Parameter type: <Epon::HeartBeatCycle> Format: - heartbeat count - range: [0...65535]	*optional parameter with default value: "60"* heart beat cycle
[no] heartbeat-count	Parameter type: <Epon::HeartBeatCount> Format: - heartbeat count - range: [0...65535]	*optional parameter with default value: "3"* heart beat times
digit-map-prof	Parameter type: <Epon::DigitMapProfile> Format: - digitmap profile id	*mandatory parameter* digitmap profile id

Parameter	Type	Description
	- range: [1...32]	

40.49 Epon Ont Pots Configuration Command

Command Description

This command configures provisioning data associated with an ONT Pots Port.

User Level

The command can be accessed by operators with sip,megaco privileges, and executed by operators with sip,megaco privileges.

Command Syntax

The command has the following syntax:

> configure voice eont (no pots (uni-idx)) | (pots (uni-idx) [[no] admin-up] [no user-account | user-account <Epon::UserAcc>] [no user-name | user-name <Epon::UserName>] [no user-pwd | user-pwd <Epon::UserPwd>] [tid-name <Epon::TidName>])

Command Parameters

Table 40.49-1 "Epon Ont Pots Configuration Command" Resource Parameters

Resource Identifier	Type	Description
(uni-idx)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PonId> / <Eqpt::OntId> / <Epon::EontSlot> / <Epon::EontPort> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::PonId> - the PON identifier Field type <Eqpt::OntId> - the ONT identifier Field type <Epon::EontSlot> - Epon Ont Slot - range: [1...16] Field type <Epon::EontPort> - Epon Ont Port - range: [1...64]	ONT UNI interface

Table 40.49-2 "Epon Ont Pots Configuration Command" Command Parameters

Parameter	Type	Description
[no] admin-up	Parameter type: boolean	*optional parameter* admin status is up (read-only for voicefxs interface)

Parameter	Type	Description
[no] user-account	Parameter type: <Epon::UserAcc> Format: - sip user account - length: x<=16	*optional parameter with default value: ""* sip user account
[no] user-name	Parameter type: <Epon::UserName> Format: - user name - length: 1<=x<=32	*optional parameter with default value: ""* sip user name
[no] user-pwd	Parameter type: <Epon::UserPwd> Format: - sip user password hidden from users - length: x<=16	*optional parameter with default value: ""* sip user password
tid-name	Parameter type: <Epon::TidName> Format: - tid name - length: 1<=x<=32	*optional parameter* h248 tid name

40.50 Epon Ont Voice Common Configuration Command

Command Description

This command configures EPON ONT voice common parameters.

ip /mask /gateway - If ip-mode is not static IP, this parameter has no meaning. When ip-mode is static, this parameter should be configured.

svlan - If tag- mode is not stack, this parameter has no meaning.

User Level

The command can be accessed by operators with sip,megaco privileges, and executed by operators with sip,megaco privileges.

Command Syntax

The command has the following syntax:

> configure voice eont (no comm-para (ont-idx)) | (comm-para (ont-idx) [no ip-mode | ip-mode <Epon::IpMode>] [ip <Epon::IpAddr>] [mask <Epon::IpAddr>] [gateway <Epon::IpAddr>] [no pppoe-mode | pppoe-mode <Epon::pppoeMode>] [no pppoe-username | pppoe-username <Epon::pppoeUsrName>] [no pppoe-pwd | pppoe-pwd <Epon::pppoePwd>] [no tagged-mode | tagged-mode <Epon::TagMode>] cvlan <Epon::cvlan> [no svlan | svlan <Epon::svlan>] [no pbit | pbit <Epon::pbit>] [no t38-admin | t38-admin <Epon::faxMode>] [no fax | fax <Epon::ctrlMode>])

Command Parameters

Table 40.50-1 "Epon Ont Voice Common Configuration Command" Resource Parameters

Resource Identifier	Type	Description
(ont-idx)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PonId> / <Eqpt::OntId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::PonId> - the PON identifier Field type <Eqpt::OntId> - the ONT identifier	Ont Index

Table 40.50-2 "Epon Ont Voice Common Configuration Command" Command Parameters

Parameter	Type	Description
[no] ip-mode	Parameter type: <Epon::IpMode> Format: (static | dhcp | pppoe) Possible values: - static : static mode - dhcp : dhcp mode - pppoe : PPPoE mode	*optional parameter with default value: "static"* mode to get the voice ip address
ip	Parameter type: <Epon::IpAddr> Format: - IPv4-address	*optional parameter* static voice ip address
mask	Parameter type: <Epon::IpAddr> Format: - IPv4-address	*optional parameter* subnet of the voice ip
gateway	Parameter type: <Epon::IpAddr> Format: - IPv4-address	*optional parameter* default gateway of the voice ip
[no] pppoe-mode	Parameter type: <Epon::pppoeMode> Format: (auto | chap | pap) Possible values: - auto : AUTO mode - chap : CHAP mode - pap : PAP mode	*optional parameter with default value: "auto"* PPPoE configure mode
[no] pppoe-username	Parameter type: <Epon::pppoeUsrName> Format: - user name - length: 1<=x<=32	*optional parameter with default value: ""* PPPoE user name
[no] pppoe-pwd	Parameter type: <Epon::pppoePwd> Format: - PPPoE password hidden from users - length: 1<=x<=32	*optional parameter with default value: ""* PPPoE password
[no] tagged-mode	Parameter type: <Epon::TagMode> Format: (transparent | tag | stacking) Possible values: - transparent : transparent mode - tag : tagged mode - stacking : stacking mode	*optional parameter with default value: "tag"* vlan tag mode
cvlan	Parameter type: <Epon::cvlan> Format: - Vlan id - range: [0...4093]	*mandatory parameter* customer vlan for voice
[no] svlan	Parameter type: <Epon::svlan> Format: - Vlan id - range: [2...4093,0]	*optional parameter with default value: 0L* Service vlan for voice
[no] pbit	Parameter type: <Epon::pbit> Format: - DBA polling level - range: [0...7]	*optional parameter with default value: 5L* dot1p of voice flow
[no] t38-admin	Parameter type: <Epon::faxMode>	*optional parameter with default*

Parameter	Type	Description
	Format: (t30 | t38) Possible values: - t30 : t30 mode - t38 : t38 mode	*value: "t30"* choose fax mode
[no] fax	Parameter type: <Epon::ctrlMode> Format: (negotiation | autovbd) Possible values: - negotiation : Negotiation mode - autovbd : auto VBD mode	*optional parameter with default value: "negotiation"* fax or modem control mode

40.51 Epon Ont Voice media gateway Configuration Command

Command Description

This command configures EPON ONT voice media gateway parameters.

gw-id - Identifier of media gateway. It is only valid if reg-mode is domain or equip.

tid-digit-len - tid digit length. Tid digit length + prefix length should be less than or equal to 32

User Level

The command can be accessed by operators with sip,megaco privileges, and executed by operators with sip,megaco privileges.

Command Syntax

The command has the following syntax:

> configure voice eont (no media-gateway (ont-idx)) | (media-gateway (ont-idx) [no udp-port | udp-port <Epon::udp>] mgc-pri-ip <Epon::IpAddr> [no mgc-pri-port | mgc-pri-port <Epon::udp>] [no mgc-sec-ip | mgc-sec-ip <Epon::IpAddr>] [no mgc-sec-port | mgc-sec-port <Epon::udp>] [no reg-mode | reg-mode <Epon::regMode>] [gw-id <Epon::gwid>] [no heartbeat-mode | heartbeat-mode <Epon::HBMode>] [no heartbeat-cycle | heartbeat-cycle <Epon::HBCycle>] [no heartbeat-count | heartbeat-count <Epon::HBVal>] tid-num <Epon::TidNum> tid-prefix <Epon::TidPrefix> tid-digit-begin <Epon::TidDigit> tid-mode <Epon::TidMode> tid-digit-len <Epon::TidLen>)

Command Parameters

Table 40.51-1 "Epon Ont Voice media gateway Configuration Command" Resource Parameters

Resource Identifier	Type	Description
(ont-idx)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PonId> / <Eqpt::OntId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::PonId> - the PON identifier Field type <Eqpt::OntId> - the ONT identifier	Ont Index

Table 40.51-2 "Epon Ont Voice media gateway Configuration Command" Command Parameters

Parameter	Type	Description
[no] udp-port	Parameter type: <Epon::udp> Format: - DBA polling interval per pon interface - range: [0...65535]	*optional parameter with default value: 2944* signal udp port of media gateway
mgc-pri-ip	Parameter type: <Epon::IpAddr> Format: - IPv4-address	*mandatory parameter* primary ip address of MGC
[no] mgc-pri-port	Parameter type: <Epon::udp> Format: - DBA polling interval per pon interface - range: [0...65535]	*optional parameter with default value: 2944* udp port of the primary MGC
[no] mgc-sec-ip	Parameter type: <Epon::IpAddr> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* secondary ip address of MGC
[no] mgc-sec-port	Parameter type: <Epon::udp> Format: - DBA polling interval per pon interface - range: [0...65535]	*optional parameter with default value: 2944* udp port of the secondary MGC
[no] reg-mode	Parameter type: <Epon::regMode> Format: (ip | domain | equip) Possible values: - ip : - domain : - equip :	*optional parameter with default value: "ip"* register mode of media gateway
gw-id	Parameter type: <Epon::gwid> Format: - media gateway identifier - length: x<=64	*optional parameter* corresponding ID for a given Media Gw
[no] heartbeat-mode	Parameter type: <Epon::HBMode> Format: (enabled | disabled) Possible values: - enabled : heartbeat function enabled - disabled : heartbeat function disabled	*optional parameter with default value: "enabled"* link checking mode
[no] heartbeat-cycle	Parameter type: <Epon::HBCycle> Format: - Heart Beat cycle - unit: secs - range: [0...65535]	*optional parameter with default value: 60L* heart beat cycle
[no] heartbeat-count	Parameter type: <Epon::HBVal> Format: - DBA polling interval per pon interface - range: [0...65535]	*optional parameter with default value: 3L* The heart beat times
tid-num	Parameter type: <Epon::TidNum> Format: - Number of rtp tid - range: [0...255]	*mandatory parameter* Number of rtp tid
tid-prefix	Parameter type: <Epon::TidPrefix> Format: - an octet string defining mnemonic of EPON - length: 1<=x<=16	*mandatory parameter* prefix of rtp tid
tid-digit-begin	Parameter type: <Epon::TidDigit> Format:	*mandatory parameter* begin number of tid digit

Parameter	Type	Description
	- DBA polling interval per pon interface - range: [0...65535]	
tid-mode	Parameter type: <Epon::TidMode> Format: (inline | notinline) Possible values: - inline : - notinline :	*mandatory parameter* tid digit line up mode
tid-digit-len	Parameter type: <Epon::TidLen> Format: - - range: [1...31]	*mandatory parameter* tid digit length

40.52 Voice Sip LSA Server Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip LSA Server profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice lsa-server (no system (name)) | (system (name) slot-id <LSA::SlotIndex> country-code <LSA::CountryCode> area-code <LSA::AreaCode> [no admin-status | admin-status <LSA::SystemAdminStatus>])

Command Parameters

Table 40.52-1 "Voice Sip LSA Server Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(name)	Format: - uniquely identify this LSA server system - length: 1<=x<=32	uniquely identify this voice LSA server system

Table 40.52-2 "Voice Sip LSA Server Configuration Commands" Command Parameters

Parameter	Type	Description
slot-id	Parameter type: <LSA::SlotIndex> Format: (lt : <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> | nt | <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::EqSlotId> | vvps-a) Possible values: - lt : 7302/7330/7360 ISAM FTTN : slot id of the NVPS-C board at which LSA server is hosted - nt : 7363 MX : slot id of NT board - vvps-a : 7302/7330/7360: slot id of the VVPS-A at which the LSA server is hosted Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number	*mandatory parameter* 7302/7330/7360 ISAM FTTN : slot-id of the physical NVPS-C board while LSA server is hosted at physical NVPS-C board; slot id of the VVPS-A if LSA server is hosted at VVPS-A board (only applicable on NANT-E/FANT-F). 7363 MX : slot id of NT board

Parameter	Type	Description
	Field type <Eqpt::EqSlotId> - the equipment slot number	
country-code	Parameter type: <LSA::CountryCode> Format: - country code of the LSA server system - length: 1<=x<=32	*mandatory parameter* country code used by this LSA server to complete the DN received in an incoming sip invite request
area-code	Parameter type: <LSA::AreaCode> Format: - area code of the LSA server system - length: 1<=x<=32	*mandatory parameter* area code used by this LSA server to complete the DN received in an incoming sip invite request
[no] admin-status	Parameter type: <LSA::SystemAdminStatus> Format: (up | down) Possible values: - up : unlock the LSA server system - down : lock the LSA server system	*optional parameter with default value: "down"* administrative status of the LSA server

40.53 Voice Sip LSA Server Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip LSA Server profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice lsa-server (no instance (name)) | (instance (name) slot-id <LSA::SlotIndex> sign-ip <LSA::IpAddressAndMask> [no sign-port | sign-port <LSA::InstanceSignPort>] sign-vlan <LSA::InstanceSignVlan> [no sign-gw-ip | sign-gw-ip <LSA::IpAddress>] [no admin-status | admin-status <LSA::InstanceAdminStatus>] [no sg-name | sg-name <LSA::InstanceSGDomName>] [no lsa-md5-realm | lsa-md5-realm <LSA::MD5Realm>] [no lsa-md5-password | lsa-md5-password <Security::Password4>])

Command Parameters

Table 40.53-1 "Voice Sip LSA Server Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(name)	Format: - uniquely identify this LSA server instance - length: 1<=x<=32	uniquely identify this LSA server instance

Table 40.53-2 "Voice Sip LSA Server Configuration Commands" Command Parameters

Parameter	Type	Description
slot-id	Parameter type: <LSA::SlotIndex> Format: (lt : <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> | nt | <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::EqSlotId> | vvps-a) Possible values: - lt : 7302/7330/7360 ISAM FTTN : slot id of the NVPS-C board at which LSA server is hosted - nt : 7363 MX : slot id of NT board - vvps-a : 7302/7330/7360: slot id of the VVPS-A at which the LSA server is hosted Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number	*mandatory parameter* 7302/7330/7360 ISAM FTTN : slot-id of the physical NVPS-C board while LSA server is hosted at physical NVPS-C board; slot id of the VVPS-A if LSA server is hosted at VVPS-A board (only applicable on NANT-E/FANT-F). 7363 MX : slot id of the NT board

Parameter	Type	Description
	Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::EqSlotId> - the equipment slot number	
sign-ip	Parameter type: <LSA::IpAddressAndMask> Format: ipv4 : <Ip::V4Address> / <LSA::PrefixLength> Possible values: - ipv4 : the address type of the LSA server instance is ipv4 Field type <Ip::V4Address> - IPv4-address Field type <LSA::PrefixLength> - prefix length of the subnet - range: [0...32]	*mandatory parameter* signaling IP address + prefix length used by this LSA server instance
[no] sign-port	Parameter type: <LSA::InstanceSignPort> Format: - the SIP LSA server Instance port - range: [1...65534]	*optional parameter with default value: 5060* signaling port used by this LSA server instance
sign-vlan	Parameter type: <LSA::InstanceSignVlan> Format: - vlan id of LSA server instance - range: [1...4093]	*mandatory parameter* signaling vlan ID used by this LSA server instance
[no] sign-gw-ip	Parameter type: <LSA::IpAddress> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* default signaling gateway IP address used by this LSA server instance
[no] admin-status	Parameter type: <LSA::InstanceAdminStatus> Format: (up | down) Possible values: - up : unlock the LSA server instance - down : lock the LSA server instance	*optional parameter with default value: "down"* administrative status of this LSA server instance
[no] sg-name	Parameter type: <LSA::InstanceSGDomName> Format: - uniquely name of this element - length: x<=32	*optional parameter with default value: "vsp1"* name of the service gateway associated with this LSA server instance
[no] lsa-md5-realm	Parameter type: <LSA::MD5Realm> Format: - the string for MD5 - length: x<=64	*optional parameter with default value: ""* realm used by this LSA server instance to authenticate an incoming SIP register request
[no] lsa-md5-password	Parameter type: <Security::Password4> Format: (prompt | plain : <Security::PlainPassword4>) Possible values: - prompt : prompts the operator for a password - plain : the password in plain text, may contain any printable character defined by the US-ASCII character set and that has a decimal code in the range 33 to 126, and the length should be less than or equal to 64 (Due to legacy reasons, the MD5 character string was included in the parameter name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a	*optional parameter with default value: "plain : "* password used by this LSA server instance to authenticate an incoming SIP register request

Parameter	Type	Description
	common configuration object for all authentication types that might require the configuration of an authentication password.) Field type <Security::PlainPassword4> - the password, may contain any printable character defined by the US-ASCII character set and that has a decimal code in the range 33 to 126, and the length should be less than or equal to 64 (Due to legacy reasons, the MD5 character string was included in the parameter name. Nevertheless, despite the MD5 string in the name of this object, from R5.8.x it is to be considered as a common configuration object for all authentication types that might require the configuration of an authentication password.) - length: x<=64	

40.54 Voice Sip LSA Server Emergency Service Configuration Commands

Command Description

This command allows the operator to manage the Voice Sip LSA Server Emergency Service profile.

User Level

The command can be accessed by operators with sip privileges, and executed by operators with sip privileges.

Command Syntax

The command has the following syntax:

> configure voice lsa-server (no emergency-service dn <LSA::Dn> gateway-dn <LSA::GtwyDn>) | (emergency-service dn <LSA::Dn> gateway-dn <LSA::GtwyDn> slot-id <LSA::SlotIndex> [no name | name <LSA::EmergServiceName>] [no sg-name | sg-name <LSA::EmergVsgName>] [no gateway-ddidn | gateway-ddidn <LSA::GtwyDDIDn>] [no gateway-dn-type | gateway-dn-type <LSA::GtwyDnType>])

Command Parameters

Table 40.54-1 "Voice Sip LSA Server Emergency Service Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
dn	Parameter type: <LSA::Dn> Format: - emergency service directory number - length: 1<=x<=8	emergency service directory number
gateway-dn	Parameter type: <LSA::GtwyDn> Format: - directory number only includes the subscriber number to which the emergency call is to be addressed. - length: 1<=x<=32	directory number only includes the subscriber number to which the emergency call is to be addressed

Table 40.54-2 "Voice Sip LSA Server Emergency Service Configuration Commands" Command Parameters

Parameter	Type	Description
slot-id	Parameter type: <LSA::SlotIndex> Format: (lt : <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> | nt | <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::EqSlotId> | vvps-a) Possible values: - lt : 7302/7330/7360 ISAM FTTN : slot id of the NVPS-C board at which LSA server is hosted	*mandatory parameter* 7302/7330/7360 ISAM FTTN : slot-id of the physical NVPS-C board while LSA server is hosted at physical NVPS-C board; slot id of the VVPS-A if LSA server is hosted at VVPS-A board (only applicable on NANT-E/FANT-F). 7363 MX :

Parameter	Type	Description
	- nt : 7363 MX : slot id of NT board - vvps-a : 7302/7330/7360: slot id of the VVPS-A at which the LSA server is hosted Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::EqSlotId> - the equipment slot number	slot id of NT board
[no] name	Parameter type: <LSA::EmergServiceName> Format: - the name of LSA emergency service - length: x<=32	*optional parameter with default value: ""* name of this LSA emergency service
[no] sg-name	Parameter type: <LSA::EmergVsgName> Format: - uniquely name of this element - length: x<=32	*optional parameter with default value: "vsp1"* name of the Voice SERVICE GATEWAY associated with this LSA emergency service
[no] gateway-ddidn	Parameter type: <LSA::GtwyDDIDn> Format: - the ISDN DDI number which the emergency call is to be addressed. - length: x<=32	*optional parameter with default value: ""* the ISDN DDI number which the emergency call is to be addressed
[no] gateway-dn-type	Parameter type: <LSA::GtwyDnType> Format: (pots | isdn-pra) Possible values: - pots : the gateway-dn is a POTS number. - isdn-pra : the gateway-dn is an ISDN PRA number	*optional parameter with default value: "pots"* the type of the gateway-dn

40.55 Voice extline Configuration Commands

Command Description

This command allows the operator to manage the Voice extline profile.

User Level

The command can be accessed by operators with all privileges, and executed by operators with all privileges.

Command Syntax

The command has the following syntax:

> configure voice (no extline (if-index)) | (extline (if-index) local-ip <ExtLine::IpAddressAndMask> [no gateway-ip | gateway-ip <ExtLine::ExtLineIpAddr>] [no ip-addr-policy | ip-addr-policy <ExtLine::IpAddrPolicy>] vlan-id <ExtLine::ExtLineVlanId> fxo-ip <ExtLine::FxoIpAddr> fxo-port <ExtLine::ExtLinePort> [no admin-status | admin-status <ExtLine::ExtLineAdminStatus>] [no codec | codec <ExtLine::ExtLineCodec>] [no ptime | ptime <ExtLine::ExtLinePtime>] [no echo-cancel | echo-cancel <ExtLine::ExtLineEc>] [no silence-suppr | silence-suppr <ExtLine::ExtLineSilence>] [no impedance | impedance <ExtLine::ExtLineImpedance>] [no rx-gain | rx-gain <ExtLine::ExtLineRxGain>] [no tx-gain | tx-gain <ExtLine::ExtLineTxGain>] [no dscp | dscp <ExtLine::ExtLineDscp>] [no pbits | pbits <ExtLine::ExtLinePbits>])

Command Parameters

Table 40.55-1 "Voice extline Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(if-index)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::PortId> - the port number	The unique internal identifier of the extension line port

Table 40.55-2 "Voice extline Configuration Commands" Command Parameters

Parameter	Type	Description
local-ip	Parameter type: <ExtLine::IpAddressAndMask> Format: <Ip::V4Address> / <ExtLine::PrefixLength> Field type <Ip::V4Address> - IPv4-address Field type <ExtLine::PrefixLength> - prefix length of the subnet - range: [0...32]	*mandatory parameter* identify the IP address of this extension line

Parameter	Type	Description
[no] gateway-ip	Parameter type: <ExtLine::ExtLineIpAddr> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* IP address of the gateway, must be set when manual mode and ip-address-policy is distributed, otherwise it is optional
[no] ip-addr-policy	Parameter type: <ExtLine::IpAddrPolicy> Format: (distributed | centralized) Possible values: - distributed : the ip address policy of the user agent operation is to use an IP address per Voice LT, not applicable for NGVR and ANSI MDU - centralized : the ip address policy of the user agent operation is to use a single IP address for the ISAM Voice, NGVR and ANSI MDU	*optional parameter with default value: "distributed"* The ip address policy of the extension line
vlan-id	Parameter type: <ExtLine::ExtLineVlanId> Format: - vlan id - range: [1...4093]	*mandatory parameter* vlan id for rtp packet
fxo-ip	Parameter type: <ExtLine::FxoIpAddr> Format: - IPv4-address	*mandatory parameter* identify the IP address of this fxo
fxo-port	Parameter type: <ExtLine::ExtLinePort> Format: - the extension line local port - range: [1...65534]	*mandatory parameter* port of the fxo
[no] admin-status	Parameter type: <ExtLine::ExtLineAdminStatus> Format: (up | down) Possible values: - up : the status of the user agent operation is up - down : the status of the user agent operation is down	*optional parameter with default value: "down"* identify the status of this line administration
[no] codec	Parameter type: <ExtLine::ExtLineCodec> Format: (g711-a-law | g711-u-law | g723 | g729) Possible values: - g711-a-law : the extension line codec is g711-a-law - g711-u-law : the extension line codec is g711-u-law - g723 : the extension line codec is g723 - g729 : the extension line codec is g729	*optional parameter with default value: "g711-a-law"* configure the line codec of this extension line
[no] ptime	Parameter type: <ExtLine::ExtLinePtime> Format: - value of the extension line package time - unit: milliseconds - range: [10...60]	*optional parameter with default value: "20"* configure the line ptime of this extension line,ptime accepts values only in multiples of 10.if codec is configuread as g723,the ptime accepts only 30 and 60.if codec is configuread as g711-a-law,g711-u-law,g729,the ptime can accept:10,20,30,40,50,60

Parameter	Type	Description
[no] echo-cancel	Parameter type: <ExtLine::ExtLineEc> Format: (on | off) Possible values: - on : the extension line echo canceller is on - off : the extension line echo canceller is off	*optional parameter with default value: "off"* configure the line EC of this extension line
[no] silence-suppr	Parameter type: <ExtLine::ExtLineSilence> Format: (on | off) Possible values: - on : the extension line echo canceller is on - off : the extension line echo canceller is off	*optional parameter with default value: "on"* configure the line silence support of this extension line
[no] impedance	Parameter type: <ExtLine::ExtLineImpedance> Format: (cdeprofile | 200 | 220 | 220minisplitter | 270 | 300 | 370 | 600 | 900 | 370minisplitter | 600splitter | 370nvlsasplitter | 370skinnysplitter | 270duratelsplitter | 2703msplitter | 100 | 150 | 220splitter | reserv19 | reserv20 | reserv21) Possible values: - cdeprofile : the impedance is default value - 200 : the line impedance is 200 ohm - 220 : the line impedance is 220 ohm - 220minisplitter : the line impedance is 220 ohm+ miniSplitter - 270 : the line impedance is 270 ohm - 300 : the line impedance is 300 ohm - 370 : the line impedance is 370 ohm - 600 : the line impedance is 600 ohm - 900 : the line impedance is 900 ohm - 370minisplitter : the line impedance is 370 ohm+ miniSplitter - 600splitter : the line impedance is 600 ohm+splitter - 370nvlsasplitter : the line impedance is 370 ohm+ NVLSASplitter - 370skinnysplitter : the line impedance is 370 ohm+ SkinnySplitter - 270duratelsplitter : the line impedance is 270 ohm+ DuratelSplitter	*optional parameter with default value: "cdeprofile"* configure the line impedance of this extension line

Parameter	Type	Description
	- 2703msplitter : the line impedance is 270 ohm+ 3MDplitter - 100 : the line impedance is 100 ohm - 150 : the line impedance is 150 ohm - 220splitter : the line impedance is 220 ohm+ splitter - reserv19 : reserved for future use 19 - reserv20 : reserved for future use 20 - reserv21 : reserved for future use 21	
[no] rx-gain	Parameter type: <ExtLine::ExtLineRxGain> Format: - the line rx_gain range -14...6 - range: [-14...6]	*optional parameter with default value: "0"* configure the line rx_gain of this extension line
[no] tx-gain	Parameter type: <ExtLine::ExtLineTxGain> Format: - the line tx_gain range -14...6 - range: [-14...6]	*optional parameter with default value: "0"* configure the line tx_gain of this extension line
[no] dscp	Parameter type: <ExtLine::ExtLineDscp> Format: - dscp mark for rtp or rtcp packets - range: [0...63]	*optional parameter with default value: 46* dscp mark for extension line packets
[no] pbits	Parameter type: <ExtLine::ExtLinePbits> Format: - the dot-1p bit value - range: [0...7]	*optional parameter with default value: 6* 802.1p bits for extension line packets.

40.56 Voice Pseudo Wire Configuration Commands

Command Description

This command allows the operator to manage the Voice pseudo wire profile.

User Level

The command can be accessed by operators with all privileges, and executed by operators with all privileges.

Command Syntax

The command has the following syntax:

> configure voice (no pseudo-wire (if-index)) | (pseudo-wire (if-index) channelset <PwEncapTdm::ChannelSet> [no ip-addr-policy | ip-addr-policy <PwEncapTdm::IpAddrPolicy>] local-ip <PwEncapTdm::IpAddressAndMask> dest-ip <PwEncapTdm::IpAddress> dest-udp-port <PwEncapTdm::UDPPort> [no gateway-ip | gateway-ip <PwEncapTdm::IpAddress>] vlan-id <PwEncapTdm::VlanId> [no p-bits | p-bits <PwEncapTdm::Pbits>] [no dscp | dscp <PwEncapTdm::Dscp>] [no admin-status | admin-status <PwEncapTdm::AdminStatus>] [no data-jitter | data-jitter <PwEncapTdm::DataJitter>] p-time <PwEncapTdm::PacketInterval> [no payload-type | payload-type <PwEncapTdm::PayloadType>])

Command Parameters

Table 40.56-1 "Voice Pseudo Wire Configuration Commands" Resource Parameters

Resource Identifier	Type	Description
(if-index)	Format: <Eqpt::RackId> / <Eqpt::ShelfId> / <Eqpt::SlotId> / <Eqpt::PortId> Field type <Eqpt::RackId> - the rack number Field type <Eqpt::ShelfId> - the shelf number Field type <Eqpt::SlotId> - the LT slot number Field type <Eqpt::PortId> - the port number	The unique internal identifier of the pseudo wire port

Table 40.56-2 "Voice Pseudo Wire Configuration Commands" Command Parameters

Parameter	Type	Description
channelset	Parameter type: <PwEncapTdm::ChannelSet> Format: - channels for pseudo wire - length: 1<=x<=31	*mandatory parameter* assign channels for pseudo wire. master DS0 Channel must appear as the first one in the DS0 channel list. Total pseudo wire DS0 should be less than 193 on every NIAT-A pack
[no] ip-addr-policy	Parameter type: <PwEncapTdm::IpAddrPolicy> Format:	*optional parameter with default value: "pw-distinct-ip-mode"*

Parameter	Type	Description
	(pw-distinct-ip-mode | pw-shared-ip-mode) Possible values: - pw-distinct-ip-mode : the ip address policy of the pseudo wire, operation is to use an IP address per NIAT-A board - pw-shared-ip-mode : the ip address policy of the pseudo wire, operation is to use a single IP address for the PW traffic	The ip address policy of the pseudo wire
local-ip	Parameter type: <PwEncapTdm::IpAddressAndMask> Format: <Ip::V4Address> / <PwEncapTdm::PrefixLength> Field type <Ip::V4Address> - IPv4-address Field type <PwEncapTdm::PrefixLength> - prefix length of the subnet - range: [0...32]	*mandatory parameter* identify the IP address and prefix length of this pseudo wire
dest-ip	Parameter type: <PwEncapTdm::IpAddress> Format: - IPv4-address	*mandatory parameter* the destination IP address of this pseudo wire
dest-udp-port	Parameter type: <PwEncapTdm::UDPPort> Format: - the udp port of pseudo wire - range: [1...65535]	*mandatory parameter* the destination udp port of pseudo wire
[no] gateway-ip	Parameter type: <PwEncapTdm::IpAddress> Format: - IPv4-address	*optional parameter with default value: "0.0.0.0"* IP address of the gateway, must be set when manual mode and ip-address-policy is pw-distinct-ip-mode, otherwise it is optional
vlan-id	Parameter type: <PwEncapTdm::VlanId> Format: - vlan id - range: [1...4093]	*mandatory parameter* vlan id for pseudo wire packet
[no] p-bits	Parameter type: <PwEncapTdm::Pbits> Format: - the dot-1p bit value - range: [0...7]	*optional parameter with default value: 6* 802.1p bits for pseudo wire packets
[no] dscp	Parameter type: <PwEncapTdm::Dscp> Format: - dscp mark for rtp or rtcp packets - range: [0...63]	*optional parameter with default value: 46* dscp mark for pseudo wire packets
[no] admin-status	Parameter type: <PwEncapTdm::AdminStatus> Format: (up | down) Possible values: - up : unlock the pseudo wire - down : lock the pseudo wire	*optional parameter with default value: "down"* identify the status of this pseudo wire administration
[no] data-jitter	Parameter type: <PwEncapTdm::DataJitter> Format: - jitter buffer for PW channel in millisecond - unit: millisecond - range: [0...200]	*optional parameter with default value: 50* data jitter buffer for pseudo wire.\nwhen p-time = 1, the range is [0...100]
p-time	Parameter type: <PwEncapTdm::PacketInterval> Format: - packetization interval for pseudo wire in millisecond	*mandatory parameter* packetization interval for pseudo wire packets.\nThe valid value is

Parameter	Type	Description
	- unit: millisecond - range: [1...10]	linked with the number of DS0(N) by the following relationship:\nN * p-time <= 31.\nThe package size is 8*N*packetization interval
[no] payload-type	Parameter type: <PwEncapTdm::PayloadType> Format: - payload type for pseudo wire packets - range: [96...127]	*optional parameter with default value: 96* payload type for pseudo wire packets