ShoreTel	Corvisa a ShoreTel company	Innovation Network App Note IN-16017 Date : Feb, 2016
Product: ShoreTel Native| Corvisa SIP Trunking		System version: ShoreTel 14.2

ShoreTel & Corvisa SIP Trunking (Native)

SIP Trunking allows the use of Session Initiation Protocol (SIP) communications from Corvisa instead of the typical analog, Basic Rate Interface **(**BRI), T-1 or E-1 trunk connections. Having the pure IP trunk to the Internet Telephony Service Provider allows for more control and options over the communication link. This application note provides the details on connecting the ShoreTel IP phone system to Corvisa for SIP Trunking.

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ShoreTel tests and validates the interoperability of the Member's solution with ShoreTel's published software interfaces. ShoreTel does not test, nor vouch for the Member's development and/or quality assurance process, nor the overall feature functionality of the Member's solution(s). ShoreTel does not test the Member's solution under load or assess the scalability of the Member's solution. It is the responsibility of the Member to ensure their solution is current with ShoreTel's published interfaces.

The ShoreTel Technical Support organization will provide Customers with support of ShoreTel's published software interfaces. This does not imply any support for the Member's solution directly. Customers or reseller partners will need to work directly with the Member to obtain support for their solution.

Overview

This document provides details for connecting the ShoreTel® system to Corvisa's SIP Trunking network, which enables audio communications. The document also focuses on the network architecture needed to set up these systems to interoperate.

Note:

<u>The validation testing and this specific Application Note are ONLY applicable to the Corvisa network based on the **SIP Trunking** infrastructure, and therefore supported features with Corvisa's other networks may vary.</u>
<u>Please consult your Corvisa representative to ensure that this is applicable to your deployment.</u>

Corvisa Overview and Contact

Corvisa SIP Trunking lets you take advantage of all the cloud has to offer while allowing you to keep your current phone system. Backed by our carrier-class network, Corvisa SIP Trunking delivers high quality, reliable digital voice service without the costs of hardware.

Corvisa SIP Trunking Technical Support: https://www.corvisa.com/support/

Document Change History

Version 1 Issue 1 02/26/2016; Initial draft

Special Notes

ShoreTel Virtual Switch Support

Starting with ShoreTel 14.2, ShoreTel added support for Virtual Trunk and Virtual Phone switches. This Application Note assumes the setup, configuration and licensing of the Virtual/Physical Switches has already been completed. If you require additional information on Virtual Trunk Switch / Virtual Phone Switch, please refer to the ShoreTel Planning and Installation guide at following location:

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http://support.shoretel.com/products/ip_phone_system/shoretel_14.2/downloads/shoretel_14.2_install_guide.pdf

Emergency 911 / E911 Services

E911 services for Corvisa SIP Trunks are disabled by default. Customers can work with Corvisa directly to have these services provisioned.

Fax Support

The Fax support is limited to G711 Passthrough and only with ShoreTel ShoreGear switches. The support for T38 fax and fax on Virtual Trunk switches will be added in a later release.

Requirements, Certification and Limitations

Please refer to the ShoreTel Administration Guide, Chapter 18 – Session Initiation Protocol, for supported and unsupported features via SIP Trunks. Following are some feature limitations via SIP Trunks:

- Fax redirect not supported via SIP Trunks using G.711 (though Direct Inward Dialing (DID) to fax endpoint is supported)
- ShoreTel supports Music On Hold (MOH) over SIP trunks. The maximum number of music on hold (MOH) streams that a SIP-enabled switch can support varies with the switch model. The range of such streams across all the voice switch models is 14–60. Limitation: MOH source needs be on SIP trunk switch.
- ShoreTel supports the Service Appliance (SA-100) conferencing / IM system from Release - 12. SIP trunk calls from / to the SA-100 is supported. The SA-100 accepts access codes in DTMF RFC2833 only.
- 4 to 6 party conferences, when a SIP trunk is involved, utilize Make Me conference ports.
- Silent Monitoring, Barge-In, Silent Coach, Park/Unpark , Call recording features are supported on a SIP trunk call only if SIP trunk is configured with SIP profile supporting media hairpinning and the trunk is on a half-width switch or when using a Virtual Trunk Switch.
- Silence detection on trunk-to-trunk transfers is not supported, it requires a physical trunk.
- The ShoreTel system does not initiate calls with a 30ms payload; all calls are initiated with a 20ms payload.
- Hunt Group to Hunt Group transfer on the legacy ShoreGear switches is not currently supported. The support for this scenario will be added in a future release.

At this time we are unable to provide additional information on a resolution to the issues mentioned above, but suggest to periodically refer to the ShoreTel 14.2 Software Release Notice (Build Notes) for updates, which can be found at the following location:

http://support.shoretel.com

There may be other feature limitations when using SIP Trunks. Please refer to Chapter 18 of the ShoreTel Administration Guide.

By default, Virtual Trunk switches include predefined “SIP Media Proxy” resources; therefore, no configuration is required. With Physical Switches, “SIP Media Proxy” resources are not allocated by default and must be configured as per requirement. Please refer to the ShoreTel Partner guide for additional details about SIP Media Proxy and SIP Trunk capacity at the following location

http://partners.shoretel.com/product_sales_tools/ip_phone_system/shoretel_13/downloads/shoretel_13_partner_guide.pdf

This same guide is also applicable for half width physical switches in14.x release.

Version Support

Products are certified via the Innovation Network Certification Process for the ShoreTel system.

		Corvisa SIP Trunking
ShoreTel Release	14.2 Build 19.43.7902.0	

Corvisa Certification Testing Results Summary

N/S = Not Supported   N/T= Not Tested   N/A= Not Applicable

Primary Switch Test Plan (ShoreTel Virtual Trunk Switch)

ID	Result	Name	Description	Notes
1.1	PASS	Setup and Initialization	Verify successful setup and initialization of the SUT	
1.2	PASS	Outbound Call (Domestic)	Verify calls outbound placed through the SUT reach the external destination	
1.3	PASS	Inbound Call (Domestic)	Verify calls received by the SUT are routed to the default trunk group destination	
1.4	PASS	All Trunks Busy – Inbound Callers	Verify an inbound callers hears busy tone when all channels/trunks are in use	
1.5	PASS	All Trunks Busy – Outbound Callers	Verify an outbound callers hears busy tone when all channels/trunks are in use	
1.6	PASS	Incomplete Inbound Calls	Verify proper call progress tones are provided and proper call teardown for incomplete inbound calls	
2.1	PASS	Codec Negotiation	Verify codec negotiation between the SUT and the calling device with each side configured for a different codec	Corvisa uses G711 as a preferred codec.
2.2	PASS	DTMF Transmission – Out of Band/ In Band	Verify transmission of in-band and out-of-band digits per RFC 2833 for various devices connected to the SUT	

ID	Result	Name	Description	Notes
2.3	PASS	Auto Attendant Menu	Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the desired extension	
2.4	PASS	Auto Attendant Menu checking Voicemail mailbox	Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the Voicemail Login Extension	
3.1	PASS	Post Dial Delay	Verify that post dial delay is within acceptable limits	
4.1	PASS	Caller ID Name and Number - Inbound	Verify that Caller ID name and number is received from SIP endpoint device	
4.2	PASS	Caller ID Name and Number - Outbound	Verify that Caller ID name and number is sent from SIP endpoint device	
4.3	N/T	Hold from SUT to SIP Reference	Verify successful hold and resume of connected call	
4.4	PASS	Call Forward - SUT	Verify outbound calls that are being forwarded by the SUT are redirected and connected to the appropriate destination	
4.5	PASS	Call Forward – External PSTN Number	Verify outbound calls that are being forwarded by the SUT are redirected and connected to the appropriate destination	
4.6	PASS	Call Transfer – Blind	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination	
4.7	PASS	Call Transfer – Consultative	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination	
4.8	N/S	Outbound 911	Verify that outbound calls to 911 are routed to the correct PSAP for the calling location and that caller ID information is delivered	Please refer to Special Notes Section
4.9	N/S	Operator Assisted	Verify that 0+ calls are routed to an operator for calling assistance	Operator Assistance services 0+ are not supported by Corvisa
4.10	PASS	Inbound / Outbound call with Blocked Caller ID	Verify that calls with Blocked Caller ID route properly and the answering phone does not display any Caller ID information	

ID	Result	Name	Description	Notes
4.11	PASS	Inbound call to a Hunt Group	Verify that calls route to the proper Hunt Group and are answered by an available hunt group member with audio in both directions using G.729 and G.711 codecs	.
4.12	PASS	Inbound call to a Workgroup	Verify that calls route to the proper Workgroup and are answered successfully by an available workgroup agent with audio in both directions using G.729 and G.711 codecs	
4.13	PASS	Inbound call to DNIS/DID and leave a voice mail message	Verify that inbound calls to a user, via DID/DNIS, routes to the proper user mailbox and a message can be left with proper audio	
4.14	PASS	Call Forward – "FindMe"	Verify that inbound calls are forwarded to a user's "FindMe" destination	
4.15	N/S	Inbound / Outbound Fax Calls	Verify that inbound / outbound fax calls complete successfully	Fax is not currently supported on Virtual Trunk switches with Corvisa SIP Trunks.
4.17	PASS	Inbound call to Bridged Call Appearance (BCA) Extension	Verify that inbound calls properly presented to all of the phones that have BCA configured and that the call can be answered, placed on-hold and then transferred	
4.18	PASS	Inbound call to a Group Pickup Extension	Verify that inbound calls properly presented to all of the phones that have Group Pickup configured and that the call can be answered, placed on-hold and then transferred	
4.19	PASS	Office Anywhere External	Verify that inbound calls are properly presented to the Office Anywhere External PSTN destination	
4.20	PASS	Simul Ring	Verify that inbound calls are properly presented to the desired extension and the "Additional Phones" destinations	
4.21	PASS	MakeMe Conference	Verify that an inbound call can be conferenced with three (or more) additional parties	
4.22	PASS	Park / Unpark	Verify that an inbound call can be parked and unparked	

ID	Result	Name	Description	Notes
4.23	PASS	Call Recording	Verify that external calls can be recorded via the SIP Trunk using ShoreTel Communicator	
4.24	PASS	Silent Monitor / Barge-In / Whisper Page	Verify that external calls can be silently monitored, barged-in and whisper paged via the SUT	
4.25	PASS	Long Duration – Inbound	Verify that an inbound call is established for a minimum of 30 minutes	
4.26	PASS	Long Duration – Outbound	Verify that an outbound call is established for a minimum of 30 minutes	
5.1	N/A	SIP Registration		

Secondary Switch Sanity Test Results (ShoreTel ShoreGear Switch)

ID	Result	Name	Description	Notes
1.2	PASS	Outbound Call (Domestic)	Verify calls outbound placed through the SUT reach the external destination	
1.3	PASS	Inbound Call (Domestic)	Verify calls received by the SUT are routed to the default trunk group destination	
2.2	PASS	DTMF Transmission – Out of Band / In Band	Verify transmission of in-band and out-of-band digits per RFC 2833 for various devices connected to the SUT	
4.5	PASS	Call Forward – External PSTN Number	Verify outbound calls that are being forwarded by the SUT are redirected and connected to the appropriate destination	
4.6	PASS	Call Transfer – Blind	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination	
4.7	PASS	Call Transfer – Consultative	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination	
4.12	PASS	Inbound call to a Workgroup	Verify that calls route to the proper Workgroup and are answered successfully by an available workgroup agent with audio in both directions using G.729 and G.711 codecs	
4.15	CONDITIONAL PASS	Inbound / Outbound Fax Calls	Verify that inbound / outbound fax calls complete successfully	Only G711 Passthrough fax is supported
4.21	PASS	MakeMe Conference	Verify that an inbound call can be conferenced with three (or more) additional parties	

ID	Result	Name	Description	Notes
4.23	PASS	Call Recording	Verify that external calls can be recorded via the SIP Trunk using ShoreTel Communicator	
4.24	PASS	Silent Monitor / Barge-In / Whisper Page	Verify that external calls can be silently monitored, barged-in and whisper paged via the SUT	
4.27	PASS	Contact Center	Verify that an inbound call can be established directly to the ShoreTel Contact Center, that all prompts are heard and the agent can answer the call	

ShoreTel Configuration

The configuration information below shows examples for configuring ShoreTel, and Corvisa. Even though configuration requirements can vary from setup to set up, the information provided in these steps, along with the Planning and Installation Guide and documentation provided by Corvisa should prove to be sufficient. However, every design can vary and some may require more planning then others.

This section provides the general system settings and trunk configurations (both group and individual) required for a ShoreTel system to support SIP Trunking.

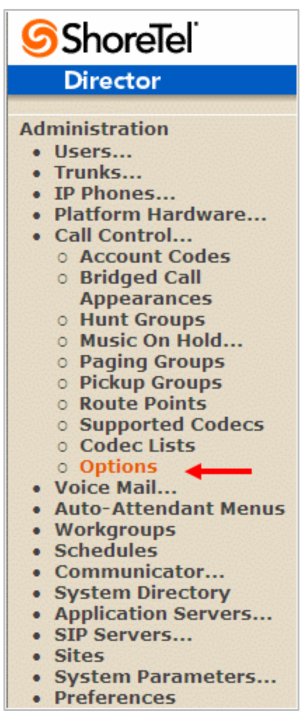
SHORETEL SYSTEM SETTINGS – GENERAL

General system settings include settings for Call Control, the Site and the Switch. If you confirm that the settings have already been configured as described in this section, proceed to the section titled, "ShoreTel System Settings – Trunk Groups". Otherwise, follow the instruction below.

CALL CONTROL SETTINGS

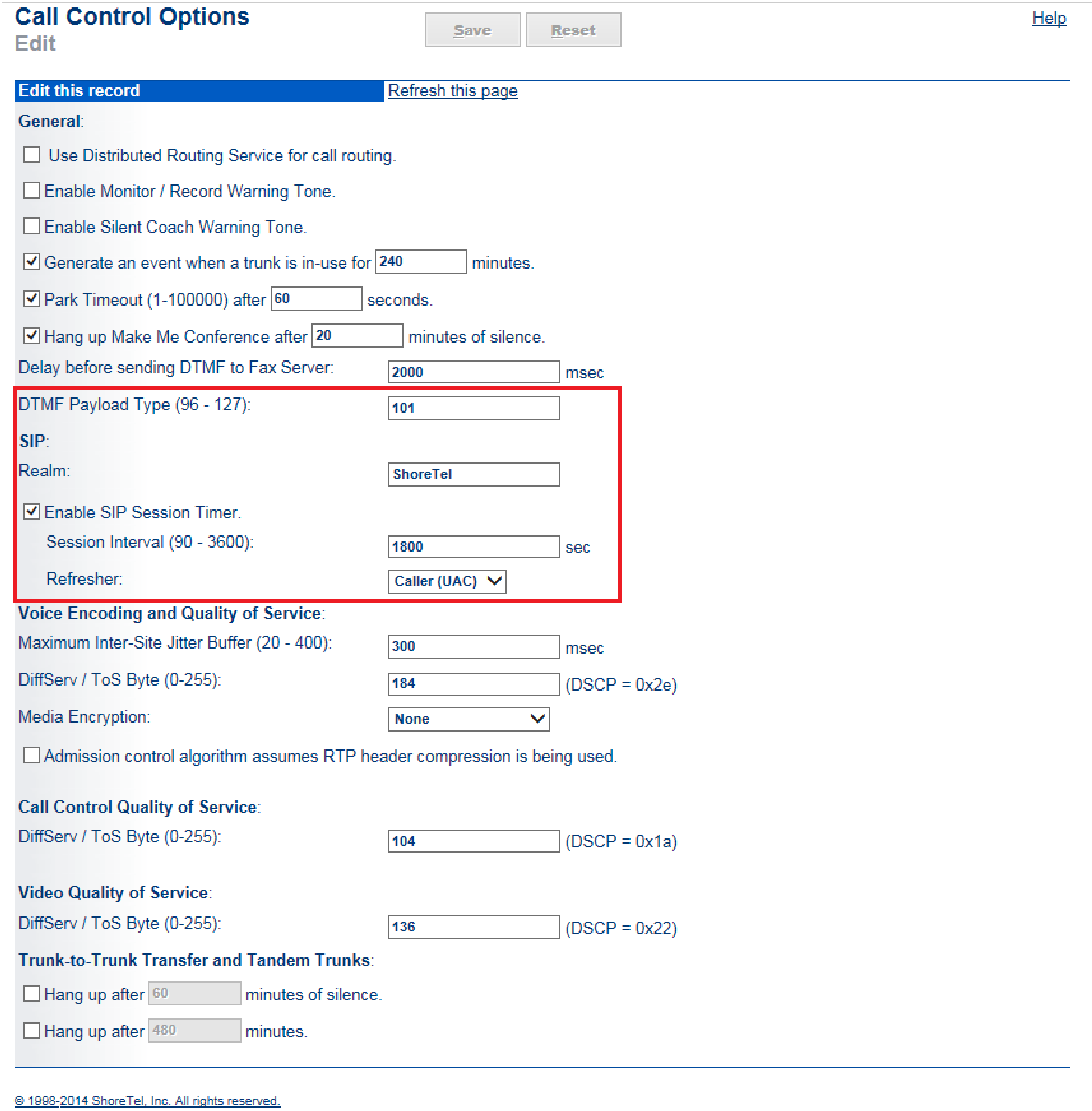
The first settings to configure within ShoreTel Director are the Call Control Options. To configure these settings for the ShoreTel system, log into ShoreTel Director and select "**Administration**" then "**Call Control**" followed by "**Options**" (**Figure 1**).

Figure 1 - Administration Call Control Options



The "Call Control Options" screen will then appear (**Figure 2**).

Figure 2 - Call Control Options



In the "**General**" parameters, the "**DTMF Payload Type (96 – 127)**" needs to be modified to a value of "101" to work properly with Corvisa SIP Trunking.

Within the "**SIP"** parameters, confirm that the appropriate settings are made for the "**Realm**" "**Enable SIP Session Timer**" parameters.

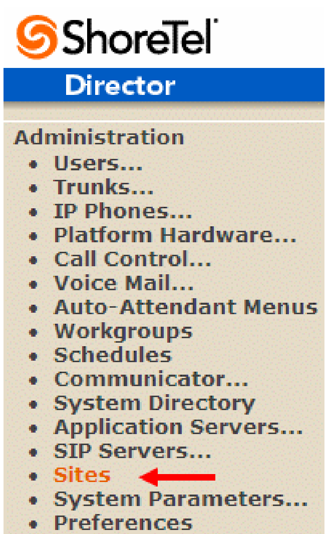
The “**Realm**” parameter is used in authenticating all SIP devices. It is typically a description of the computer or system being accessed. Changing this value will require a reboot of all ShoreGear switches serving SIP extensions. It is not necessary to modify this parameter to get the ShoreTel IP PBX system functional with Corvisa. Verify that the “**Enable SIP Session Timer**” box is checked (enabled). Next the Session Interval Timer needs to be set. The recommended setting for “**Session Interval**” is “1800” seconds. The last item to select is the appropriate refresher (from the pull down menu) for the SIP Session Timer. The “**Refresher**” field will be set either to “Caller (UAC)” [User Agent Client] or to “Callee (UAS)” [User Agent Server]. If the “Refresher” field is set to “Caller (UAC)”, the Caller’s device will be in control of the session timer refresh. If “Refresher” is set to “Callee (UAS)”, the device of the person called will control the session timer refresh.

The next settings to verify are the “**Voice Encoding and Quality of Service**”, specifically the “**Media Encryption**” parameter, make sure this parameter is set to “None”, otherwise you may experience one-way audio issues. Please refer to ShoreTel’s Administration Guide for additional details on media encryption and the other parameters in the “Voice Encoding and Quality of Service” area.

SITES SETTINGS

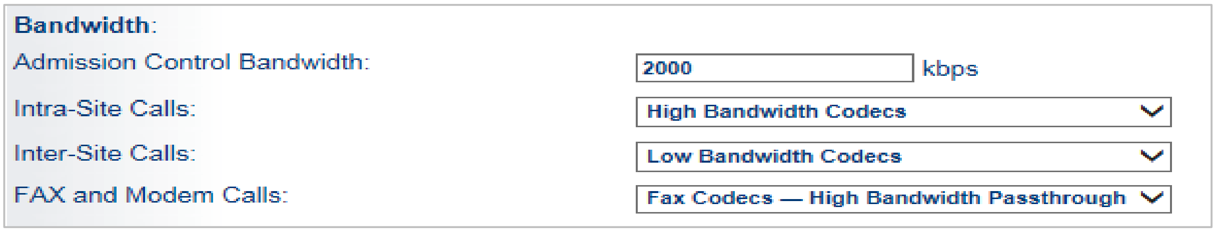
The next settings to address are the administration of sites. These settings are modified under the ShoreTel Director by selecting “**Administration**”, then “**Sites**” (**Figure 3**).

Figure 3 – Site Administration



This selection brings up the “Sites” screen. Within the “Sites” screen, select the name of the site to configure. The “Edit Site” screen will then appear. The only changes required to the “Edit Site” screen is to the “**Admission Control Bandwidth**” , “**Intra-Site / Inter-Site Calls**” and “**Fax and Modem Calls**” parameters (**Figure 4**).

Figure 4 – Site Bandwidth settings



Note: Bandwidth of 2000 is just an example. Please refer to the *ShoreTel Planning and Installation Guide* for additional information on setting Admission Control Bandwidth.

Sites Edit screen – Admission Control Bandwidth

The Admission Control Bandwidth defines the bandwidth available to and from the site. This is important as SIP trunk calls may be counted against the site bandwidth. Bandwidth needs to be set appropriately based on site setup and configuration with Corvisa SIP Trunking. See the ShoreTel Planning and Installation Guide for more information.

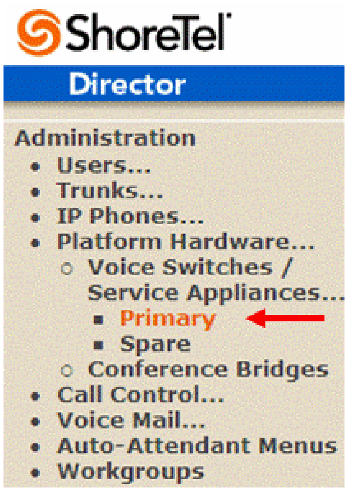
Sites Edit screen – Intra / Inter-Site Calls

By default, ShoreTel 14.x has 13 built-in codecs, these codecs can be grouped as “Codec Lists” and defined in the sites page for “Inter-site” and “Intra-site” calls. Configure the "Intra-Site Calls" option to a “Codec List” that contains the desired codecs and save the change. In the example above, we are using default codec list “High Bandwidth Codecs” for Intra-Site Calls and “Low Bandwidth Codecs” for Inter-Site calls. The site that the SIP Trunk Group belongs to will determine which “Intra-Site” Codec List will be utilized, be sure to move the desired codec up the list for higher priority. Please refer to the ShoreTel Planning and Installation Guide for additional information.

Switch Settings – Allocating Ports for SIP Trunks

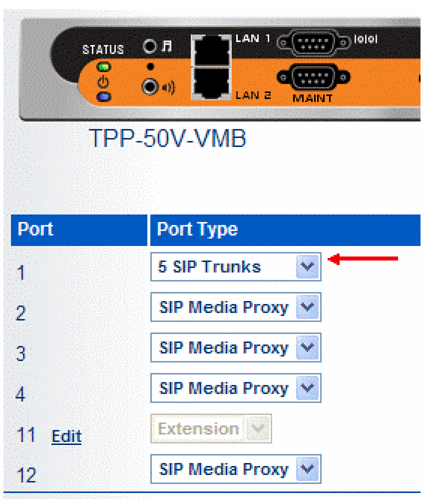
The final general settings to input are the ShoreGear switch settings. These changes are modified by selecting “**Administration**”, then “**Platform Hardware…**”, then “**Voice Switches / Service Appliances…**” followed by “**Primary**” in ShoreTel Director (**Figure 5**).

Figure 5 - Administration Switches



This action brings up the "**Switches**" screen. From the "Switches" screen simply select the name of the switch to configure. The "**Edit ShoreGear Switch**" screen will be displayed. Within the "Edit ShoreGear …Switch" screen, select the desired number of SIP Trunks from the ports available (**Figure 6**).

Figure 6 - ShoreGear Switch Settings



Each port designated as a SIP Trunk enables the support for 5 individual trunks.

Note: If you would like Music On Hold (MOH) to be played when calls are on hold, then the MOH source needs to be the same ShoreGear switch as the SIP Trunks. This is only applicable to ShoreTel physical switches as virtual trunk switch only supports File based MOH.

Starting with ShoreTel 13 and up through release 14.2, an additional option was added to the "Port Type" of half-width ShoreGear switches. The new selection is "SIP Media Proxy", it ensures that the ShoreTel system that is using SIP Trunks to have feature parity with PRI trunks. These include RFC

2833 DTMF detection for Office Anywhere External or Simultaneous Ring calls, three party mesh conferencing (without needing to configure "MakeMe" conference ports), call recording, Silent Monitoring, Barge-In, Whisper Page, Invites with no SDP and when there's no common codec between ITSP and the local extension.

With the introduction of ShoreTel 14.2, ShoreTel Virtual Trunk Switches include "SIP Media Proxy" resources, therefore, no configuration is required. With physical ShoreGear switches, "SIP Media Proxy" resources are not allocated by default and must be reserved/enabled to support various SIP features and functions (described in the previous paragraph).

For further information on "SIP Media Proxy" please refer to Chapter 18 of the ShoreTel 14.2 System Administration Guide.

If you are using the older full-width ShoreGear switches and you want perform 3 (or more) party conference calls with Corvisa SIP Trunking, please make sure that you have enabled a minimum of four "MakeMe" conference port resources. Conference resources are required with ShoreTel 14.2 on full-width ShoreGear switches for 3-way conference calls to function as expected. These resources may be on *any* switch that has spare ports and supports "MakeMe" conference resources.

SHORETEL SYSTEM SETTINGS – SIP PROFILES

Corvisa SIP Trunking uses default SIP Profile “Default ITSP”. Hence, no modification is necessary in this section.

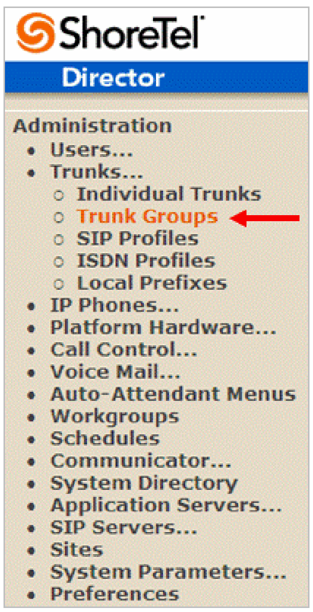
SHORETEL SYSTEM SETTINGS – TRUNK GROUPS

ShoreTel Trunk Groups only support Static IP Addresses for Individual Trunks.

In trunk planning, the following needs to be considered.

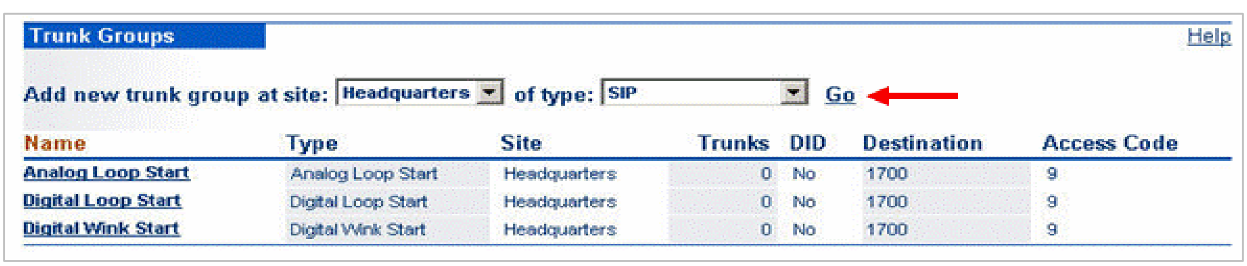
The settings for Trunk Groups are changed by selecting “**Administration**”, then “**Trunks**” followed by “**Trunk Groups**” within ShoreTel Director (**Figure 7**).

Figure 7 - Administration Trunk Groups



This selection brings up the “Trunk Groups” screen (**Figure 8**).

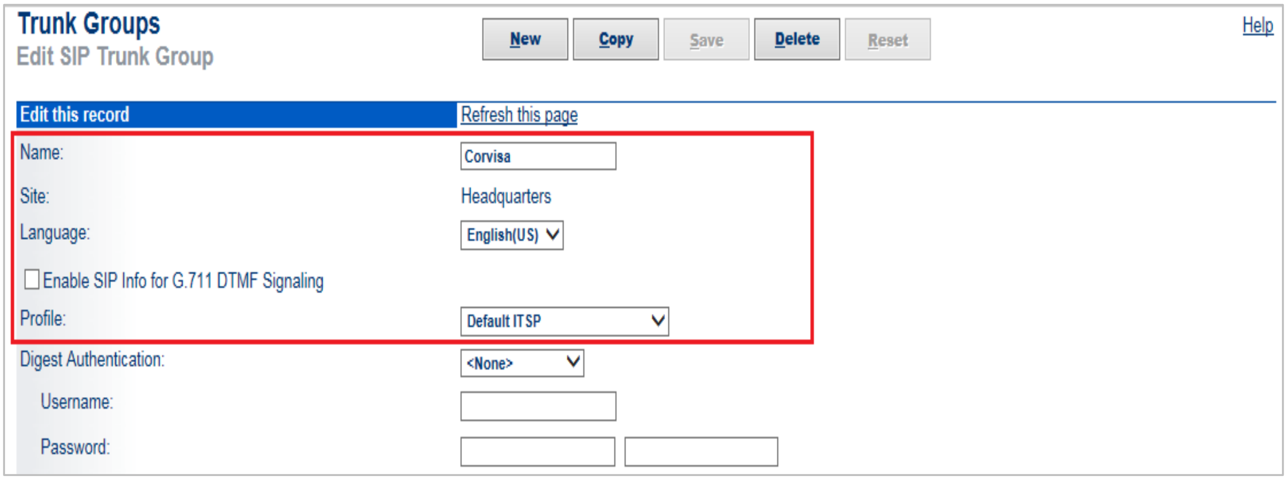
Figure 8 - Trunk Groups Settings



From the pull down menus on the “Trunk Groups” screen, select the site desired and select the “**SIP**” trunk type to configure. Then click on the “**Go**” link from “**Add new trunk group at site**”.

The “Edit SIP Trunk Group” screen will appear (**Figure 9**).

Figure 9 – Edit SIP Trunk Group



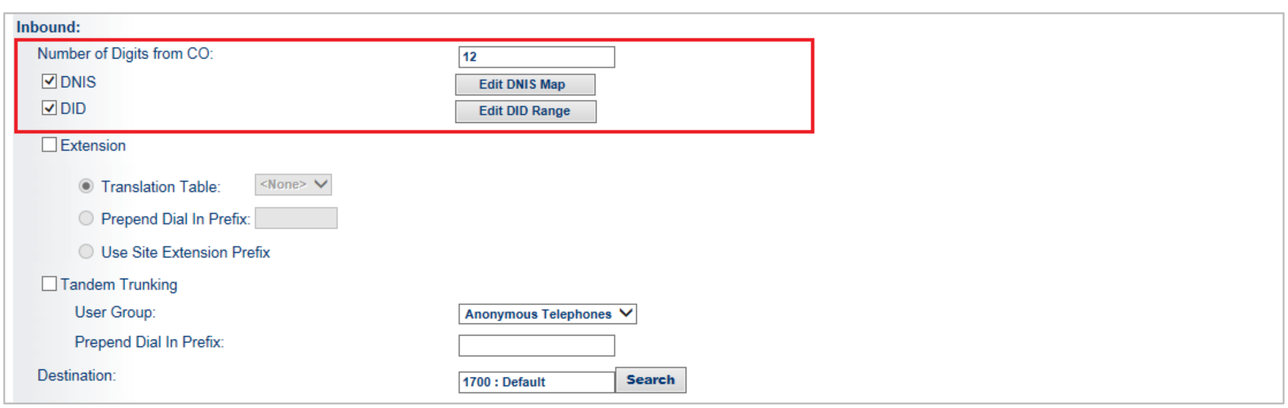
The next step within the “Edit SIP Trunks Group” screen is to input the name for the trunk group. In the example in **Figure 9**, the name “Corvisa” has been created.

The “**Enable SIP Info for G.711 DTMF Signaling**” parameter should not be enabled (checked). Enabling SIP info is currently only used with SIP tie trunks between ShoreTel systems.

In the **“Profile:”** parameter, use the down arrow (pull-down menu) and select “Default ITSP” sip profile.
The “**Enable Digest Authentication**” parameter defaults to “None” and need not to be configured for Corvisa SIP Trunks.

The next item to change in the “Edit SIP Trunks Group” screen is to make the appropriate settings for the “**Inbound:**” parameters. (**Figure 10**).

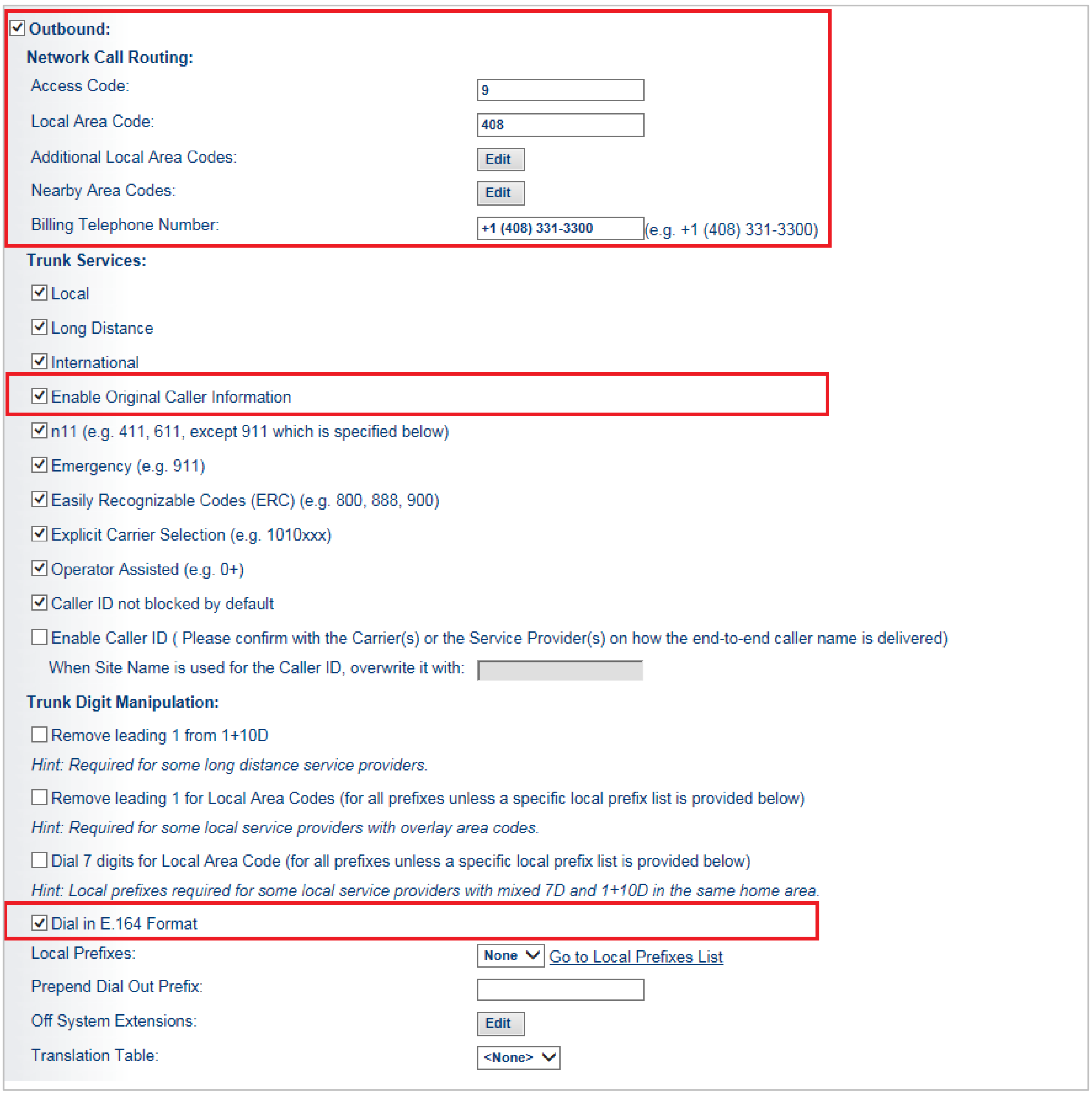
Figure 10 – Inbound



Within the "**Inbound:**" settings, ensure the "**Number of Digits from CO:**" is configured to a value of "12", this is the number of digits that the ShoreTel SIP Trunk Switch will be receiving from Corvisa SIP Trunking. Enable (check) the "**DNIS**" or "**DID**" parameters as needed. It is no longer needed to enable the "**Extension**" parameter. We also recommend that the "**Tandem Trunking**" parameter should be kept to the default value of disabled (unchecked) unless it is specifically required by the customer setup. For additional information on these parameters, please refer to the *ShoreTel Administration Guide*.

Note: The following section is configured no different than any normal Trunk Group

Figure 11 – Outbound and Trunk Services:



If outbound call service is required, enable (check) the "**Outbound**" parameter and define a Trunk "**Access Code**" and "**Local Area Code**" as appropriate. In addition you should also define the "**Billing Telephone Number**" with the appropriate main number provided by Corvisa SIP Trunking.

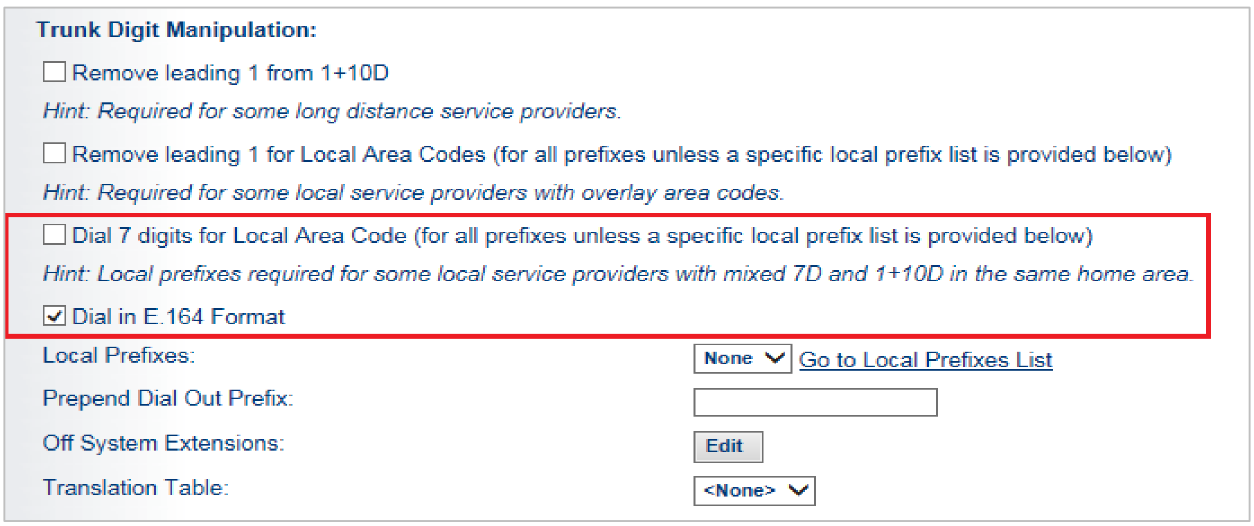
In the "**Trunk Services:**" area, make sure the appropriate services are enabled or disabled based on what Corvisa supports and what features are needed from this Trunk Group. Please select checkbox **"Enable Original Caller Information"** to enable diversion header required for call forwarding scenario.

The parameter "**Caller ID not blocked by default**" determines if the call is sent out as <unknown> or with caller information (Caller ID). User DID will impact how information is passed out to the SIP Trunk group.

After these settings are made to the "Edit SIP Trunk Group" screen, select the "**Save**" button to input the changes.

The final parameters for configuration in the Trunk Group are "**Trunk Digit Manipulation**" (**Figure 12**):

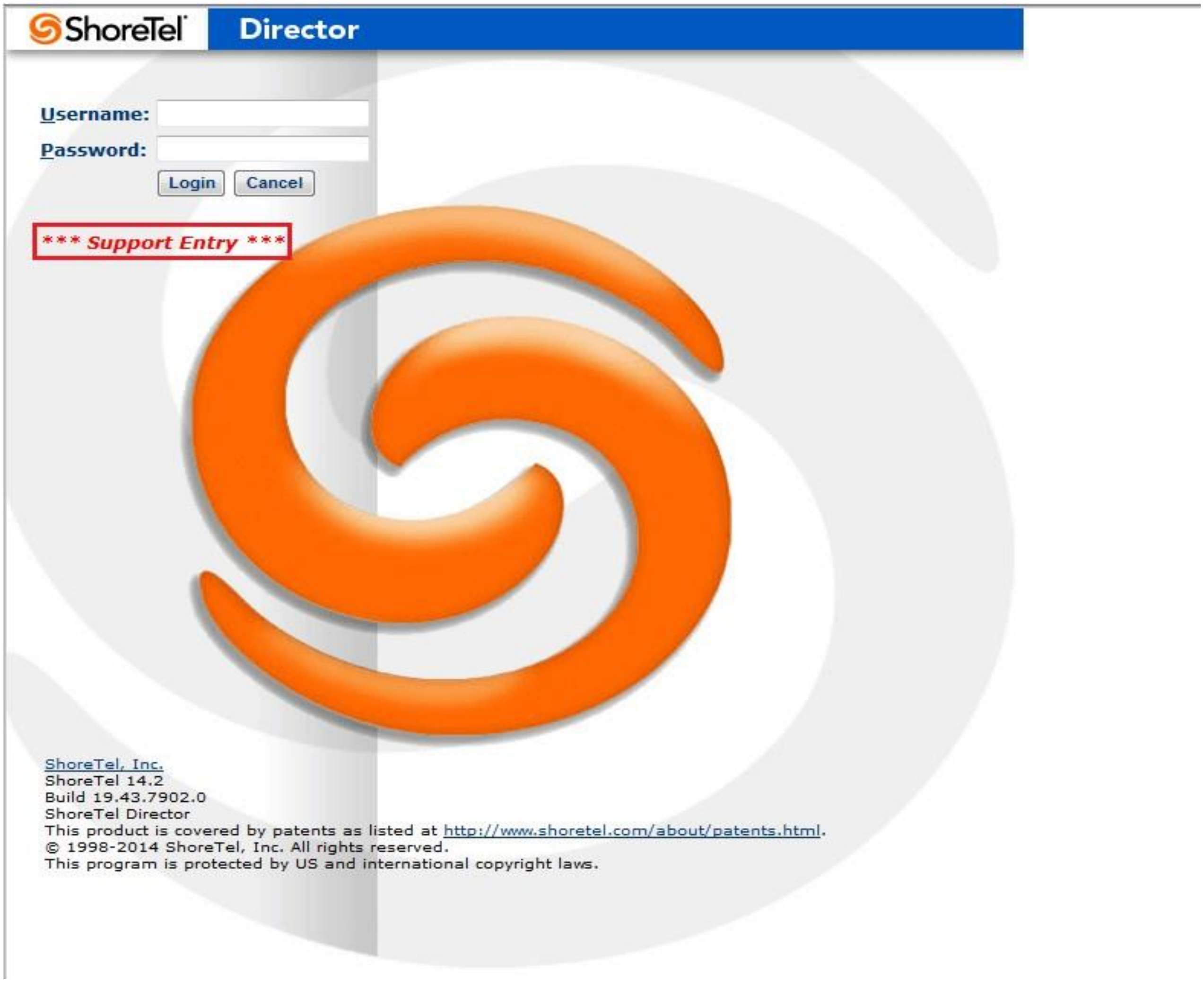
Figure 12 – Trunk Digit Manipulation:



The only other parameters that require adjustment (from default) to interface with Corvisa SIP Trunking are **"Dial 7 digits for Local Area Code"** and "**Dial in E.164 Format**". Disable (Uncheck) the parameter "**Dial 7 digits for Local Area Code**" and enable (Check) the "**Dial in E.164 Format**" parameter. **Save** the changes.

Logout of ShoreTel Director, you will then be presented with the ShoreTel Director login page. On your keyboard, hold down the <**CTRL**> and <**Shift**> keys and with the mouse pointer click on the "**Username:**" field. This will enable the "Support Entry" mode of the ShoreTel Director, as referenced below in (**Figure 13**).

Figure 13 – ShoreTel Director Support Entry:



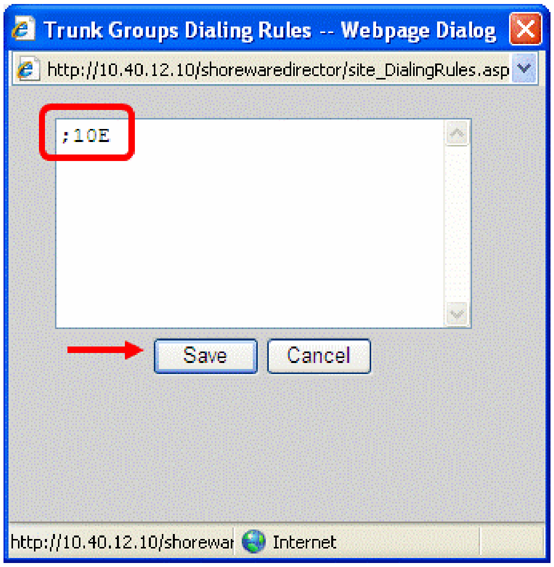
Log into ShoreTel Director with your normal administration user credentials.

Navigate to the "Edit SIP Trunk Group" page, by selecting "**Administration**" followed by "**Trunks…**", then "**Trunk Groups**", then in the "Trunk Groups" page, select the Trunk Group you created for Corvisa (see **Figure 9**). This action brings up the "**Edit SIP Trunk Group**" page. Scroll down to the bottom of the page, in the "**Trunk Group Dialing Rules:**" parameter section, to the right of the "**Custom:**" parameter click on the "**Edit**" button. As noted below in **Figure 14**.

Figure 14 – Trunk Group Dialing Rules:

This action brings up the "Trunk Groups Dialing Rules – Webpage Dialog" as noted below in **Figure 15**.

Figure 15 – Trunk Groups Dialing Rules – Webpage Dialog:



In the blank area of the "Webpage Dialog" enter **;10E** and click on the "**Save**" button. Be sure to enter the exact syntax, this includes the semicolon, one, zero followed by a capital E. This syntax is case sensitive, verify that it matches **Figure 15**.

This entry provides correct formatting for outbound Caller ID numbers.

This completes the settings needed to set up the trunk groups on the ShoreTel system.

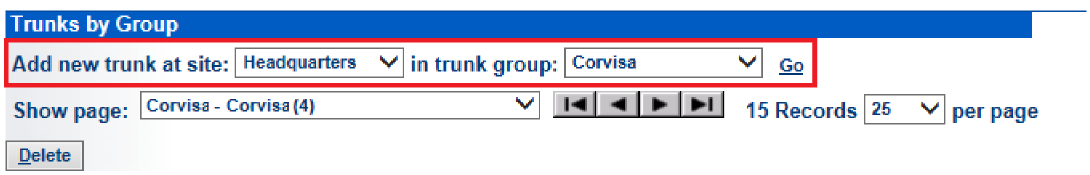
SYSTEM SETTINGS – INDIVIDUAL TRUNKS

This section covers the configuration of the individual trunks. Select "**Administration**", then "**Trunks**" followed by "**Individual Trunks**" to configure the individual trunks (**Figure 16**).

Figure 16 – Individual Trunks

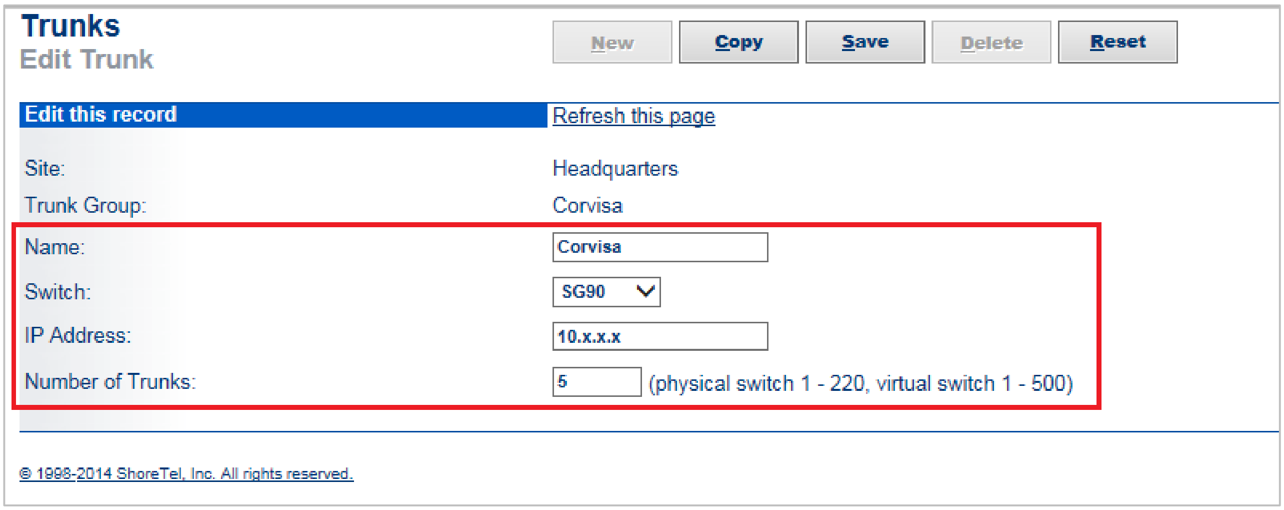
The "**Trunks by Group**" screen that is used to change the individual trunks settings then appears (**Figure 17**).

Figure 17 – Trunks by Group:



Select the site for the new individual trunk(s) to be added and select the appropriate trunk group from the pull down menu in the "**Add new trunk at site**" area. In this example, the site is "Headquarters" and the trunk group is "Corvisa"**.** Click on the "**Go**" button to bring up the "Edit Trunk" screen (**Figure 18**).

Figure 18 - Edit Trunks Screen for Individual Trunks



From the individual trunks "Edit Trunk" screen, input a "**Name:**" for the individual trunks, then select the appropriate "**Switch**". When selecting a name, the recommendation is to name the individual trunks the same as the name of the trunk group so that the trunk type can easily be tracked. Select the switch upon which the individual trunks will be created. For the parameter "**IP Address**", define the IP address of the Corvisa SIP Server. The last step is to select the number of individual trunks desired "**Number of Trunks (1 – 220)**" (each one supports "one" audio path – example if 10 is configured, then 10 audio paths can be up at one time). Once these changes are complete, select the "**Save**" button to commit changes.

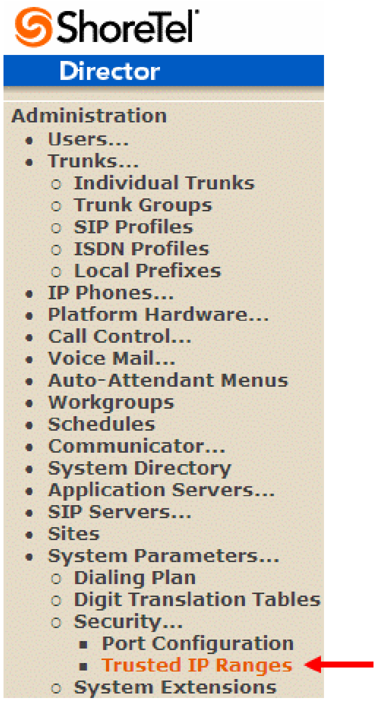
After setting up the trunk groups and individual trunks, refer to the ShoreTel Product Installation Guide to make the appropriate changes for the User Group settings.

SHORETEL SECURITY SETTINGS

The ShoreTel Service Appliances and Virtual Trunk Switch are sealed appliances, optimized for resiliency and security, designed to run ShoreTel services. In order to utilize the ShoreTel Service Appliances and Virtual Trunk Switch with Corvisa SIP Trunking platform, you will need to add Corvisa's Signaling and Media Gateway IP address into the "Trusted IP Ranges".

Select "**Administration**", then "**System Parameters…**", then "**Security…**" followed by "**Trusted IP Ranges**", as noted below in **Figure 19**.

Figure 19– Trusted IP Ranges

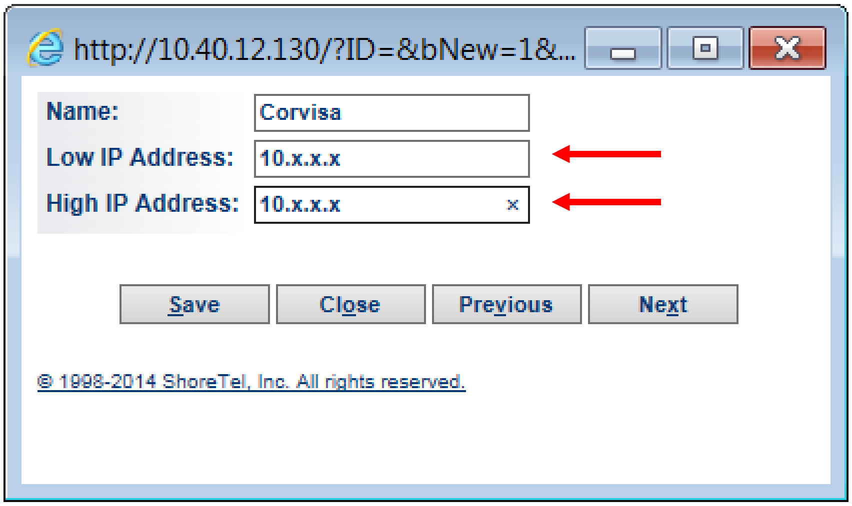


This action causes the Trusted IP Ranges page to appear. Select the "**New**" button, as shown below in **Figure 20**.

Figure 20– Trusted IP Ranges Page

This action causes the "Trusted IP Range Info" pop-up window to be displayed, as shown below in **Figure 21**.

Figure 21 – Trusted IP Range Info Pop-up



Define a name, we chose "Corvisa", then in the "**Low IP Address:**" and "**High IP Address:**" define the Corvisa Signaling and Media Gateway IP addresses. In our example, the Corvisa Media and Signaling Gateway IP address is 10.x.x.x. Please verify the actual IP addresses that you need to use with your Corvisa Representative. Once you have completed defining the values, select the "**Save**" button.

This completes the changes necessary on the ShoreTel Director to interoperate with Corvisa SIP Trunking.

Corvisa Configuration & Support

Corvisa will configure SIP trunks on its network and provide customers with IP addresses of SIP Proxy, and phone numbers assigned to customers before scheduled service activation date. For any queries, please contact following:

Corvisa Sales and Customer Support: 877-487-9256
Corvisa SIP Trunking Sales: sales@corvisa.com

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To be “ShoreTel Certified” means that Technology Partner's product will inter-operate with the ShoreTel system, but ShoreTel does not certify that the features or functionality of Technology Partner's product will perform as specified by Technology Partner nor that Technology Partner's product will meet your specific application needs or requirements. To inter-operate means that Technology Partner's product is able to exchange, use and share information with the ShoreTel system.

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