VoIP SVI Resource Glossary

SQ00396

V2.5

Change History

Version	Change Made	Author	Authorised	Date
1	Initial Release	Various	MW	25.1.13
2.2	Added VOIP Stack option Radius-Alway-Use-From-Field (svi-mgc.v11_1_4.FC5)	JF	JF	29.1.14
2.3	Added VoIP Destination RTP Options new value Secure-RTP	BT	BT	05.2.14
2.4	Added new voip destination Options Delayed-FS-TCS-Negotiation	BT	BT	14.4.14
2.5	Updated VD:options (ringback on/off/sdp) Added VS:Stateless proxy methods and VS:options – encrypt proxy parameters	JE	JE	14.5.14

Content

1.0 Introduction

The SVI controls all aspects of the system by abstracting the system down to a number of database resources. These resources are further divided into sub resource types that describe the operation of a given resource. There are a number of instances of a given resource and each resource has a number of configurable attributes.

These resources are used to allow the administrator to configure and maintain the system.

All of these resources are arranged into a parent-child hierarchical arrangement. When started up the SVI will bring the resources into service starting with the system resource and continuing down through the children resources. A resource will only be brought into service once its parent resource is configured and in-service.

Each resource has the following components

1.1 Resource Name

Each resource has a unique name which provides a unique operation for the SVI product.

1.2 Resource Identifier

An individual resource instance is identified by a unique name.

1.3 Attribute

The attribute field is the attribute of the resource that is to be configured. These are described for each resource within this document.

1.4 Value

This field contains the value of the attribute. The following type of values are valid:

Type	Description
Integer	This is a number in either decimal or hexadecimal format. If the number is in hexadecimal format the 'H' character is prefixed to the number to indicate it is a hexadecimal number (Ha3)
Drop Down List	This type of value provides a definitive list of options from which the user can select one of the options
Resource Identifier	This is a drop down list which will provide all configured resources of the same type allowing the user to select a single instance of a resource
Bit Mask Flag	This type of value provides the user with a list of options from which the user can choose multiple choices.
Integer range	The integer range field is a list including ranges of integers for example 1,2,3-10,12-17,20
User String	This is a NULL terminated ASCII character string
IP Address	This describes an IP address ie (192.168.2.182)
Boolean	A Boolean flag is either True or False
Resource Identifier Range	This allows for multiple resources to be identified
Digit Match	A digit match is a string of E164 numbers with the special characters '

2.0 Resources

2.1 Common Attributes

The following section contains attributes that are common to all resources

2.1.1 Status

This defines the status of the attribute, which will either be in service (INS) or out of service (OOS).

2.1.1.1 Description

This defines the status of the resource instance. When configuring managed resources this should be set to Start, all other resource types should be INS.

2.1.1.2 Definition

Applicable	Modes	User Type
All	Read Write Licensed	Drop Down List

2.1.1.3 Values

Value	Description
Niu	Instance of resource not configured on system
Osf	Resource is waiting for parent to be in service before it attempts to come into service
Oos	The Resource has been manually removed from service
Osc	The Resource is waiting for the parent to come into service before it attempts to configure this resource
Osd	The resource's parent is out of service and will come straight back into service once the resource's parent is in service.
Osfd	The
Oscp	The resource is currently attempting to be configured
Osp	The resource is currently attempting to come into service
INS	The resource is INSERVICE
Start	This pre-configures the resource to its correct startup state
ERR	An Error has occurred on this resource.

2.1.2 Flags

2.1.2.1 Description

The following flags define internally the behaviour of the resource. These are not user defined and are provided only for information purposes.

2.1.2.2 Definition

Applicable	Modes	User Type
All	Read	None

2.1.2.3 Values

Value	Description
Secure	This resource will be secured on a fail over
Applied	This indicates that a resource has been reconfigured and the system needs to reapply the configuration to the resource owner
Add	Indicates that the resource is scheduled to be added to the system hierarchy
Remove	Indicates that the resource is scheduled to be removed from the system hierarchy
Start	Indicates that the resource is scheduled to be started
Reconfigured	Indicates that the resource has been reconfigured and requires to be Removed then added and then started
Read	This attribute has read permissions
Write	This attribute has write permissions
No-Default	This resource will not be displayed in the CLI if the default value is configured
No-Secure	Do not secure this resource at failover
No-Init	Do not initialise this
No-GUI	Do not display this attribute in the GUI
Licensed	The number of instances of this resource is licensed

2.1.3 Inservice

2.1.3.1 Description

The Inservice attribute displays the time that this resource came into service.

2.1.3.2 Definition

Applicable	Modes	User Type
All resources	Read	Time

2.1.4 Outservice

2.1.4.1 Description

The Outservice attribute displays the time that this resource last went from an in service state to an out of service state.

2.1.4.2 Definition

Applicable	Modes	User Type
All resources	Read	Time

2.1.5 Description

2.1.5.1 Description

Description is a free text field to allow for a human readable name of the resource. Please note that this field is automatically populated by the SVI-MS from the “Name” input field.

2.1.5.2 Definition

Applicable	Modes	User Type
	Read Write No-Default	

2.1.6 Options

2.1.6.1 Description

These are a set of options which relate to the resource in question that you can enable or disable at any time.

2.1.6.2 Definition

Applicable	Modes	User Type
	Read Write No-Default	

2.1.6.3 Values

Value	Description
SNMP-Trap	If the resource changes state, an SNMP trap will be sent to the address configured on the debug resource.
Radius-Alway-Use-From-Field	This will always use the from Field in the Invite for Radius Authentication

2.2 IP

2.2.1 Description

This defines the IP sockets used to provide the SIP Service. The IP definition is connected to the VoIP Stack to specify the service listening address.

2.2.2 Products

SVI-MG | SVI-MGC | SVI-C4 | SVI-C5 | SVI-SBC

2.2.3 Attributes

2.2.3.1 Type

2.2.3.1.1 Description

The behaviour type for the resource instance.

2.2.3.1.2 Definition

Applicable	Modes	User Type
UDP SIP Service	Read Write	Drop down list

2.2.3.1.3 Values

Sip IP	For SIP UDP this should be set top this value

2.2.3.2 Local Address

2.2.3.2.1 Description

This defines the local address on the SVI that the SIP service will run on.

2.2.3.2.2 Definition

Applicable	Modes	User Type
UDP SIP Service	Read Write No-Default	IP address

2.2.3.3 Port

2.2.3.3.1 Description

This defines the local address port on the SVI that the SIP service will run on.

2.2.3.3.2 Definition

Applicable	Modes	User Type
UDP SIP Service	Read Write	Integer

2.2.3.4 Socket Type

2.2.3.4.1 Description

This defines the type of socket used. This should always be set to UDP peer. This resource is not used for other SIP transport.

2.2.3.4.2 Definition

Applicable	Modes	User Type
UDP SIP Service	Read Write	

2.2.3.4.3 Values

Value	Description
UDP Peer	

2.2.3.5 Name

2.2.3.5.1 Description

This must have a unique user name across all IP resources. This name is used to assist with debugging.

2.2.3.5.2 Definition

Applicable	Modes	User Type
UDP SIP Service	Read Write	Unique user string

2.2.3.6 MNT Owner

2.2.3.6.1 Description

This describes the internal process that handles the maintenance actions of this socket. This should always be set to "Maintenance".

2.2.3.6.2 Definition

Applicable	Modes	User Type
UDP SIP Service	Read Write	User string set to "SIP"

2.2.3.7 Packet Owner

2.2.3.7.1 Description

This describes the internal process that handles the received UDP packages. This should always be set to "SIP".

2.2.3.7.2 Definition

Applicable	Modes	User Type
UDP SIP Service	Read Write	User string set to "SIP"

2.2.3.8 Automatic

2.2.3.8.1 Description

This specifies if the IP process should automatically recover a failed socket. This should always be set to “True”.

2.2.3.8.2 Definition

Applicable	Modes	User Type
UDP SIP Service	Read Write	Boolean always set to True

2.2.3.9 VIP

2.2.3.9.1 Description

This defines if the IP being used for SIP service is on a VIP (Virtual IP Address). This should always be set to “True” for redundant systems.

2.2.3.9.2 Definition

Applicable	Modes	User Type
UDP SIP Service	Read Write	False = Non Redundant True = Redundant

2.2.3.10 Options

2.2.3.10.1 Description

These bit mask flags allow for additional functionality to be added to the IP resource.

2.2.3.10.2 Definition

Applicable	Modes	User Type
UDP SIP Service	Read Write No-Default	Bit mask options

2.2.3.10.3 Values

Value	Description
SNMP-Trap	If this option is applied

2.2.3.11 IP TOS

2.2.3.11.1 Description

This attribute along with IP Priority allows for QoS to be added to this socket. This attribute sets the IP type of service and the precedence.

2.2.3.11.2 Definition

Applicable	Modes	User Type
UDP SIP Service	Read Write No-Default	Integer

2.2.3.12 IP Priority

2.2.3.12.1 Description

This attribute along with IP TOS allows for QoS to be added to this socket. This attribute sets the IP level.

2.2.3.12.2 Definition

Applicable	Modes	User Type
UDP SIP Service	Read Write No-Default	integer

2.2.3.13 Tx Stats

2.2.3.13.1 Description

This attribute references to a Statistics resource which contains the transmission statistics of the IP resource

2.2.3.13.2 Definition

Applicable	Modes	User Type
UDP SIP Service	Read Write	Resource identifier

2.3 VoIP Stack

2.3.1 Description

The VoIP Stack defines the service access point of a VoIP Service. Multiple VoIP stacks can be applied to run multiple services on different protocols and on different IP addresses and subnets.

2.3.2 Products

SVI-MG | SVI-MGC | SVI-C4 | SVI-C5 | SVI-SBC

2.3.3 Attributes

2.3.3.1 Type

2.3.3.1.1 Description

The behaviour type of the resource instance.

2.3.3.1.2 Definition

Applicable	Modes	User Type
All defined products	Read Write	Drop Down List

2.3.3.1.3 Values

Value	Description
H323 Stack	This VoIP Stack is running as a H323 service
SIP Stack	This VoIP Stack is running as a SIP service

2.3.3.2 Address

2.3.3.2.1 Description

This defines the IP Address of the VoIP Stack's service.

2.3.3.2.2 Definition

Applicable	Modes	User Type
All defined products	Read Write No-Default	IP Address

2.3.3.3 Port

2.3.3.3.1 Description

This defines the IP Address Port of the VoIP Stack' service

2.3.3.3.2 Definition

Applicable	Modes	User Type
All defined products	Read Write	IP Port address

2.3.3.4 KeepAliveStart

2.3.3.4.1 Description

Activates for TCP signalling transport the TCP-keep-alive functionality. This attribute defines the idle period in seconds before a keep alive message is sent.

2.3.3.4.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Seconds

2.3.3.5 KeepAliveInterval

2.3.3.5.1 Description

Activates for TCP signalling transport the TCP-keep-alive functionality. This attribute defines the interval in seconds between keep alive messages.

2.3.3.5.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Seconds

2.3.3.6 KeepAliveCount

2.3.3.6.1 Description

Activates for TCP signalling transport the TCP-keep-alive functionality. This attribute defines the consecutive number of keep alive failures that may occur before the link is reset.

2.3.3.6.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Integer

2.3.3.7 IPSocket

2.3.3.7.1 Description

This attribute references to an IP resource which contains the IP information of the VoIP Stack

2.3.3.7.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write	Resource identifier

2.3.3.8 FastStart

2.3.3.8.1 Description

This specifies if by default the H323 Fast Start procedures are initiated on all calls to and from this stack

2.3.3.8.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Boolean

2.3.3.9 H245 Tunnelling / H245Tunneling

2.3.3.9.1 Description

This specifies if by default the H323 H245Tunneling procedures are initiated on all calls to and from this stack.

2.3.3.9.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Boolean

2.3.3.10 H225MaintainConnection

2.3.3.10.1 Description

This specifies if by default the H323 H225 Maintain Connection procedures are initiated on all calls to and from this stack.

2.3.3.10.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Boolean

2.3.3.11 H225MultipleCalls

2.3.3.11.1 Description

This specifies if by default the H323 H225 Multiple Calls Connection procedures are initiated on all calls to and from this stack

2.3.3.11.2 Definition

Applicable	**Modes**	**User Type**
H323 VoIP Stack	Read Write	Boolean

2.3.3.12 H245FacilityPDU

2.3.3.12.1 Description

This specifies that when doing out signalling H245 procedures Facility messages are supported for all calls on this stack.

2.3.3.12.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Boolean

2.3.3.13 H225SendStartH245

2.3.3.13.1 Description

If set to True the start H245 will be sent in a separate Facility message and not included as default in the H225 message for all calls on this stack

2.3.3.13.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Boolean

2.3.3.14 T35CountryCode

2.3.3.14.1 Description

Allows for the configuration of the H225 vendor identifier's T35 country code value.

2.3.3.14.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Integer

2.3.3.15 T35Extension

2.3.3.15.1 Description
Allows for the configuration of the H225 vendor identifier's T35 extension value.

2.3.3.15.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Integer

2.3.3.16 ManufacturerCode
2.3.3.16.1 Description
Allows for the configuration of the H225 vendor identifier's T35 manufacturer code value.

2.3.3.16.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Integer

2.3.3.17 H245TCSREJ
2.3.3.17.1 Description
If set to True if a far end does not support TCS if the SVI receives a TCS REJ due to timeout the SVI will handle this as an ACK to allow the call to continue where possible.

2.3.3.17.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Boolean

2.3.3.18 H245StartPort
2.3.3.18.1 Description
When the call is using H245 procedures this defines the lowest IP port address used for any in call H245 listening connection.

2.3.3.18.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	IP Port Address

2.3.3.19 H245EndPort
2.3.3.19.1 Description
When the call is using H245 procedures this defines the highest IP port address used for any in call H245 listening connection.

2.3.3.19.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	IP Port Address

2.3.3.20 User=phone
2.3.3.20.1 Description
If set to True this will add to all outgoing SIP address fields after the address's port ";user=phone".

2.3.3.20.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write	Boolean

2.3.3.21 Sipversion

2.3.3.21.1 Description

This describes the version of SIP being supported. This show currently be set to 2.0.

2.3.3.21.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write	String

2.3.3.22 T1

2.3.3.22.1 Description

For SIP UDP transport retransmission this defines the T1 timer in milliseconds. The T1 timer specifies the initial retransmission timer on failure of receiving an acknowledgement to a method or PRACK.

2.3.3.22.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Milliseconds

2.3.3.23 T2

2.3.3.23.1 Description

For SIP UDP transport retransmission this defines the T2 timer in milliseconds. Each time a concurrent retransmission is made T1 is doubled. T2 defines the maximum time in milliseconds that T1 can be doubled to.

2.3.3.23.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Milliseconds

2.3.3.24 T4

2.3.3.24.1 Description

For SIP UDP transmission retransmission this defines the T4 timer in milliseconds. The T4 timer defines the period when the transaction is completed but still deemed to be active to capture any wayward retransmission to ensure that a retransmission is not interpreted as a new method.

2.3.3.24.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Milliseconds

2.3.3.25 D

2.3.3.25.1 Description

For SIP UDP transmission retransmission this defines the D timer in milliseconds. This timer defines the period from when an outgoing INVITE reject is received and the method is still deemed to be active to capture any wayward retransmission to ensure that a retransmission is not interpreted as a new method.

2.3.3.25.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Milliseconds

2.3.3.26 Tb

2.3.3.26.1 Description

For SIP UDP transmission retransmission this defines the Tb timer. The Tb timer specifies for outgoing INVITE methods the period of concurrent retransmissions before a method has deemed as failed due to non-response.

2.3.3.26.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Milliseconds

2.3.3.27 Tf

2.3.3.27.1 Description

For SIP UDP transmission retransmission this defines the Tf timer. The Tf timer specifies for outgoing non INVITE methods the period of concurrent retransmissions before a method has deemed as failed due to non-response.

2.3.3.27.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Milliseconds

2.3.3.28 Th

2.3.3.28.1 Description

For SIP UDP transmission retransmission this defines the Th timer in milliseconds. This timer defines the period from when an incoming INVITE method is received and the method is still deemed to be active to capture any wayward retransmission to ensure that a retransmission is not interpreted as a new method.

2.3.3.28.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Milliseconds

2.3.3.29 Tj

2.3.3.29.1 Description

For SIP UDP transmission retransmission this defines the Tj timer in milliseconds. This timer defines the period from when an incoming non INVITE method is received and the method is still deemed to be active to capture any wayward retransmission to ensure that a retransmission is not interpreted as a new method.

2.3.3.29.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Milliseconds

2.3.3.30 ForceSipExpiry

2.3.3.30.1 Description

If this attribute is set to True for any incoming registrations the incoming Register expiry timer will be overwritten with the SipExpiry value

2.3.3.30.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write	Boolean

2.3.3.31 SipExpiry

2.3.3.31.1 Description

This contains the value of the forced registration sip expiry timer

2.3.3.31.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write	Milliseconds

2.3.3.32 Proxy Media

2.3.3.32.1 Description

If set to True all media will pass through the SVI. For this to operate the SVI must have RTP Routers configured on the unit. If set to False the media will go direct between the two endpoints

2.3.3.32.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack with RTP Router installed	Read Write	Boolean

2.3.3.33 Values

Value	Description

2.3.3.34 Contact Header

2.3.3.34.1 Description

This allows you to configure the format of the outgoing contact header, with or without the CGPN as a prefix to the IP address.

2.3.3.34.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read	Drop Down List

	Write	

2.3.3.34.3 Values

Value	Description
No User	Does not include the user id (cgpn) in the Contact header in outgoing SIP messages
Include User	Includes the user id (cgpn) in the Contact header in outgoing SIP messages

2.3.3.35 RPID Header

2.3.3.35.1 Description

If the incoming destination is trusted (No setting of VoIP Destination flag "Non-Trusted-Domain" and the RPID header is set to True the SVI will attempt to interwork to the outgoing message the "P-Asserted-Identity" Header

2.3.3.35.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write	Boolean

2.3.3.36 Append CallID

2.3.3.36.1 Description

If set to True when a call with an internally generated SIP call-id leaves the SVI it will append the call-id with the specified "Append CallId" string. Any subsequent method responses will have the "Appended call-id" removed from the SIP call-id before being passed internally. This allows for direct call loop back on the SVI.

2.3.3.36.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	User string

2.3.3.37 ExpiryGrace

2.3.3.37.1 Description

This defined time allows for an extended timeout grace period to be added to the registration expiry timer for all incoming registrations. This allows for some internal tolerance on when a registration is deemed to have expired

2.3.3.37.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write	milliseconds

2.3.3.38 NAT Forwarding Address

2.3.3.38.1 Description

Override the media address in the outgoing SDP in a ISUP-SIP call with the specified address, for situations when the SVI is on the private side of a NAT firewall.

2.3.3.38.2 Definition

Applicable	Modes	User Type
	Read Write No-Default	IP address

2.3.3.38.3 Values

Value	Description

2.3.3.39 Options

2.3.3.39.1 Description

These bit mask options are used to setup identified behaviour of the VoIP Stack

2.3.3.39.2 Definition

Applicable	Modes	User Type
As defined	Read Write No-Default	Bit Mask Flag

2.3.3.39.3 Values

Value	Description
None	No flags are set
Ignore-image	If the SDP component contains a "m=image" line if this flag is set the SVI will ignore the contents and respond with the image line set to "m:image= 0"
Terminate-Updates	If this flag is set the SVI will not pass across an in call UPDATE methods and will internally ACKNOWELDGE back to the originator
Full-Contact-Header	For an incoming registering end point if this flag is set the outgoing METHODS request URI will contain the REGISTER contact details not the REGISTER destination
Use-P-Asserted-As_CGPN	For backward compatibility when both the P-Asserted and the RPID headers are present, with this flag set the calling party number is taken from the P-Asserted header and when not set the calling party is taken from the RPID header
Encrypt Proxy Parameters	The parameters added to the received 'Via' header, by the stateless proxy task, will be encrypted. Note that this means that the RFC3581 'received' and 'rport' parameters will not be added. (see section 2.3.3.63)
SNMP-Trap	If set True and SNMP Traps have been setup on the SVI, an SNMP trap will be sent on a state change of this resource

2.3.3.40 Transport

2.3.3.40.1 Description

If set to True this defines the type of transport used for all calls in and out of this VoIP Stack

2.3.3.40.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write	Drop Down List

2.3.3.40.3 Values

Value	Description
TCP	Use TCP as the SIP signalling transport
UDP	Use UDP as the SIP signalling transport

2.3.3.41 Max Forwards

2.3.3.41.1 Description

This attribute sets the maximum allowed value of an outgoing initiated method's max-forward attribute.

2.3.3.41.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write	Integer

2.3.3.42 ReInvite Renegotiate

2.3.3.42.1 Description

This flag allows for different handling of a RE-INVITE begin received without any SDP or with the SDP IP Address set to "0.0.0.0". If this is set to False this will be interpreted as a hold instruction. If set to True this will be interpreted as a off hold/update message.

2.3.3.42.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write	Boolean

2.3.3.43 ISUP Version

2.3.3.43.1 Description

This user string allows for the setting of the ISUP version field in the SIP-I MIME header

2.3.3.43.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack running SIP-I/SIP-T	Read Write No-Default	User string

2.3.3.44 ISUP Base Version

2.3.3.44.1 Description

This user string allows for the setting of the ISUP base version field in the SIP-I MIME header

2.3.3.44.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack running SIP-I/SIP-T	Read Write	User string

	No-Default	

2.3.3.44.3 Values

Value
Itu-t88
Itu-t92+
Ansi88
Ansi00
Etsi121
Etsi356
Gr217
Ttc87
Ttc93+

2.3.3.45 Retransmits

2.3.3.45.1 Description

This user records the number of retransmissions of SIP messages the VoIP Stack has needed to do.

2.3.3.45.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read	Integer

2.3.3.46 SVIRoute Priority

2.3.3.46.1 Description

This sets priority for selection of destination when SVIRoute header is processed to route the call. The value zero will set the priority off. Default value is 100.This is applicable only for the SVI-SBC product.

2.3.3.46.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Integer

2.3.3.47 TLS Cipher

2.3.3.47.1 Description

This sets the type of TLS Cipher set that can be supported. Default value is PERFORMANCE. This is applicable only for the SVI-SBC product.

2.3.3.47.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Drop Down List

2.3.3.47.3 Values

Value
NONE
PERFORMANCE
NORMAL
SECURE128
SECURE192

SECURE256
SUITEB128
SUITEB192
EXPORT

2.3.3.48 TLS Version

2.3.3.48.1 Description

This sets Version of TLS that should be used TLS 1.0, TLS 1.1 etc. Default value is TLS 1.0.This is applicable only for the SVI-SBC product.

2.3.3.48.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Drop Down List

2.3.3.48.3 Values

Value
VERS-TLS-ALL
VERS-SSL3.0
VERS-TLS1.0
VERS-TLS1.1
VERS-TLS1.2

2.3.3.49 TLS VerifyCert

2.3.3.49.1 Description

This sets flag to indicate whether we need to validate the certificate provided by TLS client when acting as a TLS server for any endpoint. Default value is off.This is applicable only for the SVI-SBC product.

2.3.3.49.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Boolean

2.3.3.50 TLS Handshake Timeout

2.3.3.50.1 Description

This sets TLS handshake timeout in milli seconds. This is applicable when we act as a TLS Client. Default value is 1 second. This is applicable only for the SVI-SBC product.

2.3.3.50.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	milliseconds

2.3.3.51 Rx Stats Register

2.3.3.51.1 Description

This creates a statistics count for SIP Registers

2.3.3.51.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Edit	

2.3.3.51.3 Rx Stats Register Resource

This creates a statistics count for SIP Registers

2.3.3.51.3.1 Retransmission Queue Maximum

2.3.3.51.3.1.1 Description

Number of Registers that can be queued

2.3.3.51.3.1.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Number of SIP-Registers that can be queued

2.3.3.51.3.2 TSample

2.3.3.51.3.2.1 Description

Period of which the registers are counted in milliseconds

2.3.3.51.3.2.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Milliseconds

2.3.3.51.3.3 Count Throttle

2.3.3.51.3.3.1 Description

The Number of registers that can be accepted with in the Tsample period

2.3.3.51.3.3.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Number of SIP-Registers

2.3.3.52 X509 CA File

2.3.3.52.1 Description

This sets the path to X509 Certificate Authority certificate file used for TLS X.509 certificate authentication. Default value is NULL. This is applicable only for the SVI-SBC product.

2.3.3.52.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write	User string

	No-Default	

2.3.3.53 X509 CRL File

2.3.3.53.1 Description

This sets the path to X509 Certificate Revocation List file used for TLS X.509 certificate authentication. Default value is NULL. This is applicable only for the SVI-SBC product.

2.3.3.53.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	User string

2.3.3.54 X509 Certificate

2.3.3.54.1 Description

This sets the path to X509 Certificate file used for TLS X.509 certificate authentication. Default value is NULL. This is applicable only for the SVI-SBC product.

2.3.3.54.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	User string

2.3.3.55 X509 Key

2.3.3.55.1 Description

This sets the path to X509 Private Key file used for TLS X.509 certificate authentication. Default value is NULL. This is applicable only for the SVI-SBC product.

2.3.3.55.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	User string

2.3.3.56 TLS Cert Handling

2.3.3.56.1 Description

This sets flag to indicate whether we need to request a certificate from the TLS client while acting as a TLS Server. Default value is Ignore. This is applicable only for the SVI-SBC product.

2.3.3.56.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Drop Down List

2.3.3.56.3 Values

Value
Ignore
Request
Require

2.3.3.57 TLS Max Chain Depth

2.3.3.57.1 Description

This sets depth to which we need to validate a X.509 certificate if the other end provides a certificate chain. Default value is 5. This is applicable only for the SVI-SBC product.

2.3.3.57.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Integer

2.3.3.58 TLS Max Cert Bits

2.3.3.58.1 Description

This sets max allowed bits in a X.509 certificate provided by the other end. Default is 8200. This is applicable only for the SVI-SBC product.

2.3.3.58.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Integer

2.3.3.59 SRP Password

2.3.3.59.1 Description

This sets the password used for SRP authentication. Default value is NULL. This attribute is reserved for future use. This is applicable only for the SVI-SBC product.

2.3.3.59.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	User string

2.3.3.60 SRP Password File

2.3.3.60.1 Description

This sets the path to password decoding file used for SRP authentication. Default value is NULL. This attribute is reserved for future use. This is applicable only for the SVI-SBC product.

2.3.3.60.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	User string

2.3.3.61 PSK Password File

2.3.3.61.1 Description

This sets the path to password decoding file used for PSK authentication. Default value is NULL. This attribute is reserved for future use. This is applicable only for the SVI-SBC product.

2.3.3.61.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	User string

2.3.3.62 URI Type

2.3.3.62.1 Description

This sets the type of uri needs to be used for TLS calls going out through this VoIP Stack. Default value is sip. This is applicable only for the SVI-SBC product.

2.3.3.62.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Drop Down List

2.3.3.62.3 Values

Value
Sip
Sips

2.3.3.63 Stateless Proxy Methods

2.3.3.63.1 Description

Adding a SIP Method to this list enables the Stateless Proxy functionality for the method, and for responses that cannot be matched to a SIP transaction.

See also – ‘Encrypt Proxy Parameters’ Voip Stack option (section 2.3.3.39)

2.3.3.63.2 Definition

Applicable	Modes	User Type
SIP VoIP Stack	Read Write No-Default	Drop Down List

2.3.3.63.3 Values

MESSAGE – only the SIP ‘MESSAGE’ method is currently supported
All Methods

2.4 VoIP Destination

2.4.1 Description

A VoIP destination defines a VoIP end point which is being communicated to by a single IP address.

2.4.2 Products

SVI-MG | SVI-MGC | SVI-C4 | SVI-C5 | SVI-SBC

2.4.3 Attributes

2.4.3.1 Type

2.4.3.1.1 Description

This describes the type of VoIP Destination.

2.4.3.1.2 Definition

Applicable	Modes	User Type
As defined	Read Write	Drop Down List

2.4.3.1.3 Values

Value	Description
SIP Endpoint	This is a static SIP endpoint
Default SIP Endpoint	A special VoIP Destination which is used as a template when creating dynamic VoIP Destinations
Registered SIP Endpoints	This is a registering SIP endpoint
H323 Endpoint	This is a H323 endpoint
H323 Gatekeeper	This is a H323 Gatekeeper

2.4.3.2 Name / Address

2.4.3.2.1 Description

This contains the static IP address of a VoIP Destination

2.4.3.2.2 Definition

Applicable	Modes	User Type
All	Read Write No-Default	IP Address

2.4.3.3 Proxy Media

2.4.3.3.1 Description

If set to True this specifies that the media from this end point need will go through the SVI because of cross border/NAT/topology hiding requirements

2.4.3.3.2 Definition

Applicable	Modes	User Type
All, RTP Router required	Read Write	Boolean

2.4.3.4 Behind NAT

2.4.3.4.1 Description

If set to True this defines that the end point is behind a firewall and will require NAT traversal functionality

2.4.3.4.2 Definition

Applicable	Modes	User Type
SIP VoIP Destinations, requires RTP Router	Read Write	Boolean

2.4.3.5 Username

2.4.3.5.1 Description

For registering endpoints this contains the end points username

2.4.3.5.2 Definition

Applicable	Modes	User Type
All	Read Write No-Default	User string

2.4.3.6 Password

2.4.3.6.1 Description

For registering end points this contains the end point password

2.4.3.6.2 Definition

Applicable	Modes	User Type
All	Read Write No-Default	User string

2.4.3.7 Contact

2.4.3.7.1 Description

This variable stores the registered endpoints contact information from the incoming REGISTER method

2.4.3.7.2 Definition

Applicable	Modes	User Type
SIP VoIP Destination	Read	Read only

2.4.3.8 CallControl

2.4.3.8.1 Description

This resource pointer contains the identity of the Call Control resource applicable to this VoIP destination. The Call Control resource contains additional call related options.

2.4.3.8.2 Definition

Applicable	Modes	User Type
All	Read Write	Resource Identifier

2.4.3.9 Transport

2.4.3.9.1 Description

This defines the SIP signalling transport used.

2.4.3.9.2 Definition

Applicable	Modes	User Type
SIP VoIP destinations	Read	Drop Down List

	Write	

2.4.3.9.3 Values

Value	Description
TCP	Use TCP as signalling transport
UDP	Use UDP as signalling transport

2.4.3.10 Port

2.4.3.10.1 Description

This defines the End Points IP address port which is used for SIP signalling. The default value of this for SIP is 5060 and for H323 is 1720.

2.4.3.10.2 Definition

Applicable	Modes	User Type
All	Read Write	Integer

2.4.3.11 Stack

2.4.3.11.1 Description

This resource identifier contains the identity of an already configured VoIP Stack that the endpoint will be connecting to.

2.4.3.11.2 Definition

Applicable	Modes	User Type
All	Read Write	Resource Identifier

2.4.3.12 FastStart

2.4.3.12.1 Description

This specifies if by default the H323 Fast Start procedures are initiated on all calls to and from this VoIP Destination

2.4.3.12.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Boolean

2.4.3.13 H245 Tunnelling / H245Tunneling

2.4.3.13.1 Description

This specifies if by default the H323 H245Tunneling procedures are initiated on all calls to and from this VoIP Destination

2.4.3.13.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Boolean

2.4.3.14 H225MaintainConnection

2.4.3.14.1 Description

This specifies if by default the H323 H225 Maintain Connection procedures are initiated on all calls to and from this VoIP Destination

2.4.3.14.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Boolean

2.4.3.15 H225MultipleCalls

2.4.3.15.1 Description

This specifies if by default the H323 H225 Multiple Calls Connection procedures are initiated on all calls to and from this VoIP Destination

2.4.3.15.2 Definition

Applicable	**Modes**	**User Type**
H323 VoIP Stack	Read Write	Boolean

2.4.3.15.3 Values

Value	Description

2.4.3.16 H245FacilityPDU

2.4.3.16.1 Description

This specifies that when doing out signalling H245 procedures Facility messages are supported for all calls on this VoIP Destination.

2.4.3.16.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Boolean

2.4.3.17 H225SendStartH245

2.4.3.17.1 Description

If set to True the start H245 will be sent in a separate Facility message and not included as default in the H225 message for all calls on this VoIP Destination

2.4.3.17.2 Definition

Applicable	Modes	User Type
H323 VoIP Stack	Read Write	Boolean

2.4.3.18 Progress Indicator

This attribute has come to its End of Life (EOL). Whilst it still appears as an attribute it has no impact on the system and will shortly be removed.

2.4.3.19 SIP CLIR

This attribute has come to its End of Life (EOL). Whilst it still appears as an attribute it has no impact on the system and will shortly be removed.

2.4.3.20 Last Registered

2.4.3.20.1 Description

For a SIP registering end point this contains the time a registering end point successfully registered with the SVI.

2.4.3.20.2 Definition

Applicable	Modes	User Type
SIP end Point	Read	Time

2.4.3.21 Registration Expires

2.4.3.21.1 Description

For SIP registering end point this contains the time the current registration will expire

2.4.3.21.2 Definition

Applicable	Modes	User Type
SIP end Point	Read	Time

2.4.3.22 Diversion Type

This attribute has come to its End of Life (EOL). Whilst it still appears as an attribute it has no impact on the system and will shortly be removed.

2.4.3.23 UpdateH323Caps

2.4.3.23.1 Description

If this flag is set to True when a H323 backward message is received with updated H323 Capabilities, the internal H323 updated capabilities will be updated.

2.4.3.23.2 Definition

Applicable	Modes	User Type
H323 end point	Read Write	Boolean

2.4.3.24 Adjaddr

2.4.3.24.1 Description

For SIP calls that are behind NAT, this stores the adjacent IP address that this VoIP destination can be contacted on.

2.4.3.24.2 Definition

Applicable	Modes	User Type
SIP VoIP Destinations behind NAT	Read	Read-only

2.4.3.25 Adjport

2.4.3.25.1 Description

For SIP calls that are behind NAT, this stores the adjacent IP port address that this VoIP destination can be contacted on.

2.4.3.25.2 Definition

Applicable	Modes	User Type
SIP VoIP Destinations behind NAT	Read	Read-only

2.4.3.26 RAS IP

2.4.3.26.1 Description

If the end point is a H323 VoIP gatekeeper this resource identifier will reference the IP resource containing the Gatekeeper IP information

2.4.3.26.2 Definition

Applicable	Modes	User Type
H323 Gatekeeper VoIP Destination	Read Write	Resource Reference

2.4.3.27 AutoDiscovery

2.4.3.27.1 Description

For a H323 VoIP gatekeeper this defines how the gatekeeper is discovered

2.4.3.27.2 Definition

Applicable	Modes	User Type
H323 Gatekeeper VoIP destination	Read Write	Drop Down List

2.4.3.27.3 Values

Value	Description
Auto	Discovers Gatekeeper by Broadcast of GRQ
GRQ	Discovers Gatekeeper by transmission of GRQ to address setup in VoIP Destination
No GRQ	Gatekeeper is not discovered but uses address setup in VoIP destination to directly register with

2.4.3.28 TimeToLive

2.4.3.28.1 Description

For H323 gatekeeper this defines the time to live of the Gatekeeper registration

For outbound SIP registers this defines the timeToLive for outgoing registration

For inbound SIP registration this stores the received timeToLive

2.4.3.28.2 Definition

Applicable	Modes	User Type
H323 Gatekeeper SIP Registering End Point	Read Write No-Default	Milliseconds

2.4.3.29 Protocol

2.4.3.29.1 Description

For SIP this defines the SIP protocol being used to this end point

2.4.3.29.2 Definition

Applicable	Modes	User Type
SIP End points	Read Write No-Default	Drop Down List

2.4.3.29.3 Values

Value	Description
Standard	SIP conforming to RFC 3261
SIPI	SIP-I conforming to ITU 1912.5
SIPT	SIP-T conforming to RFC 3372
X-SIPI	Proprietary SIPI implementation

2.4.3.30 IP Low

2.4.3.30.1 Description

When used in conjunction with IP High this enables a range of IP address to be assigned to the VoIP destination from which incoming calls may be received. All outgoing calls will still be made to the defined VoIP Destination address.

2.4.3.30.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	IP Address

2.4.3.31 IP High

2.4.3.31.1 Description

When used in conjunction with IP Low this enables a range of IP address to be assigned to the VoIP destination from which incoming calls may be received. All outgoing calls will still be made to the defined VoIP Destination address.

2.4.3.31.2 Definition

Applicable	Modes	User Type
SIP endpoints	Read Write No-Default	IP Address

2.4.3.32 Toptions

2.4.3.32.1 Description

If Toptions is set to a value in seconds which is greater than zero, when a SIP call is in the answer state a heartbeat OPTIONS method is transmitted every Toptions second. On none response to the OPTIONS method the call will be cleared.

2.4.3.32.2 Definition

Applicable	Modes	User Type
SIP end point	Read Write No-Default	seconds

2.4.3.33 Tocoptions

2.4.3.33.1 Description

To heartbeat an end point to ensure that is available an out of call OPTIONS method can be sent every Tocoptions second. If no response is received then the VoIP Destination will be taken out of service and will only come back into service once an out of call OPTIONS method is received.

2.4.3.33.2 Definition

Applicable	Modes	User Type

SIP end Point	Read Write No-Default	Seconds

2.4.3.34 Options

2.4.3.34.1 Description

These bit mask options are used to setup identified behaviour of the VoIP Destination

2.4.3.34.2 Definition

Applicable	Modes	User Type
As defined	Read Write No-Default	Bit Mask Flag

2.4.3.34.3 Values

Value	Description
None	
IP_EXCHANGE	EOL
Auto_CLIR	while interworking from TDM to SIP(ISUP-SIP,Q931-SIP,IUP-SIP etc),this option will enable to send clir according to the presentation indicator coming from the TDM side. Eg:if presentation is restricted, then anonymous should go as cgpn.
Auto-International	Automatic international form conversion for TDM(ISUP,Q931,IUP etc) to SIP and SIP-TDM calls. If the call is going for SIP to TDM, then the number (cdpn, cgpn etc) should not contain a "+".If this option is enabled, then the leading "+" from the number (cgpn, cdpn etc) is removed and the nature of address is changed to international. Similarly in case of calls from TDM to SIP, a "+" is added to the number if the nature of address indicator is international.
Rad_AuthReg	EOL
Rad_AuthCall	This option will enable radius authentication for SIP Calls(INVITE). Radius resource must be configured properly and added to routing criteria in order for this to work.
IC-Registration	This will enable SVI to accept registration from an endpoint and allow it to make calls. Username and password needs to be configured for this to work.
OG-Registration	This will enable SVI to register as an endpoint to another SIP server and make calls through it. IP, Port, username and password needs to be configured for this feature to work.
OG-AA-Registration	This option will enable radius authentication for SIP registration (REGISTER). Radius resource must be configured properly and added to routing criteria in order for this to work.
Release-Header	This will enable SVI to add a Reason header in the BYE, CANCEL or 4XX-6XX responses send out from SVI for ISUP-SIP calls. The format of the Reason header will be as below Reason: Q.850 ;cause=16 ;text="Normal Clearing"
Use-Contact	This will use the saved contact in the VoIP destination as username while creating Contact header for outgoing SIP messages.

SS7Route-Header	This option will add a Custom header into the outgoing INVITE for ISUP-SIP calls. The header format is as below SS7Route: cic=2;opc=1000;dpc=2000;ni=2;billing-ani=44123456789 Note: The billing-ani attribute will be present only when the ISUP variant is UK ISUP.
Force-Privacy-Rules	This will enable to pass on the incoming number from the SIP “From:” header as calling party number in the outgoing “From:” header for SIP to SIP calls. This option is of use when the incoming INVITE of the SIP-SIP call comes in with “P-Asserted-Id” or “P-Preferred-Identity” or “Remote-Party-ID” or a combination of these three headers. If the option is not enabled, then the number coming in the mentioned headers will go in the “From:” header of outgoing SIP INVITE.
Force-Contact-Match	This will make the voip destination check in SVI strict for an incoming SIP call. Only if the incoming contact match the contact saved in the voip destination, the call will go through.
G729-Default-No-AnnexB	If G729 is received from UA without G729 fmtp default as AnnexB=no.
G729-No-AnnexB	If flag is set and G729 with annex b is received respond with Annexb=no.
PRACK-Supported	This will add a “Supported: 100rel” header into the outgoing INVITE from SVI for calls going to the voip destination.At the same time,this will send “Require: 100rel, RSeq: 8002” etc in 18X responses going from SVI in case of a call coming in from the voip destination provided the voip destination has sent “Supported: 100rel” in the INVITE coming to SVI. Used to force reliable provisional responses.
Fax-Use-G711	This will enable SVI to use the G711 to send fax in case of SIP-ISUP fax calls rather than using t38.The first fax re-invite itself will contain the G711 codec.This can be used in case of endpoints which do not support fax fallback from t38 to G711.The flavor of G711 used will depend on 1.The G711 flavour received in the initial INVITE from the sip endpoint. 2.If there are no G711 codecs present in initial INVITE,then “FAX FALLBACK PCMA” attribute of voip destination will decide the G711 flavour. If on G711A will be used,otherwise G711U will be used.
Do-Not-Use-Contact-Header	If there is a contact configured in the voip destination resource or a contact got populated by SVI in case of a registering voip destination,this option will prevent SVI from using the saved contact in the outgoing INVITE request going towards that endpoint.
SNMP-Trap	This will enable SVI to send SNMP traps to to IP/Port configured in the debug resource,in case of a state change.(trap send upon resource going out of service)
Full-Anonymous	If calling party number is “anonymous” for an outgoing SIP call, this feature will make the “From” header URI of outgoing INVITE/REGISTER in the format anonymous@anonymous.invalid instead of anonymous@SVI-IP:Port
PRACK-Required	This will add a “Require: 100rel” header into the outgoing

	INVITE from SVI for calls going to the voip destination. At the same time,this will send “Require: 100rel, RSeq: 8002” etc in 18X responses going from SVI in case of a call coming in from the voip destination provided the voip destination has sent “Supported: 100rel” in the INVITE coming to SVI.Used to force reliable provisional responses.
Always-Ringback	Ringback will be played to **outgoing calls to this VoIP Destination**, once the call is ringing (SIP '180') It will not be turned off, even if remote media details are received in SDP, until the call is answered. This setting will override any other ringback setting on other resources.
No-Ringback	Ringback will be not played to **outgoing calls to this VoIP Destination.** This setting will override any other ringback setting on other resources.
No-Ringback-If-SDP	Ringback will be played to **outgoing calls to this VoIP Destination**, once the call is ringing (SIP '180'). It will be turned off if remote media details are received in SDP, or when the call is answered. This setting will override any other ringback setting on other resources.
No-Number-Display	This option will prevent the caller id from being sent as display also for ISUP-SIP interworking. By default for ISUP to SIP calls, the Remote-Party-ID, P-Asserted-Id etc will have the number itself as display. For SIP-SIP calls, this can prevent the number display from going only in Remote-Party-ID header. Eg: Without option P-Asserted-Id: “44123456” <44123456@192.168.1.1:5060> With option P-Asserted-Id: <44123456@192.168.1.1:5060>
Microsoft_OCS	Microsoft OCS sends INVITE without SDP to indicate call hold. This option will enable SVI to detect INVITE without SDP as hold and send back response accordingly.
Custom Responses	This will enable SVI to pass through unsupported 4XX,5XX and 6XX cause codes as is with a text string ”Unknown” after the cause code. If not configured, all unsupported codes will generate 406 from SVI.
SIP_REGISTER_CHALLENGE_NO_QOP	While sending a SIP register 401 challenge, this option will prevent “qop” parameter from going in the “WWW-Authenticate:” header of 401 response. This has been added to interwork with a certain type of Mitel PBX which does not support qop parameter.
Contact-as-URI	uses the incoming contact header to be used as the request URI for in calls.
P-Preferred-Transparent	This will enable SVI to proxy across P-Preferred-Identity header in case of SIP-SIP calls.
Non-Trusted-Domain	This will prevent SVI from sending any kind of Privacy headers(P-Preferred-Identity/P-Asserted-Identity/Remote-Party-ID) to the far end if “Privacy:id” is received in incoming SIP invite. This is applicable only for SIP-SIP calls.
Pass-Through-CGPN-With-Privacy	This will pass the actual Calling ID to the far end, even if the Privacy: header contains id. With this option disabled the

	Calling ID will be passed in the From: header as 'anonymous'
Delayed-FS-TCS-Negotiation	This will make TCS negotiation for H323-SIP interworking without proxymedia happen after connect + OpenLogicalChannel.

2.4.3.35 Registration State

2.4.3.35.1 Description

Internally used variable to store the current status of a registering VoIP endpoint

2.4.3.35.2 Definition

Applicable	Modes	User Type
SIP Registering Endpoints	Read	Read only

2.4.3.35.3 Values

Value	Description
None	VoIP destination has not yet registered
Unregistered	VoIP destination is unregistered
Internal Registered	VoIP destination is registered directly on the unit through the VoIP destination
External Registered	VoIP destination is registered through an external registering agent

2.4.3.36 G729VAD

2.4.3.36.1 Description

For VoIP to TDM traffic this defines if G729 VAD is applied to the G729 codec

2.4.3.36.2 Definition

Applicable	Modes	User Type
VoIP to TDM	Read Write No-Default	Boolean

2.4.3.37 FAX FALLBACK PCMA

2.4.3.37.1 Description

For VoIP to TDM traffic if this flag is set to True and T.38 negotiation with the end point fails, the call will fall back to G711 transmission.

2.4.3.37.2 Definition

Applicable	Modes	User Type
VoIP to TDM	Read Write No-Default	Boolean

2.4.3.38 Host Name

2.4.3.38.1 Description

This allows for a user defined string to be used instead of the IP address for all outgoing URI method requests.

2.4.3.38.2 Definition

Applicable	Modes	User Type

SIP endpoints	Read Write No-Default	User string

2.4.3.39 Supported Media

2.4.3.39.1 Description

This defines the media supported for a particular endpoint.

2.4.3.39.2 Definition

Applicable	Modes	User Type
	Read Write No-Default	

2.4.3.39.3 Values

Value	Description
Audio	The endpoint supports audio only
Image	The end point supports video only
All	The end point supports both video and audio

2.4.3.40 Privacy

2.4.3.40.1 Description

This option allows for what the value of the privacy ID should be set to when a calling party number is withheld.

2.4.3.40.2 Definition

Applicable	Modes	User Type
SIP endpoint	Read Write No-Default	Drop Down List

2.4.3.40.3 Values

Value
Id
Header
Session
User
Critical

2.4.3.41 Allow

2.4.3.41.1 Description

This bit mask field allows for the user to define the Allowed methods to be identified to the end point via the "Allow" SIP header

2.4.3.41.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	Bit mask flag

2.4.3.41.3 Values

Value
INVITE
CANCEL
BYE
ACK
OPTIONS
INFO
SUBSCRIBE
REGISTER
NOTIFY
PRACK
REFER
UPDATE
MESSAGE

2.4.3.42 RTP options

2.4.3.42.1 Description

This bit mask field allows for the user to define various features which control how the media data is passed across the Squire RTP router. It can also enable the collection of call related QOS statistics. The Secure-RTP options enables the SRTP feature for the SIP endpoint. This is applicable only for the SVI-SBC product.

2.4.3.42.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	Bit mask flag

2.4.3.42.3 Values

Value
Secure-Address
Secure-PT
Report-Stats
Report-Jitter
Secure-VoIP
Jitter-Settled
Secure-RTP

2.4.3.43 TLS Authentication

2.4.3.43.1 Description

This bit mask field allows for the user to define the type of TLS authentications supported by the VoIP Destination. Default value is X.509. Other values are for future use. This is applicable only for the SVI-SBC product.

2.4.3.43.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write	Bit mask flag

	No-Default	

2.4.3.43.3 Values

Value
X.509
SRP
PSK
Anonymous

2.4.3.44 X509 CA Cert File

2.4.3.44.1 Description

This sets the path to X509 Certificate Authority certificate file used for TLS X.509 certificate authentication. Default value is NULL. This is applicable only for the SVI-SBC product.

2.4.3.44.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	User string

2.4.3.45 X509 CRL File

2.4.3.45.1 Description

This sets the path to X509 Certificate Revocation List file used for TLS X.509 certificate authentication. Default value is NULL. This is applicable only for the SVI-SBC product.

2.4.3.45.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	User string

2.4.3.46 X509 Certificate File

2.4.3.46.1 Description

This sets the path to X509 Certificate file used for TLS X.509 certificate authentication. Default value is NULL. This is applicable only for the SVI-SBC product.

2.4.3.46.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	User string

2.4.3.47 X509 Private Key File

2.4.3.47.1 Description

This sets the path to X509 Private Key file used for TLS X.509 certificate authentication. Default value is NULL. This is applicable only for the SVI-SBC product.

2.4.3.47.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	User string

2.4.3.48 SRP Username

2.4.3.48.1 Description

This sets the username used for TLS SRP authentication. Default value is NULL. This is reserved for future use. This is applicable only for the SVI-SBC product.

2.4.3.48.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	User string

2.4.3.49 SRP Password

2.4.3.49.1 Description

This sets the password used for TLS SRP authentication. Default value is NULL. This is reserved for future use. This is applicable only for the SVI-SBC product.

2.4.3.49.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	User string

2.4.3.50 PSK Username

2.4.3.50.1 Description

This sets the username used for TLS PSK authentication. Default value is NULL. This is reserved for future use. This is applicable only for the SVI-SBC product.

2.4.3.50.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	User string

2.4.3.51 PSK Key

2.4.3.51.1 Description

This sets the key used for TLS PSK authentication. Default value is NULL. This is reserved for future use. This is applicable only for the SVI-SBC product.

2.4.3.51.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	User string

2.4.3.52 Algorithm

2.4.3.52.1 Description

This drop down field allows for the user to define the type of TLS key exchange algorithm supported by the VoIP Destination. Default value is none. This attribute is for future use. This is applicable only for the SVI-SBC product.

2.4.3.52.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	Drop Down List

2.4.3.52.3 Values

Value
RSA
DSA

2.4.3.53 Algorithm Options

2.4.3.53.1 Description

This drop down field allows for the user to define the type of TLS key exchange algorithm options supported by the VoIP Destination. Default value is none. This attribute is for future use. This is applicable only for the SVI-SBC product.

2.4.3.53.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	Drop Down List

2.4.3.53.3 Values

Value
Diffie-Helman
Elliptical-Curve

2.4.3.54 TLS HostName

2.4.3.54.1 Description

This sets CN parameter in the X.509 certificate sent by server during a TLS authentication. This will be matched for certificate validation. Default value is NULL. This is applicable only for the SVI-SBC product.

2.4.3.54.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	User string

2.4.3.55 TLS VerifyCert

2.4.3.55.1 Description

This sets flag to indicate whether we need to validate the certificate provided by TLS server when acting as a TLS client towards this endpoint. Default value is on. This is applicable only for the SVI-SBC product.

2.4.3.55.2 Definition

Applicable	Modes	User Type
SIP Endpoints	Read Write No-Default	Boolean