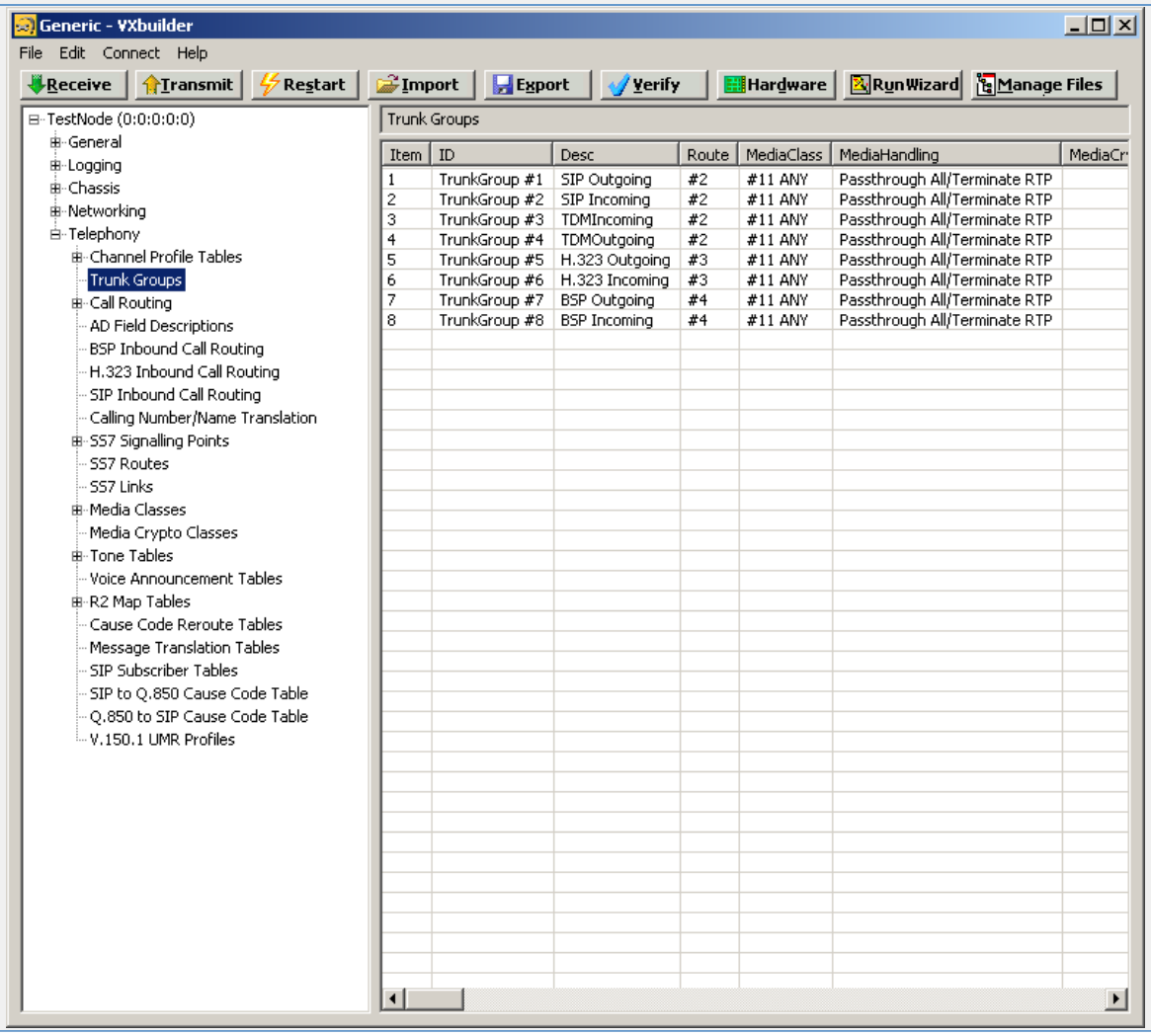
Managing Trunk Groups

Trunk group types are derived from the port on which the trunk group trunks being configured.

View Trunk Groups	At the directory tree, select **Telephony** > **Trunk Groups** to view the **Trunk Groups** screen.
Add Trunk Groups	Insert a new line in the **Trunk Groups** screen, then double-click on the new line to present the **Edit Trunk Groups** dialog box.
Modify Trunk Groups	Either double-click on a line in the Trunk Groups table, or highlight a line and use the **Edit...** popup menu option to access the **Edit Trunk Groups** dialog box with tabs.

Trunk Groups Directory

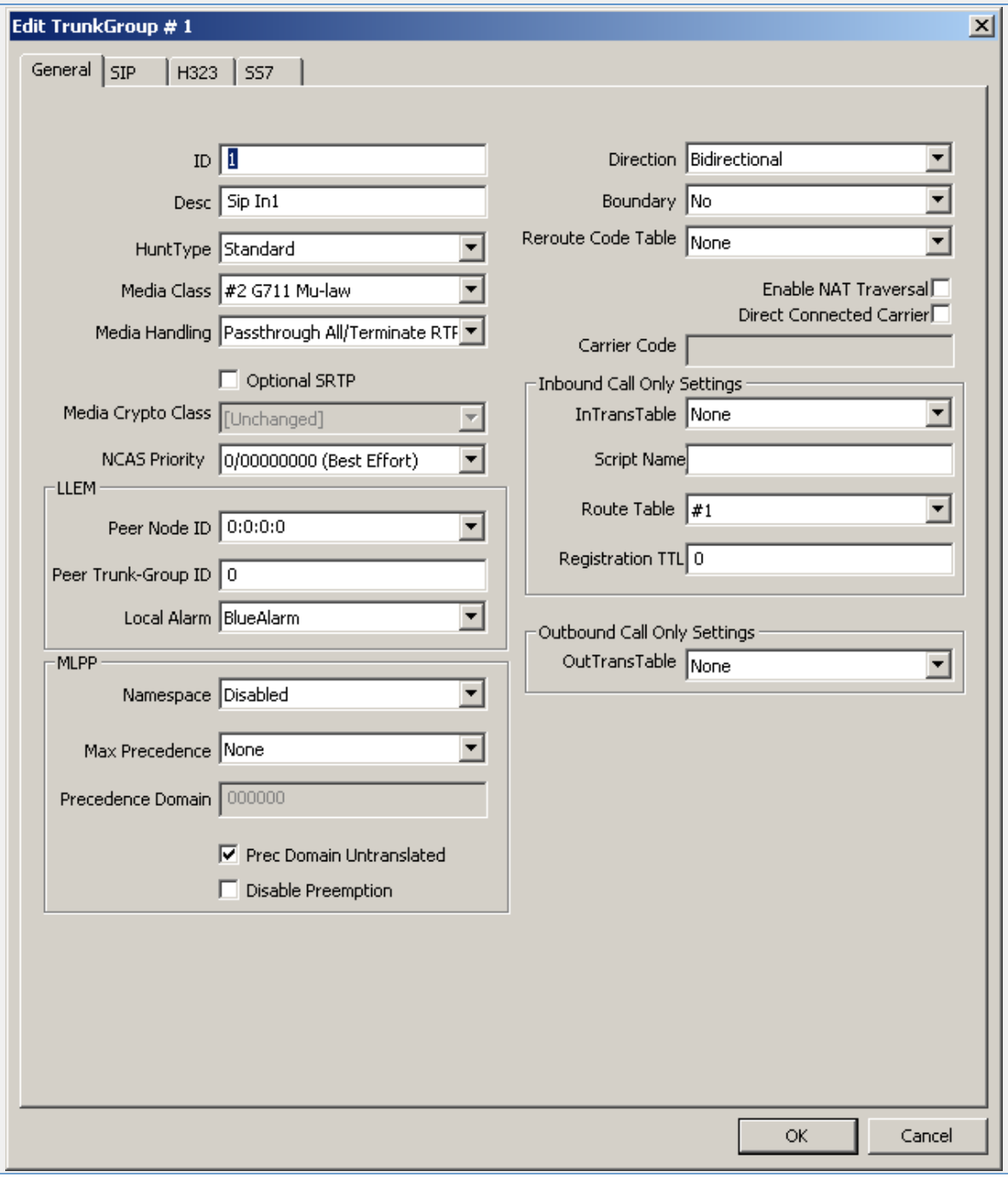


Edit Trunk Groups Controls

- General Tab
- SIP Tab
- H.323 Tab
- SS7 Tab

General Tab

Edit Trunk Groups Dialog General Tab



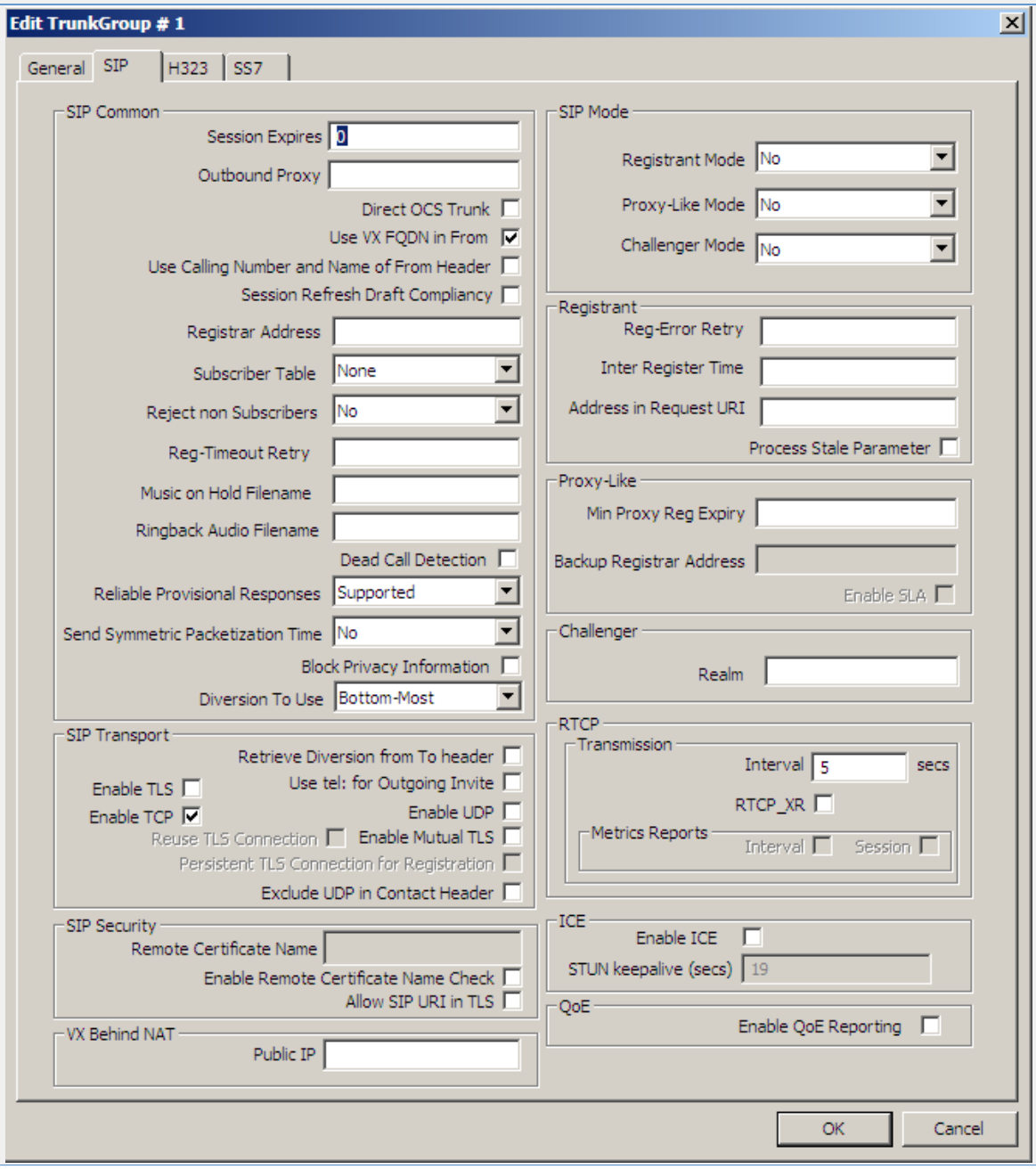
Field	Description
ID	The number of the trunk group in the range 1 - 10000. This is the number used in the routing and the channel assignments. The numbers do not have to be consecutive.
Description	An optional description text field.

Hunt Type	The hunting type to use for outgoing calls on this trunk. Option include: • Standard--Standard hunting, this hunting type starts with the first available channel and works up the channel list. Each new call will use the lowest available channel in the trunk group. • Uniform--this hunting type uses the next available channel in the hunt group. It will go though the entire trunk group before going back the first channel. • Least Idle--this hunt type will use the last channel to clear to place the next call to. This hunting type is not recommended for CAS circuits as channels are seized repeatedly. • Most Idle: This hunt type will use a channel the has not been used for the longest time for the next call. • One to One--in this hunt type the index of the outbound channel within the outbound trunk group will map the index of the channel from the inbound trunk group. The type can be used for predictable channel selection. • Match Own Number: A trunk group configured with this Hunt Type can only be used with CAS Tunnel and CAS MRD type ports.
Media Class	Select a media class from the drop-down list. The media class selection determines what codecs are allowed for this call. It also determines the preference order for codec selection. Note that codec selection applies to all calls, even if no codec would ultimately be involved (such as a TDM-TDM call). The media class is selected on the call route, the inbound trunk group, and outbound trunk group for this call. The allowed codecs are the intersection between all three of these. • Call route has the highest dominance for codec preference order • Outbound trunk group has the second highest • Inbound trunkgroup has the least
Optional SRTP	When configuring SRTP on trunkgroups select Pass-through or Termination. When Pass-through is selected you must choose a passthrough mode. When Termination is selected, you can select a Media-Crypto class from the list of algorithms.
Passthrough Mode	When Pass-through is the selected media handling, you can select Secure-only to allow only secure calls or Secure & Unsecure to allow both secure and unsecure calls.
Media Crypto Class	Select a media crypto class from the list of algorithms when Terminate is selected as the media handling type.
NCAS Priority	Select the Diffserv value for non-call signaling traffic.
LLEM	
Peer Node ID	The ID of the peer node for the LLEM configuration.
Peer Trunk-group ID	The trunk group to be associated with this LLEM configuration.
Local Alarm	Select the alarm type from the dropdown dialog box.
MLPP	
Namespace	Specifies the type of namespace as: Disabled, Transparent, Untranslated, DSN, DRSN, or Q.735.
Max Precedence	Specifies precedence options as: Flash-OVerride-Override, Flash-Override, Flash, Immediate, Priority, Routine, or None.
Precedence Domain	A hexadecimal string (six digits) that identifies the precedence domain for a trunk group. A value of 000000 indicates no domain is received. A precedence domain identifies the group of calls that can be ordered by level of importance. Calls from on precedence domain cannot override calls from a different precedence domain.
Prec Domain Untranslated	Select the checkbox to specify that the precedence domain is to be used as it is received.
Disable Preemption	Enables/disables Preemption.

Direction	Options include: • Bidirectional: Allow calls to go in and out of the trunk • Inbound: Allow the trunk to receive calls only • Outbound: Allow the trunk to originate calls only
Boundary	Option include Yes to enable boundary and No to disable boundary ⚠ Set this option to Yes when the node is the last node in the call path. This option is used primarily for billing purposes to indicate that this node is the last node in the billing network.
Reroute Code Table	Selects the Cause Code Reroute Table used on this trunk group for determining when to reroute Without this table, no call failures will be rerouted. See the topic Cause Code Reroute Table for specific details.
Enable NAT Traversal	Enables functionality that allows SIP or H.323 to better handle calls through Network Address Translation (NAT).
Direct Connected Carrier	Enables a direct connected carrier.
Carrier Code	When a direct connected carrier is enabled enter the carrier code.
Inbound Call Only Settings	
InTransTable	Specifies the calling number/name translation table to be used by inbound calls
Script Name	The voice scrept that will be invokded when a call is received on this channel--voice bearer only.
Route Table	Routing table which is associated with this trunk group.
Registration TTL	This field is the amount of time suggested by Unified Messaging Server (UMS) to keep the subscription alive.
Outbound Call Only Settings	
OutTransTable	Specifies the calling number/name translation table to be used on outbound calls The node will refresh subscriptions before this time expires. This field also applies to registrant mode. In registrant mode, the node will refresh registration within the time specified in this field.

SIP Tab

Edit Trunk Groups Dialog SIP Tab



Field	Description
SIP Common	
Session Expires	Number of seconds after which a reINVITE or UPDATE message should be sent to exchange keep alive information during a SIP call.
Outbound Proxy	Names the outbound proxy.
Direct OCS Trunk	Check the box to enable that a 183 with SDP is sent immediately after 100 trying so that media negotiation can start.
Use VS FQDN in From	Check the box to enable that the Ethernet FQDN setting or DNS-Suffix field is used to construct the domain portion of the From header.

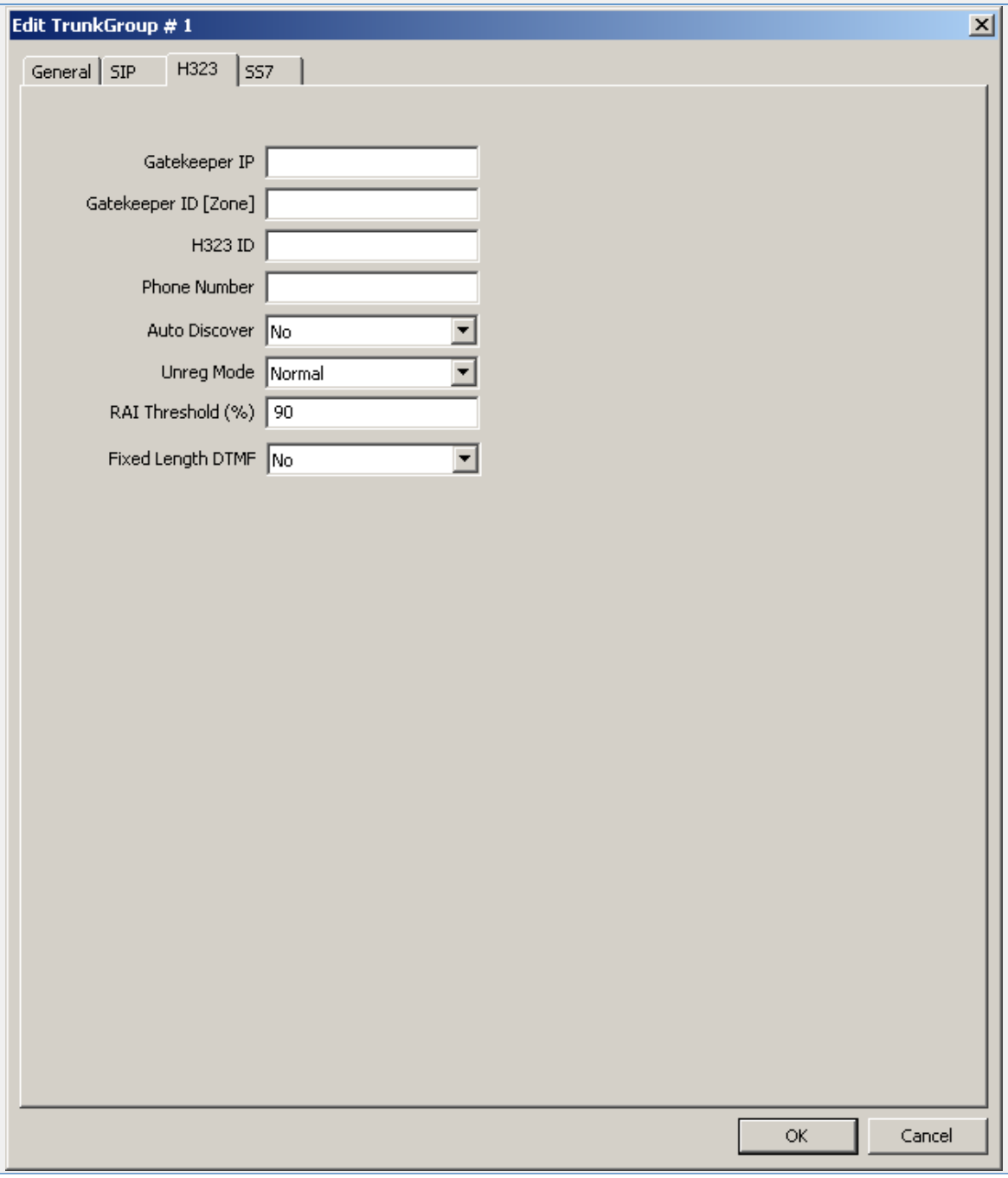
Use Calling Number and Name of From Header	Check the box to enable that the From header is used for the Calling Number and Name in the Outgoing Invite.
Session Refresh Draft Compliancy	Check the box to enable that the refresh Timeout is set as per draft-ietf-sip-session-timer-08 (or set as 15 when VX is not a refresher).
Registrar Address	Either an FQDN or IP address of the SIP Registrar to which the REGISTER requests should be sent This address is used when Subscriber tables are used or if VX is acting in Proxy-like Mode. This setting is the address to which all the SIP REGISTER request will be sent for all the subscribers in the subscription table. The IP address of the Registrar is resolved using a Domain Name Server.
Subscriber Table	The name of the table of subscribers.
Reject non Subscribers	This parameter applies to the incoming SIP calls. When enabled the calls whose called numbers cannot be found in the Subscriber table will be rejected. (Default: disabled)
Reg-Timeout Retry	Specifies the delay before an attempt to subscribe to a particular user after a subscription failure. The VX waits for this duration before retrying the REGISTER request with the Registrar again. Enter the number of seconds for this retry duration. (minimum 100, maximum 1800, default 300)
Music on Hold Filename	Enter the name of the audio file to be played when a channel using this Trunk Group is put on hold. The file name should be entered without the file extension. Appropriate extensions are added dynamically depending on the codec negotiated for the call.
Ringback Audio Filename	Enter the name of the audio file to be played ringback generation is enabled on Trunk Group. The file name should be entered without the file extension. Appropriate extensions are added dynamically depending on the codec negotiated for the call.
Dead Call Detection	Check the box to enable the dead call detection feature of RTCP.
Reliable Provisional Responses	When enabled (checked), the VX does not include 100rel in 18x message.
Send Symmetric Packetization Time	
Block Privacy Information	Check the box to indicate that the VX will not send P-Asserted-Identity, Privacy, and P-Preferred-Identity headers in outgoing SIP messages.
Diversion to Use	Select which diversion header (Top-Most or Bottom-Most) should be used when multiple diversion headers come through in the Invite message.
SIP Transport	
Retrieve Diversion from To header	
Use tel: for Outgoing Invite	When checked, enables the telephone number to be used for the Outgoing invite.

Enable TLS **Enable TCP** **Enable UDP** **Reuse TLS Connecton** **Enable UDP** **Enable Mutual TLS**	Use the checkboxes to enable TCP, TLS, UDP, and Mutual TLS. • Mutual TLS enables mutual authentication. Mutual TLS enables TLS transport. • When all three transports are selected, the order of transport use is TLS then TCP then UDP. When TLS is in use, TCP is implicitly used as the underlying transport. However, if TCP is not enabling in the Trunk Group, the TLS connection will not fallback to TCP in the case of TLS failure. • If only TLS is selected there will be no fallback to TCP or UDP transport if the TLS connection fails. • Reuse TLS connection is used for all SIP connections that will reuse TLS connections. Default: enabled (checked).
Persistent TLS Connection For Registration	When selected, the TLS client registering with the VX can make a persistent TLS connection. All outbound calls to this TLS client will use that persistent connection.
Exclude UDP in Contact Header	Check the box to enable that transport=UDP is not added in Contact URI.
SIP Security	
Remote Certificate Name	Specifies the name of the remote Transport Layer security (TLS) server's certificate. It should be a Fully Qualified Domain Name (FQDN) or a DNS name. The default is an empty string. This field applies to a TLS client only.
Enable Remote Certificate Name check	Check to enable remote TLS server certificate name check against it's FQDN to validate that the server identity. Default is disabled. This field applies to TLS clients only
Allow SIP URI in TLS	Check to allow a URI in the TLS.
VX Behind NAT	
Public IP	Public IP address of the NAT. Configure when the VX is behind the NAT on the private side.
SIP Mode	
Registrant Mode	Enables or disables registrant mode.
Proxy-like Mode	Enables or disables proxy-like mode. Select the checkbox to enable SIP proxy-like mode When enabled, this Trunkgroup will act in proxy like mode for Registrations and SIP Phone calls. If a SIP phone tries to register with this Trunkgroup, the REGISTER request is sent to the "Registrar Address." If the Registrar challenges the request, the challenge information is sent to the phone. When the phone resends a REGISTER request with the credentials, that information is sent across to the Registrar again. If the Registrar accepts the registration, the user's address of record is also stored in VX. A SIP call will be in proxy like mode only if both the incoming and outgoing Trunkgroups are acting in Proxy Like mode. Therefore, this configuration is appropriate for SIP to SIP call. Authentication and Authorization functions the same way for INVITE requests also.
Registrant	
Reg-Error Retry	Specifies the delay before an attempt to subscribe to a particular user after a subscription failure. The VX waits for this duration before retrying the REGISTER request with the Registrar again. Enter the number of seconds for this retry duration. (minimum 100, maximum 1800, default 300)

Inter Register Time	Time between registration requests sent out by VX when in Registrant mode. This parameter can be used to slow down the number of registration requests per second that VX sends to the registrar.
Address in Request URI	Specifies the request URI to be used when registering with registrar If not set, the same information as the to: will be used for the registration
Process Stale Parameter	Check the box to enable that the VX will immediately re-send the Register request if it is received 'stale' = true in the 410 response of Register request.
Proxy Like	
Min Proxy Reg Expiry	The proxy registration time to live value.
Backup Registrar Address	If it is determined that the Primary registrar is not reachable, VX sends the SIP Registration requests to the backup registrar defined in this field. This feature is supported for the proxy-like mode only. Either the IP address or FQDN can be used to configure this value.
Challenger	
Realm	Specifies the Realm in which the subscriber's information is valid This is one of the SIP Digest authentication parameters. This parameter defines the context for the authentication. It is similar to a zone-id in the H.323 protocol. Leave this parameter blank if any realm is acceptable. (Specify a character string with a maximum of 127 characters. The default value is blank.
RTCP	
Interval (secs)	Enter the transmission interval in seconds. This is the interval at which the RTCP reports are transmitted. The range is 5 seconds to 75 seconds.
RTCP XR	Select this checkbox to enable RTCP Extended Reports
Metrics Reports	
Interval Reports	Check this box to enable interval reports to be sent to a SIP Server
Session Reports	Check this box to enable session reports to be sent to a SIP Server
ICE	
Enable ICE	Enabling Interface Connectivity Establishment (ICE) on the Trunk group allows the SIP channels in the Trunk group to propose ICE candidates in INVITE and 183/200 responses based on the call direction. If no candidate is proposed by the other side the call proceeds as if ICE was not even used. If ICE is disabled on a Trunk group and the incoming offer contains ICE, VX will respond with no ICE attributes in the SDP. That is VX does not support ICE for both incoming and outgoing calls when ICE is disabled.
STUN Keep alive (secs)	ICE requires that RTP no-op packets be sent periodically as keep-alive messages in order to keep the pin holes in the Network Address Translators (NATs) open. The recommended default value is 20 seconds since most of the NATs close the pin holes in 30 seconds.
QoE	
Enable QoE Reporting	Enables sending Quality of Experience reports to Microsoft QoE Server.

H.323 Tab

Edit Trunk Groups Dialog H.323 Tab



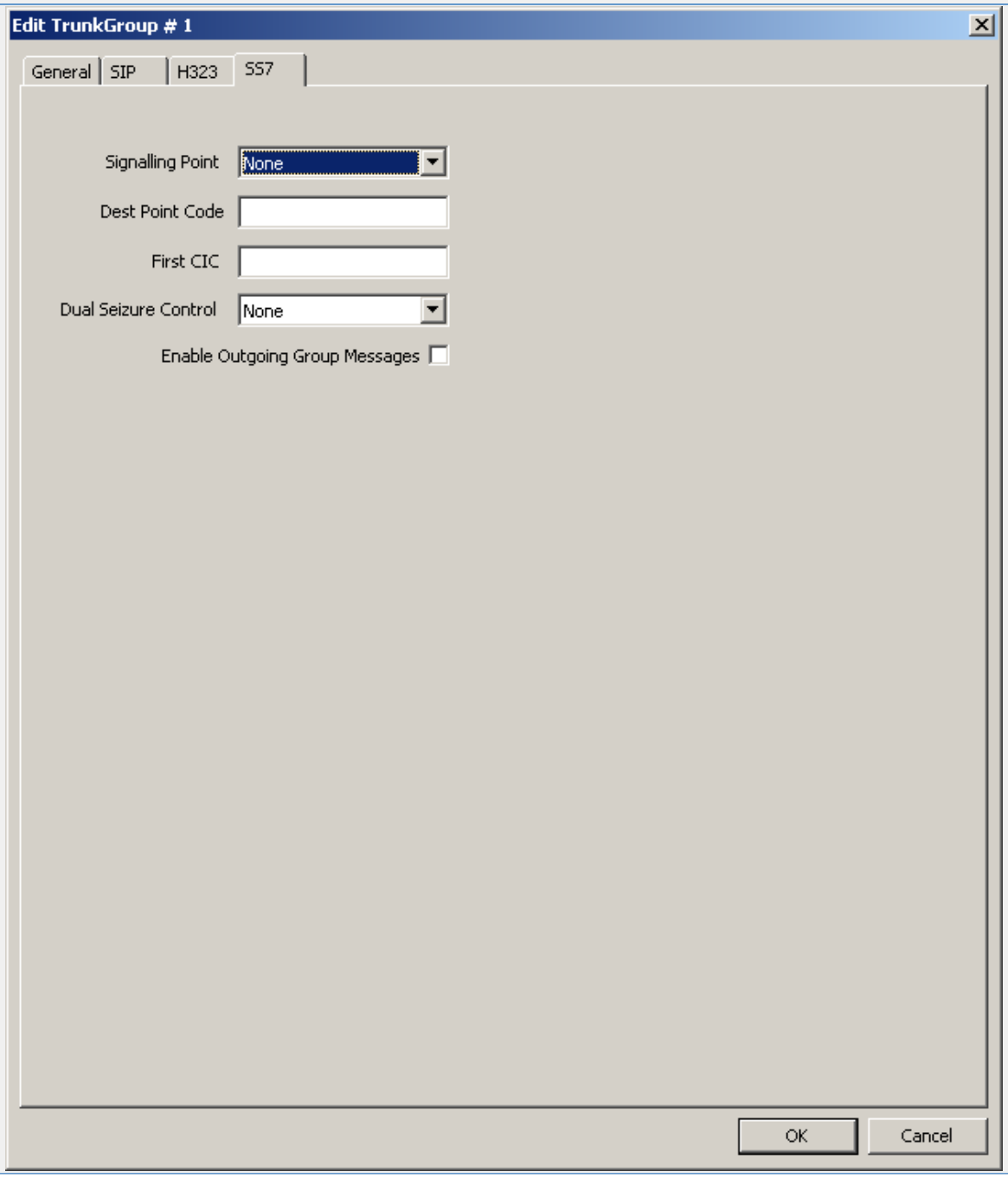
Field	Description
Gatekeeper	IP address of the gatekeeper
Gatekeeper ID (Zone)	This field is used for working with Cisco Routers Refer to the Cisco Router documentation for additional information.

Phone Number	The phone number is a comma separated list of E.164 phone numbers that will be used in the gatekeeper registration. The valid characters are as follows:
	`1 2 3 4 5 6 7 8 9 0 # * , \`
	• The "*" character placed after a phone number indicates to the gatekeeper that the number is a supported phone prefix. Gatekeepers use this prefix to match which destination to send a call to. The prefix is normally in the form 123#. Since the tech prefix is appended onto the called number, the # makes it easy to see where the prefix stops and the rest of the number starts.
	⚠ For VX, the tech prefix is specified in the H.323 phone number by putting an "**" after the number. For example, for the number 123#, the tech prefix would be 123#**. Multiple tech prefixes and phone numbers can be entered by separating them with commas: 123#*, 654#
	• The " \ " character is used to escape printing special characters, such as "**". 1234\ * sends the phone number *1234*** to the gatekeeper instead of the supported prefix 1234, for example:
	`1234,5678,333*,555#\*`
	The string shown above registers the following numbers with the gatekeeper:
	1234
	5678
	555#*
	and the following prefixes
	333

Auto Discover	Select yes or no to enable or disable autodiscovery. If set to enable (ON) and the above gatekeeper field is not set (no IP address entered) VXbuilder will perform discovery to determine the closest gatekeeper. If enabled and the above gatekeeper field is filled in (an IP address is entered) the autodiscovery option is ignored.
Disable Fast Start	Disables the fast start media negotiation protocol The device has to then rely on H.245 to setup the audio. In most cases this is disabled since fast start ensures fast call setups, ringback and so forth.
RAI Threshold (%)	The RAI is a notification from VX to a gatekeeper that the current call capacity for that trunk group is about to be exceeded The threshold sets at what level of capacity of the trunk group the message is sent. Once the message is sent, the gatekeeper will not route any more calls to the VX.

SS7 Tab

Edit Trunk Groups Dialog SS7 Tab



Field	Description
Signaling Point	Specify the SS7 Signaling point
Dest Point Code	Target SS7 point code for this trunk group's peer
First CIC	The first CIC code channels in this trunk group will use

Dual Seizure Control	The exchange with the higher Signaling point code will control all even-numbered circuits (by CIC) and the other exchange control odd numbered circuit Note: The controlling exchange is the exchange that gets the call completed in the event of dual seizure, for example, the non-controlling exchange has to process its received IAM. Options include: • None - makes VX the non-controlling exchange for all circuits in a trunk group.This is the required setting when the trunk group direction is Inbound. Otherwise, VXbuilder will return the error message: `Dual Seizure Control must be set to None for inbound SS7 TrunkGroup`. • All - makes VX the controlling exchange for all circuits in a trunk group. This is the required setting when the trunk group direction is Outbound. Otherwise, VXbuilder will return the error message: `Dual Seizure Control must be set to All for inbound SS7 TrunkGroup`. • Even/Odd as defined in ITU Q.764 for bidirectional traffic.For this setting, the point code and CIC number method is used to determine which is the controlling exchange. Dual Seizure Control must be configured as Even/Odd in bidirectional SS7 trunkgroups for all ITU ISUP variants. Otherwise, VXbuilder will return the error message: `Dual Seizure Control must be set to Even/Odd for ITU SS7 TrunkGroup`.
Enable Outgoing Group Messages	Select this checkbox if the far end device supports group blocking/unblocking messages Enabling this option is the most efficient method of blocking and resetting circuits.