ShoreTel	fusion	Innovation Network App Note IN-14055 Date : November, 2014
Product: ShoreTel Native| Fusion SIP Trunking		System version: ShoreTel 14.2

ShoreTel & Fusion SIP Trunking (Native)

SIP Trunking allows the use of Session Initiation Protocol (SIP) communications from Fusion 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 Fusion 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 Fusion'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:

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

Fusion Overview and Contact

Fusion delivers a comprehensive suite of cloud solutions including SIP Trunking. In addition we can provide a variety of business continuity and disaster recovery solutions utilizing a customer's existing broadband, Fusion broadband, or a combination. Communications has been the core of Fusion's business for more than 15 years and we've never lost focus on driving the right solution at the right time and at the right price to customers both large and small. We've built a robust, carrier-grade network that ensures the highest quality, diversity and redundancy with clear connections to our cloud. All supported by a US based, Fusion employed, Sales and Technical Support teams.

Fusion Sales and Customer Support: 888-301-1721
Fusion SIP Trunking Technical Support: technicalsupport@fusionconnect.com

Document Change History

Version 1 Issue 1	11/24/2014; Initial draft
Version 1 Issue 2	12/5/2014; Final Application Notes

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

Codec Support
Fusion SIP Trunking platform only support G711 as a preferred Codec. Hence, only G711 codec will be supported on the ShoreTel system

Fax Support
Currently we only support G711-Passthrough Fax for both Physical and Virtual switch types. Support for T38 fax will be added later.

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.
- If the ShoreTel server has a conference bridge 4.2 installed, you should not enable SIP. The conference bridge is not compatible with a ShoreTel system that has SIP enabled due to the dynamic RTP port required for SIP.
- 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 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.

- External Party might not hear Music on Hold when Bridge Call Appearance User places call on hold, due to ShoreTel Defect 1-708568181. Please note that this issue is only identified on this specific scenario.
- Fusion SIP Trunking platform does not support sending 200OK as a valid response for SIP OPTIONS Message. To get around this issue, ShoreTel SIP Trunk profile need to be modified to accept 404 as a valid response in lieu of 200OK. At this time, we do not anticipate this to cause any issue but it may impact certain failover scenarios.

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.

		Fusion SIP Trunking
ShoreTel Release	14.2 Build 19.43.7902.0	

Fusion Certification Testing Results Summary

Basic test plan:

TABLE 1-1: INITIALIZATION AND BASIC CALLS

ID	Name	Description	Results
1.0	Configuration Application Note	Innovation Network Lab will use the configuration application note provided by the vendor to configure the vendor's product to work with the ShoreTel system.	Pass
1.1	Setup and initialization	Verify successful setup and initialization of the SUT	Pass
1.2	Outbound Call (Domestic)	Verify calls outbound placed through the SUT reach the external destination.	Pass
1.3	Inbound Call (Domestic)	Verify calls received by the SUT are routed to the default trunk group destination.	Pass
1.4	Device restart – Power Loss	Verify that the SUT recovers after power loss to the SUT	Pass
1.5	Device restart – Network Loss	Verify the SUT recovers after loss of network link to the SUT.	Pass
1.6	All Trunks Busy – Inbound Callers	Verify an inbound callers hears busy tone when all channels/trunks are in use	Pass
1.7	All Trunks Busy – Outbound Callers	Verify an outbound callers hears busy tone when all channels/trunks are in use	Pass
1.8	Incomplete Inbound Calls	Verify proper call progress tones are provided and proper call teardown for incomplete inbound calls.	Pass

TABLE 1-2: MEDIA AND DTMF SUPPORT

ID	Name	Description	Notes
2.1	Media Support – ShoreTel to SUT	Verify call connection and audio path from a ShoreTel phone to an external destination through the service provider using all supported codes with both sides set to a common codec.	Pass **Conditional** *See Note 1*
2.2	Media Support – SIP Reference to SUT	Verify call connection and audio path from a SIP Reference phones to an external destination through the service provider using all supported codes with both sides set to a common codec.	Pass **Conditional** *See Note 1*
2.3	Codec Negotiation	Verify codec negotiation between the SUT and the calling device with each side configured for a different codec.	Pass **Conditional** *See Note 1*
2.4	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.	Pass
2.5	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.	Pass
2.6	Auto Attendant Menu “Dial by Name”	Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the desired extension using the “Dial by Name” feature.	Pass
2.7	Auto Attendant Menu checking Voice Mail mailbox	Verify that inbound calls are properly terminated on the ShoreTel Auto Attendant menu and that you can transfer to the Voice Mail Login Extension.	Pass

TABLE 1-3: PERFORMANCE & QUALITY OF SERVICE

ID	Name	Description	Notes
3.1	Voice Quality Service Levels	Verify the SUT can provide a voice quality SLA across the WAN from the customer premises to the SUT SIP gateway.	Not Tested
3.2	Capacity Test	Verify the service provider interface can sustain services through period of heavy outbound and inbound load.	Not Tested
3.3	Post Dial Delay	Verify that post dial delay is within acceptable limits.	Pass

ID	Name	Description	Notes
3.4	Billing Accuracy	Verify that all test calls made are accurately reflected in the SUT's CDR and billing reports.	Pass

TABLE 1-4: ENHANCED SERVICES AND FEATURES

ID	Name	Description	Notes
4.1	Caller ID Name and Number - Inbound	Verify that Caller ID name and number is received from SIP endpoint device	Pass
4.2	Caller ID Name and Number - Outbound	Verify that Caller ID name and number is sent from SIP endpoint device	Pass
4.3	Hold from SUT to SIP Reference	Verify successful hold and resume of connected call	Pass
4.4	Call Forward - SUT	Verify outbound calls that are being forwarded by the SUT are redirected and connected to the appropriate destination.	Pass
4.5	Call Transfer – blind	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination.	Pass *See Note 2*
4.6	Call Transfer – Consultative	Verify a call connected from the SUT to the ShoreTel phone can be transferred to an alternate destination.	Pass
4.7	Conference – ad hoc	Verify successful ad hoc conference of three parties	Pass
4.8	Inbound DID/DNIS	Verify the SUT provides inbound "dialed number information" and is correctly routed to the configured destination.	Pass
4.9	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.	Not Tested *See Note 3*
4.10	Operator Assisted	Verify that 0+ calls are routed to an operator for calling assistance.	Not Tested *See Note 3*
4.11	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.	Pass
4.12	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.711 codecs.	Pass
4.13	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.711 codecs.	Pass

ID	Name	Description	Notes
4.14	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.	Pass
4.15	Call Forward – “FindMe”	Verify that inbound calls are forwarded to a user’s “FindMe” destination.	Pass
4.16	Call Forward Always	Verify that inbound calls are immediately automatically forwarded to a user’s external destination.	Pass
4.17	Inbound / Outbound Fax calls	Verify that inbound / outbound fax calls complete successfully.	Pass **Conditional** *See Note 4*
4.18	ShoreTel Converged Conferencing Server	Verify that inbound calls are properly forwarded to the ShoreTel Converged Conferencing Server and it properly accepts the access code and you’re able to participate in the conference bridge.	Pass
4.19	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.	Pass **Conditional** *See Note 5*
4.20	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.	Pass
4.21	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.	Pass
4.22	Office Anywhere External	Verify that inbound calls are properly presented to the Office Anywhere External PSTN destination.	Pass
4.23	Simul Ring	Verify that inbound calls are properly presented to the desired extension and the “Additional Phones” destinations.	Pass
4.24	MakeMe Conference	Verify that an inbound call can be conferenced with three (or more) additional parties	Pass

ID	Name	Description	Notes
4.25	Park / Unpark	Verify that an inbound call can be parked and unparked	Pass
4.26	Call Recording	Verify that external calls can be recorded via the SIP Trunk using ShoreTel Communicator	Pass
4.27	Silent Monitor / Barge-In / Whisper Page	Verify that external calls can be silently monitored, barged-in and whisper paged via the SUT.	Pass
4.28	Long Duration – Inbound	Verify that an inbound call is established for a minimum of 30 minutes.	Pass
4.29	Long Duration – Outbound	Verify that an outbound call is established for a minimum of 30 minutes.	Pass
4.30	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.	Pass

Table 0-5: Security

ID	Name	Description	Notes
5.1	Registration / Digest Authentication	Verify the SUT supports the use of registration / digest authentication for service access for inbound and outbound calls.	Pass

Note 1: Fusion only support G711 as Preferred codec irrespective of codec priority on ShoreTel. Hence, ShoreTel will only support G711 codec in its configuration.

Note 2: When a ShoreTel user attempts to blind transfer a call to an external number, Fusion's SIP Trunks generate a "183 Session Progress" message to the ShoreTel system, resulting in a consultative transfer.

Note 2: Fusion supports calls to emergency numbers (911) and to Operator Assistance on their production network, but these services were not enabled/tested during validation.

Note 4: T38 Fax is not currently supported on ShoreTel system due to defect on Virtual Trunk Switch. It will be addressed in future releases.

Note 5: Please refer to the section "Requirements, Certification and Limitations" for additional details.

ShoreTel Configuration

The configuration information below shows examples for configuring ShoreTel, and Fusion. 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 Fusion 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.

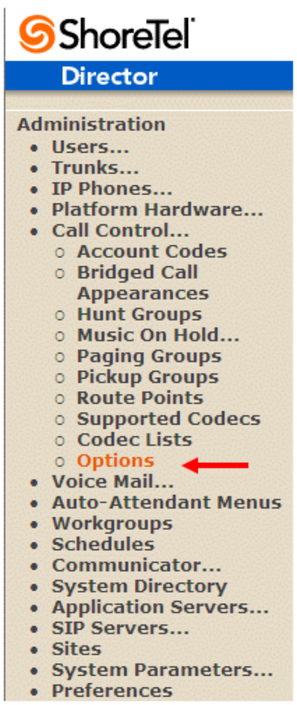
SHORETEL SYSTEM SETTINGS – GENERAL

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

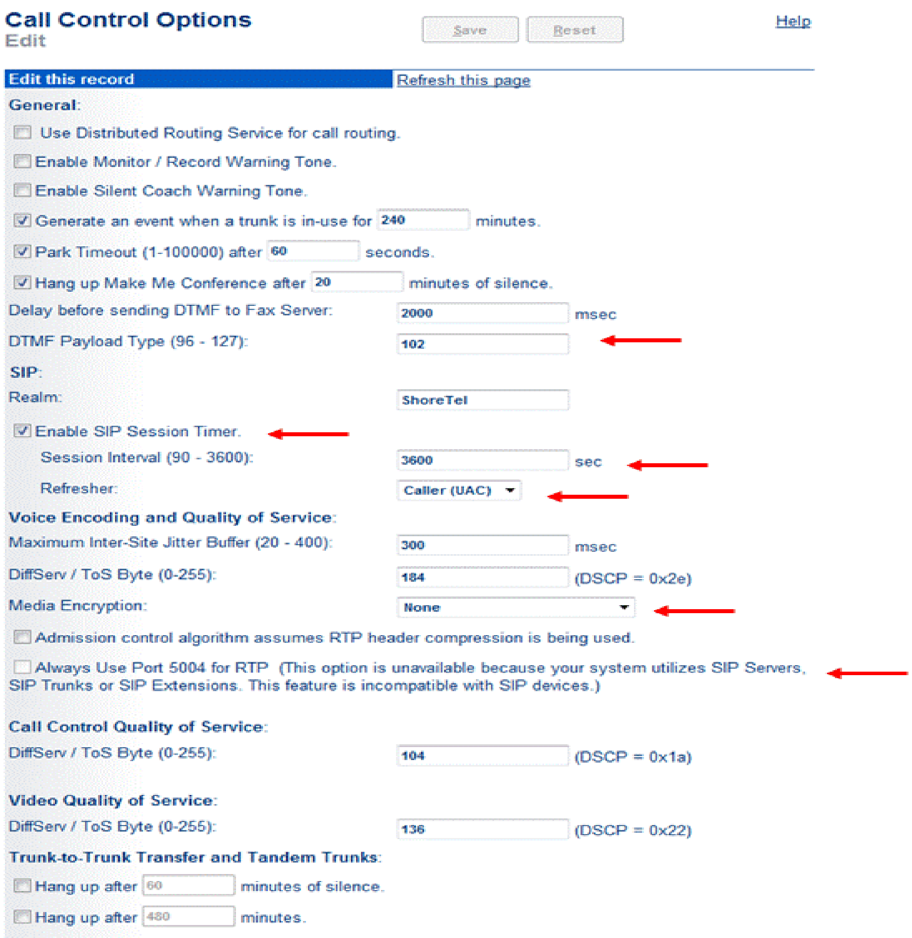
The first settings to configure within ShoreTel Director are the Call Control Options. To configure these settings for the ShoreTel system, log into ShoreTel Director and select "**Administration**" then "**Call Control**" followed by "**Options**" (**Figure 4**).

Figure 4 - Administration Call Control Options



The "Call Control Options" screen will then appear (**Figure 5**).

Figure 5 - Call Control Options



In the "**General**" parameters, the "**DTMF Payload Type (96 – 127)**" defaults to a value of "102", and no modification is necessary to interoperate with Fusion.

Within the "**SIP"** parameters, confirm that the appropriate settings are made for the "**Realm**" "**Enable SIP Session Timer**" and "**Always Use Port 5004 for RTP**" 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 Fusion. 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 "3600" 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.

The ShoreTel legacy parameter "**Always Use Port 5004 for RTP**" should be disabled by default, if it's enabled you will need to disable it. It is required for implementing SIP on the ShoreTel system. For SIP configurations, Dynamic User Datagram Protocol (UDP) must be used for RTP Traffic. If the parameter is disabled, Media Gateway Control Protocol (MGCP) will no longer use UDP port 5004; MGCP and SIP traffic will use dynamic UDP ports. Once this parameter is disabled (unchecked), make sure that "everything" (IP Phones, ShoreGear® Switches, ShoreTel Server, Distributed Voice Mail Servers / Remote Servers, Conference Bridges and Contact Centers) is "fully" rebooted – this is a "one time only" item. By not performing a full system reboot, one-way audio will probably occur during initial testing.

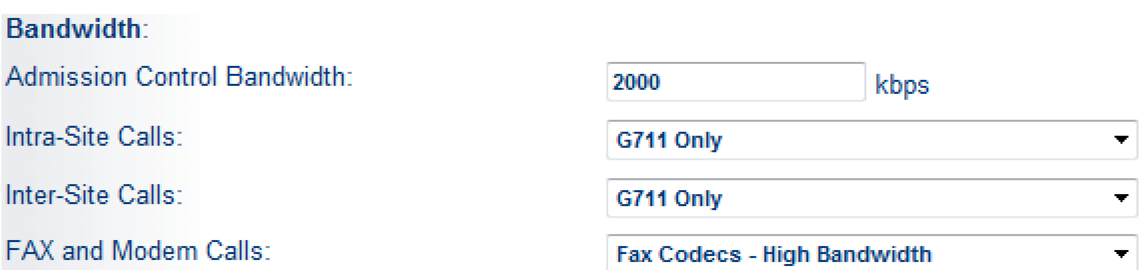
SITES SETTINGS

The next settings to address are the administration of sites. These settings are modified under the ShoreTel Director by selecting "**Administration**", then "**Sites**" (**Figure 6**).

Figure 6 – 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**" and "**Intra-Site / Inter-Site Calls**" parameters (**Figure 7**).

Figure 7 – 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 Fusion SIP Trunking. See the *ShoreTel Planning and Installation Guide* for more information.

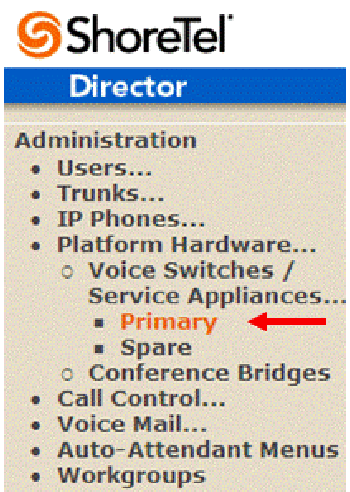
Sites Edit screen – Intra / Inter-Site Calls

By default, ShoreTel 14.x has 12 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 have created custom codec list to only contain PCMU/8000 codec. 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

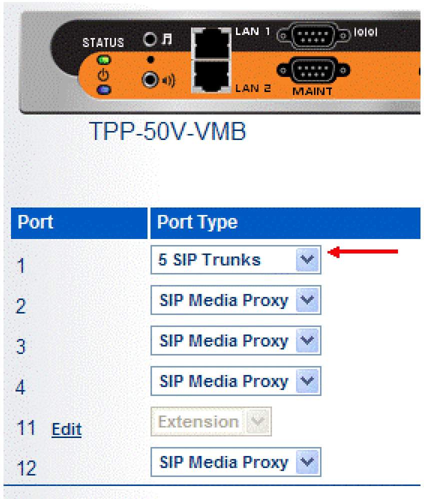
The final general settings to input are the ShoreGear switch settings. These changes are modified by selecting "**Administration**", then "**Platform Hardware…**", then "**Voice Switches / Service Appliances…**" followed by "**Primary**" in ShoreTel Director (**Figure 8**).

Figure 8 - 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 9**).

Figure 9 - 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 Fusion 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.

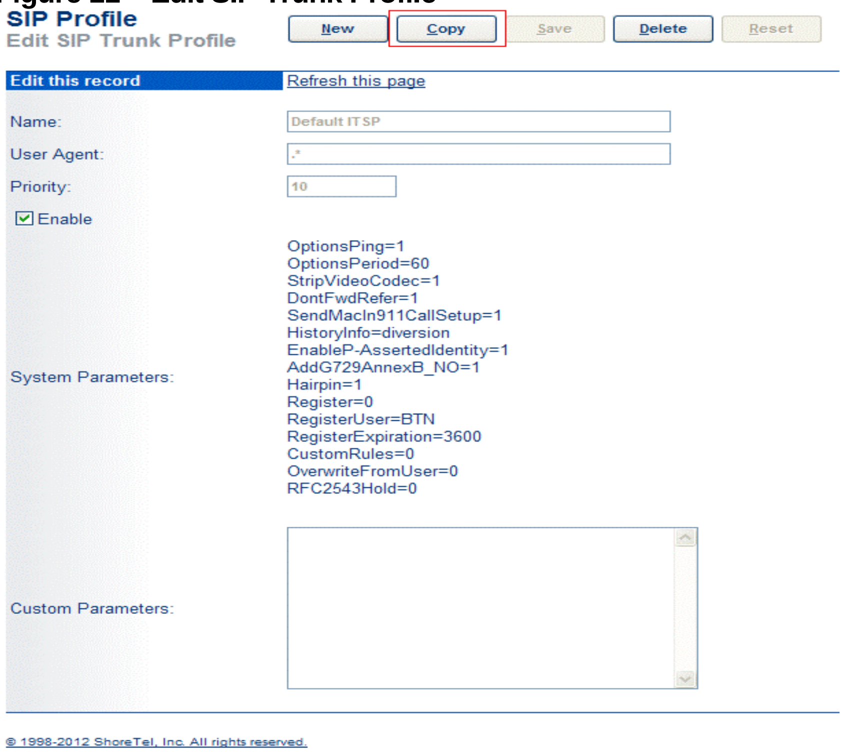
SHORETEL SYSTEM SETTINGS – SIP PROFILES

You will need to create a new SIP Trunk Profile for use with Fusion SIP Trunking, we recommend that you make a copy of an existing SIP Trunk Profile. From ShoreTel Director select "**Administration**", then "**Trunks…**" followed by "**SIP Profiles**" (**Figure 10**).

Figure 10 – Administration SIP Profiles

This action brings up the SIP Trunk Profiles page, select the "Default ITSP" SIP Profile. This will bring up the "Edit SIP Trunk Profile" page, select the "**Copy**" button (**Figure 11**).

Figure 11 – Edit SIP Trunk Profile



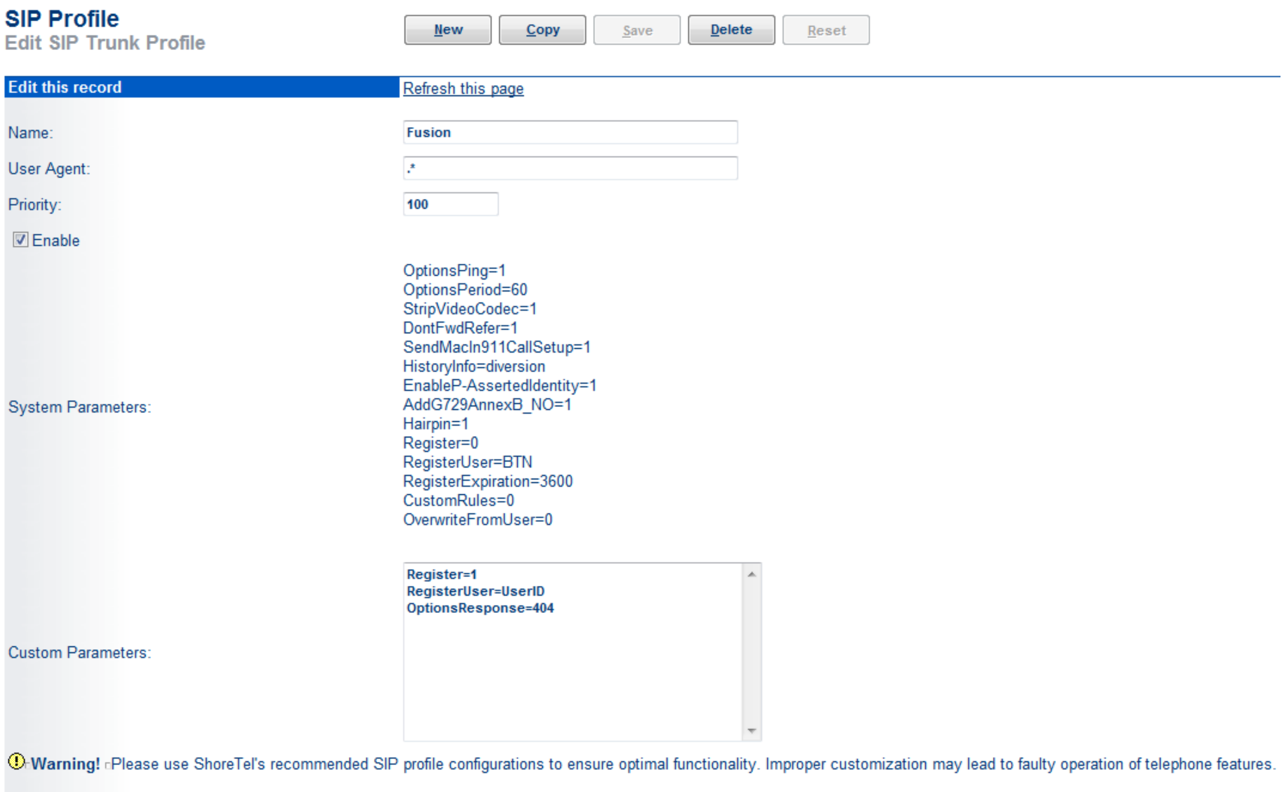
This action brings up a "Copy of Default ITSP" SIP profile, change the name to something else. In this example, we used name "Fusion". Scroll down to the "Custom Parameters:" section and enter the following parameters:

Register=1
RegisterUser=UserID
OptionsReponse=404

These parameters are case sensitive. After entering customer parameters, please select "**Save**" at the top of the page. The above given parameters helps with enabling Registration on ShoreTel system as well accept

404 as a valid response for SIP OPTIONS message. These parameters are also shown in example given below in **Figure 12**.

Figure 12 – Edit SIP Trunk Profile - II



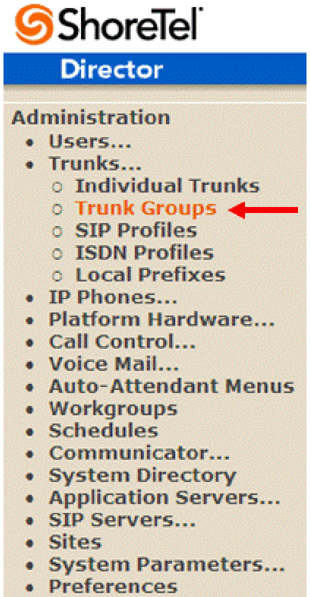
SHORETEL SYSTEM SETTINGS – TRUNK GROUPS

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

In trunk planning, the following needs to be considered.

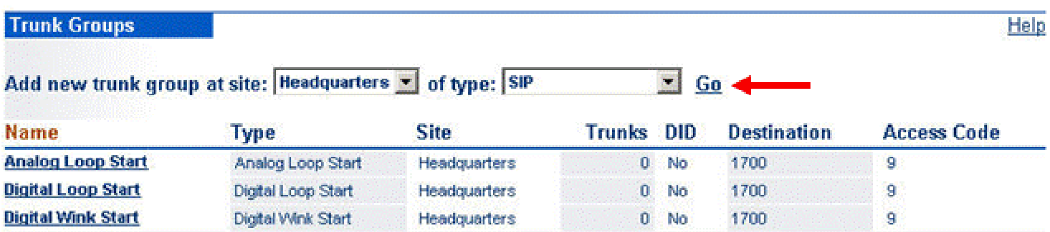
The settings for Trunk Groups are changed by selecting "**Administration**", then "**Trunks**" followed by "**Trunk Groups**" within ShoreTel Director (**Figure 13**).

Figure 13 - Administration Trunk Groups



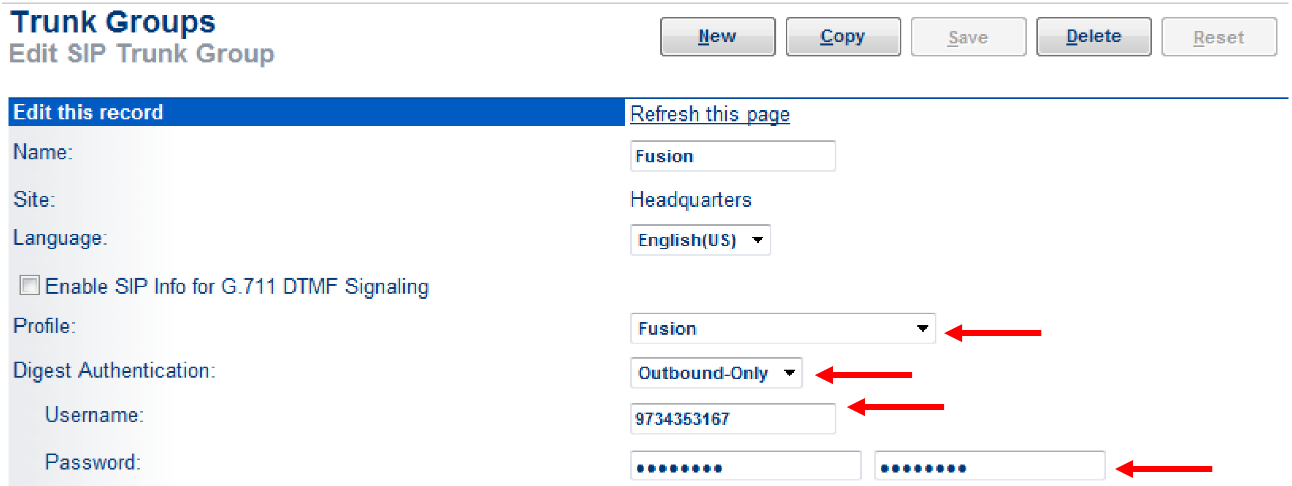
This selection brings up the “Trunk Groups” screen (**Figure 14**).

Figure 14 - 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 15**).

Figure 15 – 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 12, the name "Fusion SIP Trunking" has been created.

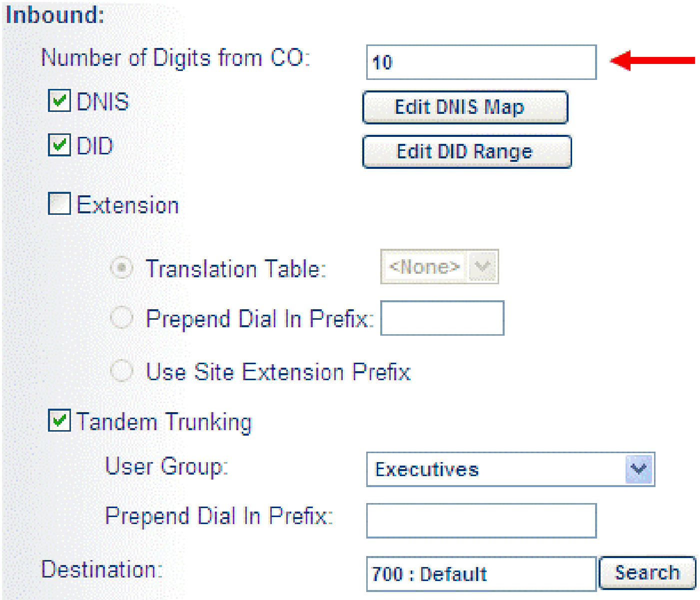
The "**Enable SIP Info for G.711 DTMF Signaling**" parameter should not be enabled (checked). Enabling SIP info is currently only used with SIP tie trunks between ShoreTel systems.

In the **"Profile:"** parameter, use the down arrow (pull-down menu) and select "Fusion" from the list, which was created earlier (see Figure 12).

The "**Enable Digest Authentication**" parameter defaults to "Outbound-Only" and enter username and password as provided Fusion SIP Trunking support.

The next item to change in the "Edit SIP Trunks Group" screen is to make the appropriate settings for the "**Inbound:**" parameters. (**Figure 16**).

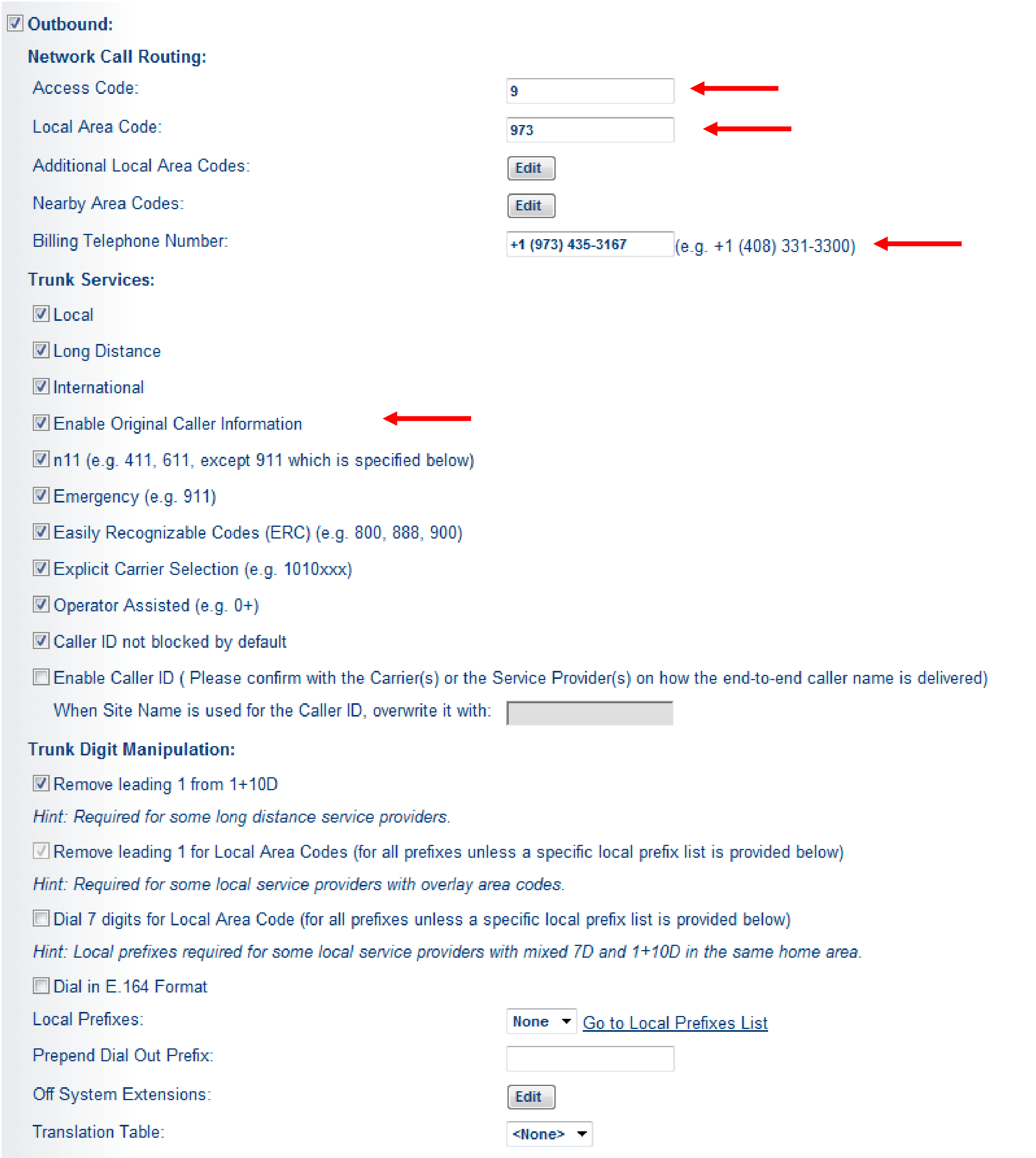
Figure 16 – Inbound



Within the "**Inbound:**" settings, ensure the "**Number of Digits from CO:**" is configured to a value of "10", this is the number of digits that the ShoreGear SIP trunk switch will be receiving from Fusion SIP Trunking. Enable (check) the "**DNIS**" or "**DID**" parameters as needed. It is no longer needed to enable the "**Extension**" parameter. We recommend that the "**Tandem Trunking**" parameter be enabled (checked) otherwise transfers to external telephone numbers will fail via SIP trunks, be sure to specify the proper "**User Group:**" that has access to the correct trunks. 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 17 – 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 Fusion SIP Trunking.

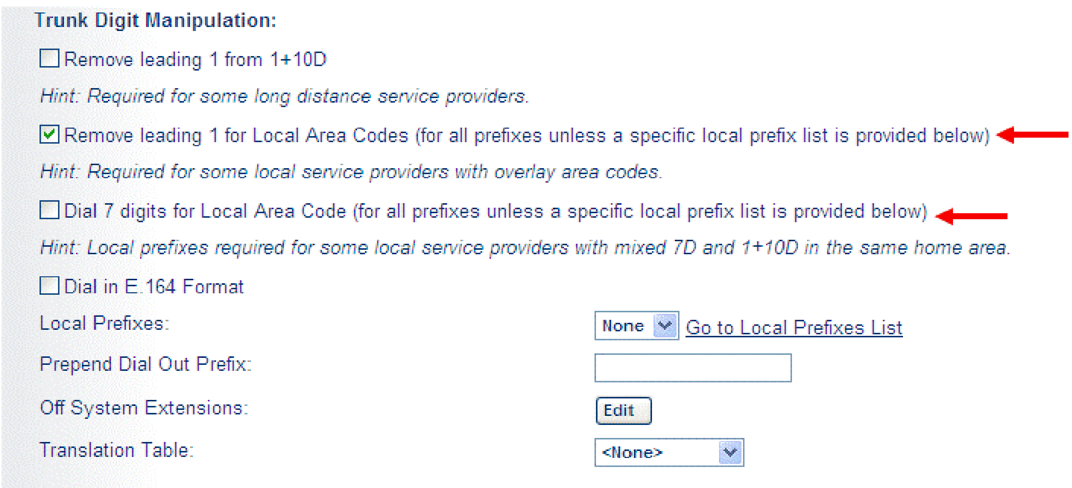
In the "**Trunk Services:**" area, make sure the appropriate services are enabled or disabled based on what Fusion supports and what features are needed from this Trunk Group. Please select checkbox **"Enable Original Caller Information"** to enable diversion header required for call forwarding scenario.

The last 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 18**):

Figure 18 – Trunk Digit Manipulation:



The only other parameters that require adjustment (from default) to interface with Fusion SIP Trunking are **“Remove leading 1 for Local Area Codes”** and “**Dial 7 digits for Local Area Code**”, enable (check) the parameter “Remove leading 1 for Local Area Codes” and disable (uncheck) the “Dial 7 digits for Local Area Code” 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 19**).

Figure 19 – ShoreTel Director Support Entry:

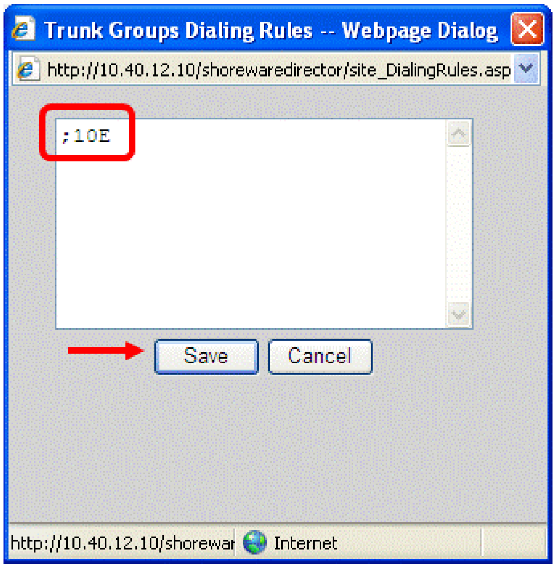
Log into ShoreTel Director with your normal administration user credentials.

Navigate to the "Edit SIP Trunk Group" page, by selecting "**Administration**" followed by "**Trunks…**", then "**Trunk Groups**" (as noted above in **Figure 10**), then in the "Trunk Groups" page, select the Trunk Group you created for Fusion (see **Figure 12**). 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 20**.

Figure 20 – Trunk Group Dialing Rules:

This action brings up the "Trunk Groups Dialing Rules – Webpage Dialog" as noted below in **Figure 21**.

Figure 21 – 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 21**.

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.

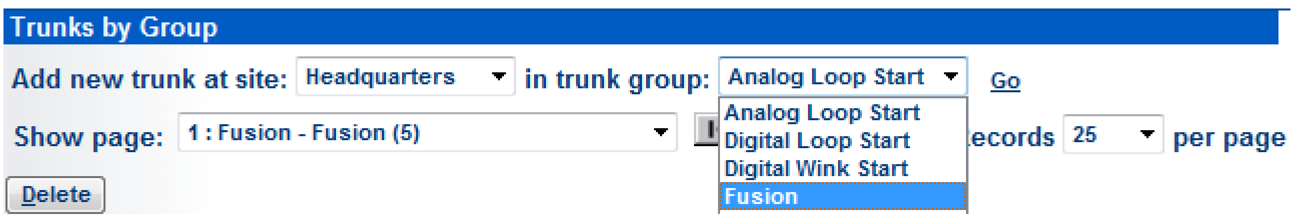
SYSTEM SETTINGS – INDIVIDUAL TRUNKS

This section covers the configuration of the individual trunks. Select "**Administration**", then "**Trunks**" followed by "**Individual Trunks**" to configure the individual trunks (**Figure 22**).

Figure 22 – Individual Trunks

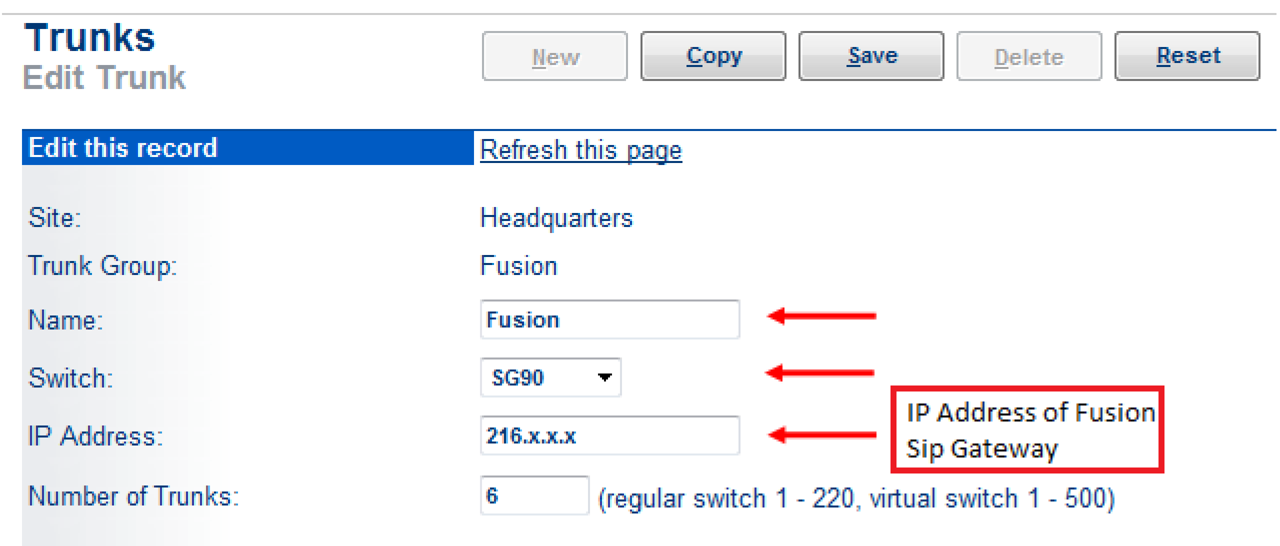
The "**Trunks by Group**" screen that is used to change the individual trunks settings then appears (**Figure 23**).

Figure 23 – 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 "Sunnyvale TPP Lab" and the trunk group is "Fusion SIP Trunking", as created above, see **Figure 12.** Click on the "**Go**" button to bring up the "Edit Trunk" screen (**Figure 24**).

Figure 24 - 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 Fusion 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 Fusion SIP Trunking platform, you will need to add Fusion’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 25**.

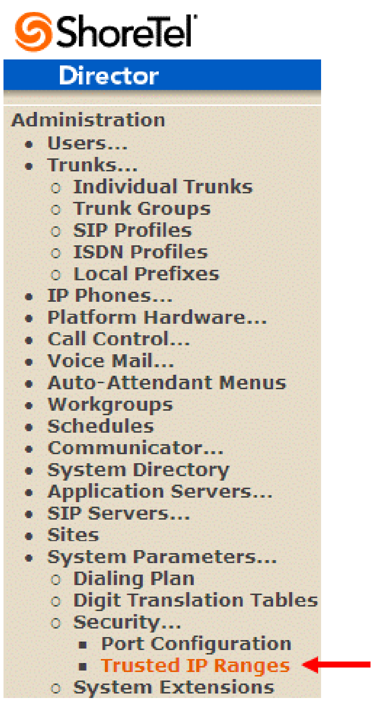


Figure 25– Trusted IP Ranges

This action causes the Trusted IP Ranges page to appear. Select the “**New**” button, as shown below in **Figure 26**.

Figure 26– Trusted IP Ranges Page

This action causes the "Trusted IP Range Info" pop-up window to be displayed, as shown below in **Figure 27**.

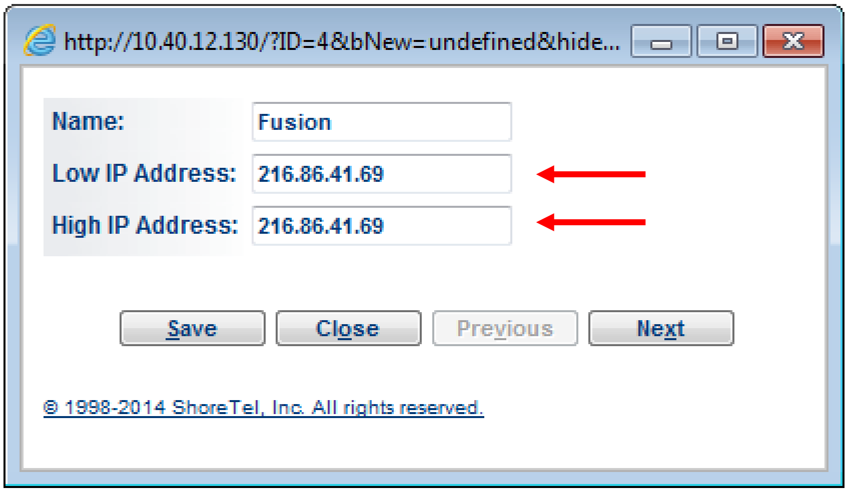


Figure 27 – Trusted IP Range Info Pop-up

Define a name, we chose "Fusion SIP Trunking", then in the "**Low IP Address:**" and "**High IP Address:**" define the Fusion Media Gateway IP address. In our example the Fusion Media Gateway IP address is 64.199.64.222, you can verify the IP address with Fusion. Once you have completed defining the values, select the "**Save**" button.

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

Fusion Configuration & Support

Fusion 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:

Fusion Sales and Customer Support: 888-301-1721
Fusion SIP Trunking Technical Support: technicalsupport@fusionconnect.com

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