

Mitel- SIP CoE

Technical

Configuration Notes

How to Guide to Configure
the Mitel 5000 CP 4.0 R24
for use with Integra Telecom



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Mitel Technical Configuration Notes - Configure the Mitel 5000 Communications Platform for use
with Integra SIP Service Provider
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Overview


This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel 5000 Communications Platform to connect to Integra. The different devices can be configured in various configurations depending on your VoIP solution. This document covers a basic setup with required option setup.

Interop History

Version	Date	Reason
1	October, 2010	Initial Interop with Mitel 5000 R24* and the Integra

Interop Status

The Interop of Integra has been given a Certification status. This service provider will be included in the SIP CoE Reference Guide. The status Integra achieved is:

	<p>The most common certification which means Integra has been tested and/or validated by the Mitel SIP CoE team. Product support will provide all necessary support related to the interop, but issues unique or specific to the 3rd party will be referred to the 3rd party as appropriate.</p>
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Software & Hardware Setup

This was the test setup to generate a basic SIP call between Integra service provider and the MITEL 5000.

Manufacturer	Variant	Software Version
Mitel	MITEL 5000 CP	4.0 R29 *
Mitel	IP set 5340, 5330	Minet
Integra	AS of Oct. 18, 2010	

* Need T38 patch for T38 fax to be working properly

Tested Features

This is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through the SIP Service provider and their PSTN gateway, call holding, transferring, conferencing, busy calls, long calls durations, variable codec.	<input checked="" type="checkbox"/>
Automatic Call Distribution	Making calls to an ACD environment.	<input checked="" type="checkbox"/>
NuPoint Voicemail	Terminating calls to a NuPoint voicemail boxes and DTMF detection.	<input checked="" type="checkbox"/>
Teleworker	Making and receiving a call through the Integra to and from Teleworker extensions.	Not Supported
Fax	T.38 and G711 Fax Calls	<input checked="" type="checkbox"/>

- No issues found

- Issues found, cannot recommend to use

 - Issues found

Device Limitations and Known Issues

This is a list of problems or not supported features when Integra has a SIP trunk connected to the Mitel 5000 Communications Platform.

Feature	Problem Description
Variable Packetization	Mitel 5000 can only support ptime 20 ms and 30 ms, Integra can't support ptime over 20ms. TAR: 36148 Recommendation: In order to work properly with this service provider, suggest to set ptime 20 ms in both Mitel 5000 and Integra Service Provider.
CLIR	Mitel 5000 does not support CLIR feature at this moment. Recommendation: Contact Mitel for further information regarding this feature.
Session Timer	Mitel 5000 does not support Session Timer at this moment. Recommendation: Contact Mitel for further information regarding this feature.
Video Calls	Mitel 5000 Communications Platform do not support video calls at this time. Recommendation: Contact Mitel for further information regarding this feature should it be required.
PRACK	Mitel 5000 Communications Platform does not support PRACK at this time. Recommendation: Contact Mitel for further information regarding this feature should it be required.

Network Topology

This diagram shows how the testing network is configured for reference.

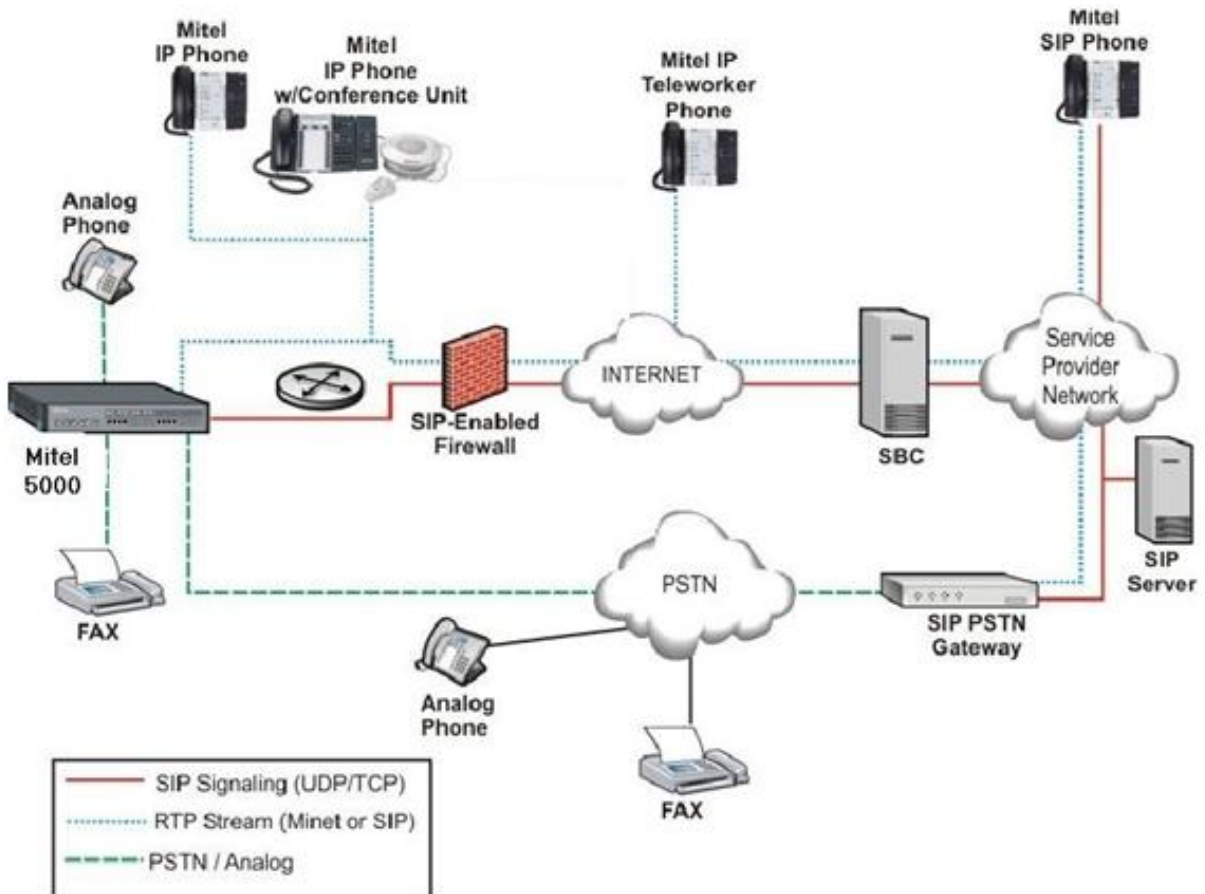


Figure 1 - Network Topology

Configuration Notes

This section is a description of how the SIP Interop was configured. These notes should give a guideline how the Mitel 5000 Communications Platform can connect to Integra sip service provider.

Disclaimer: Although Mitel has attempted to setup the interop testing facility as closely as possible to a customer premise environment, implementation setup could be different onsite. YOU MUST EXERCISE YOUR OWN DUE DILIGENCE IN REVIEWING, planning, implementing, and testing a customer configuration.

Mitel 5000 Communications Platform Configuration Notes

The following steps show how to program Mitel 5000 Communications Platform to interconnect with Integra Service Provider.

Network Requirements

- There must be adequate bandwidth to support the voice over IP. As a guide, the Ethernet bandwidth is approx 85 Kb/s per G.711 voice session and 29 Kb/s per G.729 voice session (assumes 20ms packetization). As an example, for 20 simultaneous SIP sessions, the Ethernet bandwidth consumption will be approx 1.7 Mb/s for G.711 and 0.6Mb/s for G729. Almost all Enterprise LAN networks can support this level of traffic without any special engineering. Please refer to the MCD 4.1 SP1 Engineering guidelines for further information.
- For high quality voice, the network connectivity must support a voice-quality grade of service (packet loss <1%, jitter < 30ms, one-way delay < 80ms).

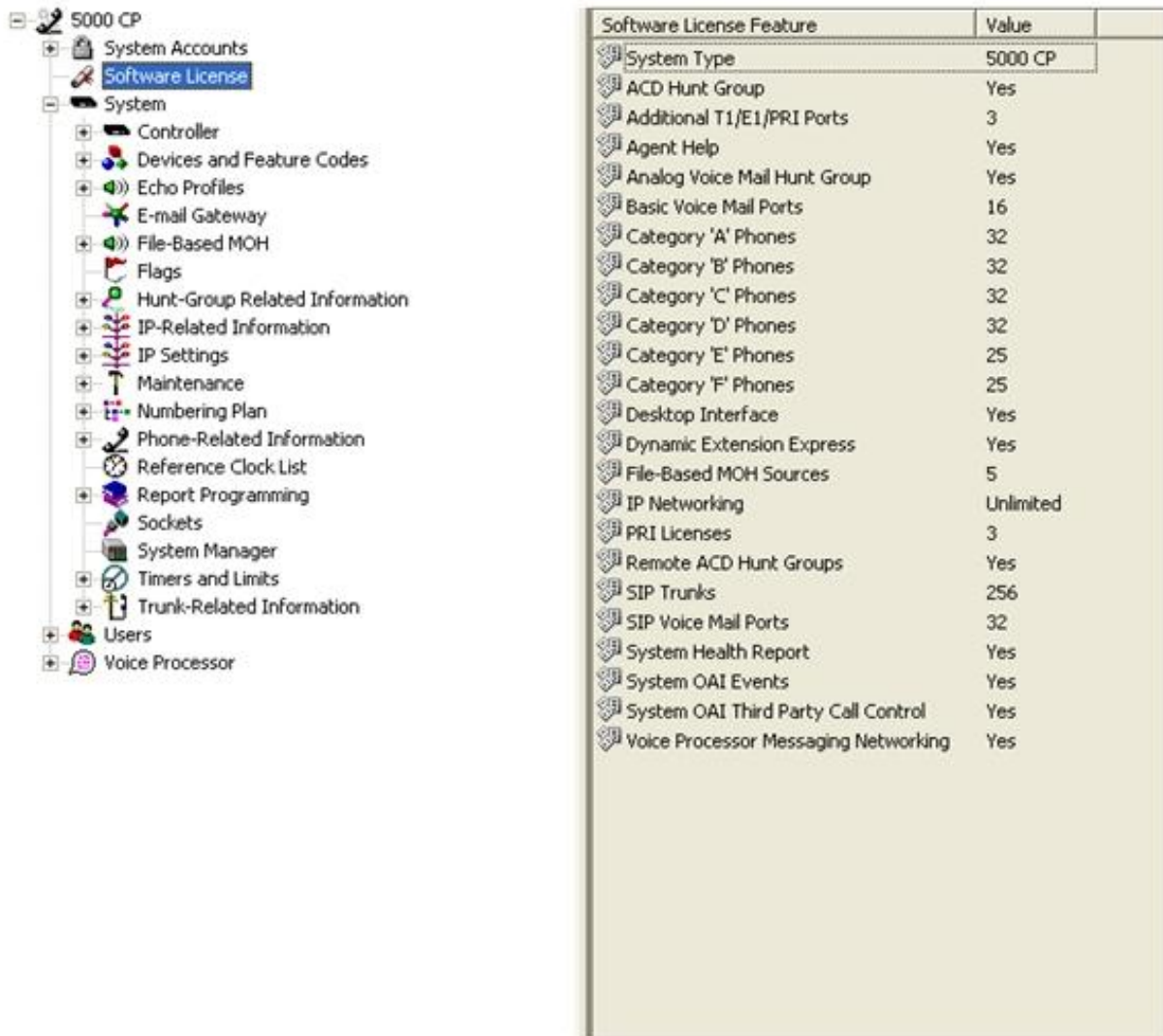
Assumptions for the Mitel 5000 Communications Platform Programming

- The SIP signaling connection uses UDP on Port 5060.

Licensing and Option Selection - SIP Licensing

Ensure that the Mitel 5000 is equipped with enough SIP trunking licences for the connection to Integra. This can be verified within the License and Option Selection form.

Enter the total number of licenses in the SIP Trunk Licences field. This is the maximum number of SIP trunk sessions that can be configured in the Mitel 5000 to be used with all service providers and applications.



Software License Feature	Value
System Type	5000 CP
ACD Hunt Group	Yes
Additional T1/E1/PRI Ports	3
Agent Help	Yes
Analog Voice Mail Hunt Group	Yes
Basic Voice Mail Ports	16
Category 'A' Phones	32
Category 'B' Phones	32
Category 'C' Phones	32
Category 'D' Phones	32
Category 'E' Phones	25
Category 'F' Phones	25
Desktop Interface	Yes
Dynamic Extension Express	Yes
File-Based MOH Sources	5
IP Networking	Unlimited
PRI Licenses	3
Remote ACD Hunt Groups	Yes
SIP Trunks	256
SIP Voice Mail Ports	32
System Health Report	Yes
System OAI Events	Yes
System OAI Third Party Call Control	Yes
Voice Processor Messaging Networking	Yes

Figure 2: Example of SIP Licensing

Creating and Configuring a SIP Peer Trunk Group

To support SIP trunks through a SIP trunk service provider, the SIP Trunk Groups folder has been added to the SIP Peers folder in DB Programming.

To create a SIP Trunk Group for Integra, you will need to right click in the right hand window panel of a SIP Trunk and then select “Create SIP Trunk Group”. (See Figure 3))

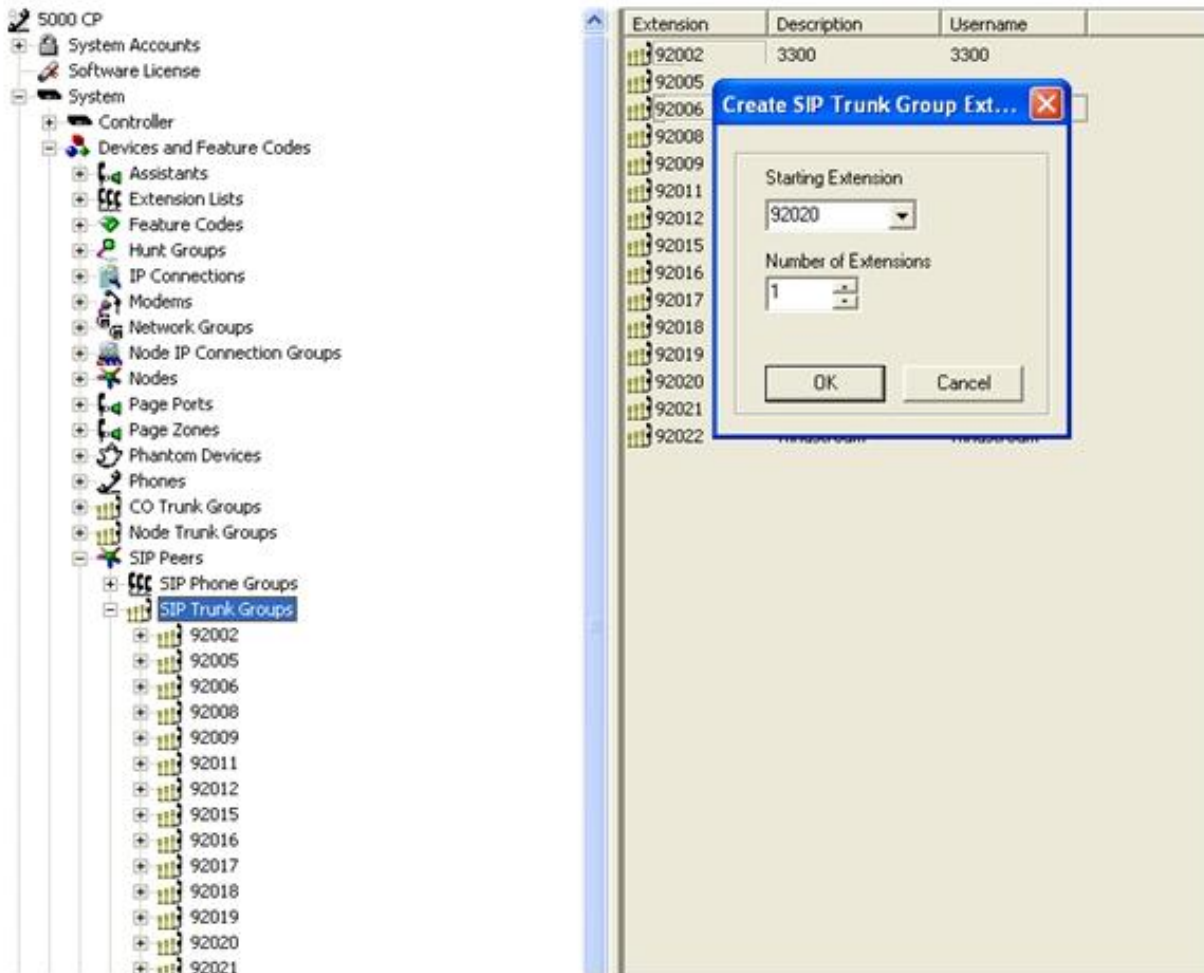


Figure 3: Example of Create SIP Trunk Group

When you create a SIP peer trunk group without using a template, such as Integra, you must obtain the necessary information from the SIP trunk service provider, and then configure this information in DB Programming

Program the Configuration folder as described below:

- **Registration:** If the SIP peer does not require registration, the fields in this folder do not need to be configured. The Enable Registration option is set to No by default and the remaining fields appear with a red “X”. Integra does require the Mitel 5000 system to register with the SIP peer network
- **Authentication:**
 - Username:* This field applies only if the SIP peer requires registration or call authentication.
 - Password:* This field applies only if the SIP peer requires registration or call authentication.Integra requires registration and call authentication.
- **Keep-Alive:** The Keep-Alive option keeps refreshing the NAT bindings for any Firewall/NAT in the path. It also helps in determining whether the SIP peer is reachable or not. It's recommended to be set to YES.
- **NAT Settings:** Specifies the NAT address type. The default is “No NAT or SIP-Aware NAT” (for systems that are using a SIP-aware firewall). If you are not using a SIP-aware firewall, you must change the setting to “Non SIP-Aware NAT”. For Integra, the default setting is used.
- **Alternate IP/FQDN List:** Some providers use multiple IP addresses to send SIP messages to the Mitel 5000. You must add All IP addresses or FQDNs other than the primary IP/FQDN to the list for all calls to be successful. For Integra, enter “default” for this field.
- **IP Address:** Indicates the IP address of the SIP peer trunk group. For Integra, it uses FQDN, keep it as default 255.255.255.255
- **Port Number:** Indicates the port that the system listens on the system for SIP peer messages. The range is 0-65535. For Integra, the listening port is 5060
- **Fully Qualified Domain Name:** Indicates the domain name of the SIP peer trunk group. Integra uses FQDN proxy1.integravoip.net.
- **Call Configuration:** Clicking **Call Configuration** takes you to the Call Configuration folder, it's set to 2 for Integra.
(System\IP-Related Information\Call Configurations*call configuration number*>).
- **Operating State:** Indicates the operating state of the SIP peer.
- **Maximum Number of Calls:** Indicates the maximum number of concurrent calls that are permitted towards the SIP peer. It is set to 5 for Integra sip trunk. DB Programming restricts this field based on the number of the SIP Trunks and SIP trunk licenses.
- **Use ITU-T E.164 Phone Number:** If set to Yes, the Mitel 5000 handles ITU-T E.164 formatted phone numbers as part of the incoming SIP INVITE messages from the SIP peer. Integra does require E.164.

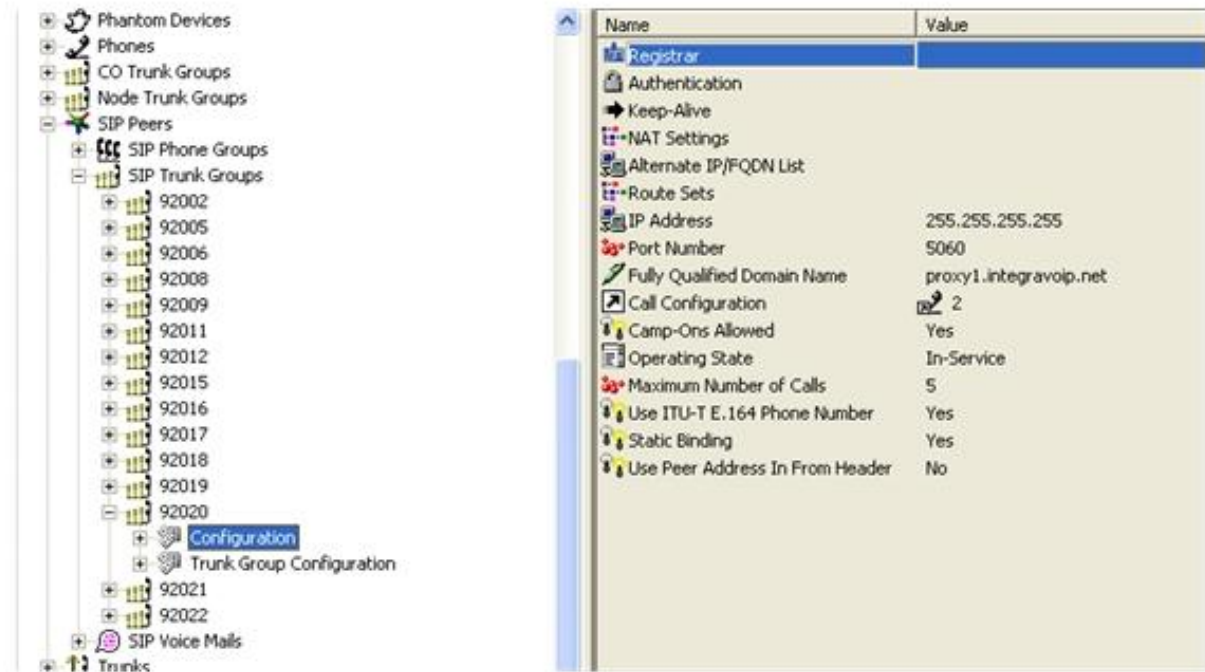


Figure 4: Example of Creating and Configuring a SIP Trunk

Integra requires the Mitel 5000 system to register with the SIP peer network. Integra provides information to you that must be programmed in DB Programming. Do the following:

- Contact Integra to obtain a user name and password to authenticate the account.
- Create a SIP peer trunk group in DB Programming to initiate registration with Integra
- Type the user name and password in DB Programming to provision the account information.

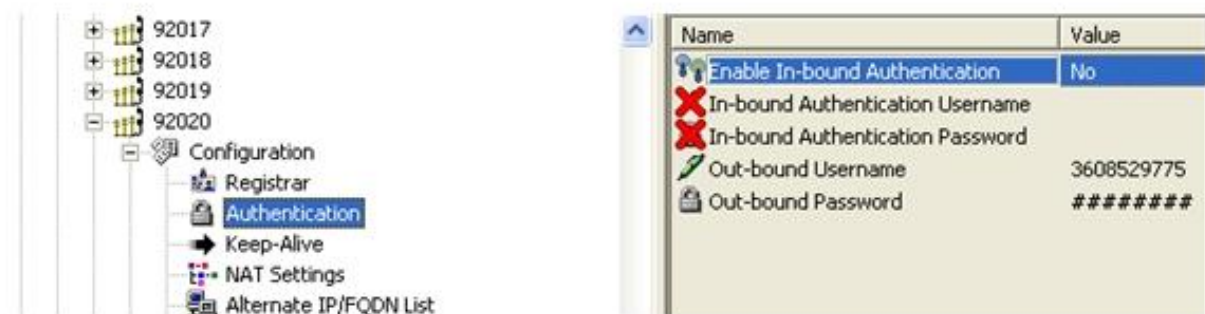


Figure 5: Configuration of a SIP peer Trunk Groups Registration Info

Programming the Trunk Group Configuration Folder

To program the Trunk Group Configuration folder:

1. Create the SIP peer trunks as follows:

- Double-click **Trunks**.
- Right-click the right pane, and the select **Create SIP Peer Trunk**. The Create SIP Peer Trunk Extension dialog box appears.
- Select the extension number you want to use for the item in the Starting Extension field. The recommended range is 94001-94999;
- Indicate the number of extensions you want to create in the Number of Extensions field. If the system is set to have more than one extension, the new trunks are assigned sequentially to the next available numbers.
- Click **OK**. For the Integra, 5 extensions were created, See Figure 6. The number of SIP peer trunks is restricted by the number of available SIP Trunks licenses.

Extension	Trunk Group	Label	Type
94004	92020	Undefined	SIP Peer Trunk
94005	92020	Undefined	SIP Peer Trunk
94006	92020	Undefined	SIP Peer Trunk
94017	92020	Undefined	SIP Peer Trunk
94018	92020	Undefined	SIP Peer Trunk

Figure 6: Example of SIP Trunks

2. See Figure 7, and then refer to the DB Programming Help for trunk programming (System\Devices and Feature Codes\SIP Peers\SIP Trunk Groups*trunkgroup*\Trunk Group Configuration) and details about these fields.

Audio for Calls Camped onto this Device: it defines the audio that a caller camped onto the device hears. The default value is 5000 CP. Other options include Silence, Tick Tone, Ringback, and File-Based MOH.

Music on hold: this option that will be heard by outside callers on the trunks in the trunk group can be set for music, tick tones, or silence.

Audio on Transfer to Ring: this option defines what outside callers will hear after their call is transferred, it can be set for Music-on-hold, Ringback, Silence, Tick Tone, 5000 CP, use Next Device's Audio Source, or File-Based MOH. For Integra, File-Based MOH is selected.

Audio on Transfer to Hold: this option defines what outside callers will hear while their call is being transferred to hold, it can be set for Music-on-hold, Ringback, Silence, Tick Tone, 5000 CP, use Next Device's Audio Source, or File-Based MOH. For Integra, File-Based MOH is selected.

Audio on Hold for Transfer Announcement: this option determines what outside callers will hear while their call is on transfer hold, after the transfer is completed, the caller will hear the Audio On Transfer to Ring (or Hold) selection. This options can be set for Music-on-hold, Ringback, Silence, Tick Tone, 5000 CP, use Next Device's Audio Source, or File-Based MOH. For Integra, File-Based MOH is selected.

Day / Night ring-in type: Ring-in type is determined separately for day and night modes, it can be the same for both modes or the combination. For Integra, "Single " was used to route the call to a device.

Propagate Original Caller ID: This flag allows the system to pass the caller ID name or number on an outgoing ISDN call if the call has not been answered by the system or for transfer announcement calls. This field is intended for customers that want to route incoming calls from the 5000 CP back to the PSTN via ISDN lines.

Calling Party Number: Each phone can be programmed to send an identifying number when a call is placed. This is called the "calling party number".

Calling Party Name: This field is similar to the existing calling party number field, it is used only for ISDN calls to the PSTN network.

Force Trunk Group Calling Party Name and Number: When selected, the system uses the trunk group calling party name and calling party number. When cleared, the system follows Caller ID forwarding.

Name	Value	Extended Value
Trunks		
Multiple Ring-In		
Emergency Outgoing Access		
Outgoing Access		
Toll Restriction		
Audio for Calls Camped onto this Device	5000 CP	
Music-On-Hold	File-Based MOH	1
Audio on Transfer to Ring	File-Based MOH	1
Audio on Transfer to Hold	File-Based MOH	1
Audio on Hold for Transfer Announcement	File-Based MOH	1
Day Ring-In Type	Single	10220
Night Ring-In Type	Single	10220
Send Station Extension/Username to Attached PEX	Yes	
Propagate Original Caller ID	Yes	
Calling Party Name		
Calling Party Number	13608529775	
Force Trunk Group Calling Party Name and Number	Yes	

Figure 7: Example of SIP Trunk Group Trunk Group configuration

IP Call Configurations

Call configurations define the settings that IP endpoints and gateways use when connected to calls. You can assign multiple devices to a specific call configuration.

By default, all IP devices are placed in Call Configuration 1, which is programmable. You do not need to add SIP endpoints to Call Configurations, because these devices negotiate call configurations before establishing a connection. You can program up to 25 different Call Configurations. Call configuration 2 is used for Integra sip trunking.

For T.38 fax to be working properly, in the call configurations folder, set option “Fax Encoding Setting (fax Transmission)” to T.38, set option “Fax Detection Sensitivity” to a Non Zero value, in Integra T.38 fax testing, the value was set to 1. Note: the default value for “Fax Detection Sensitivity” is 0.

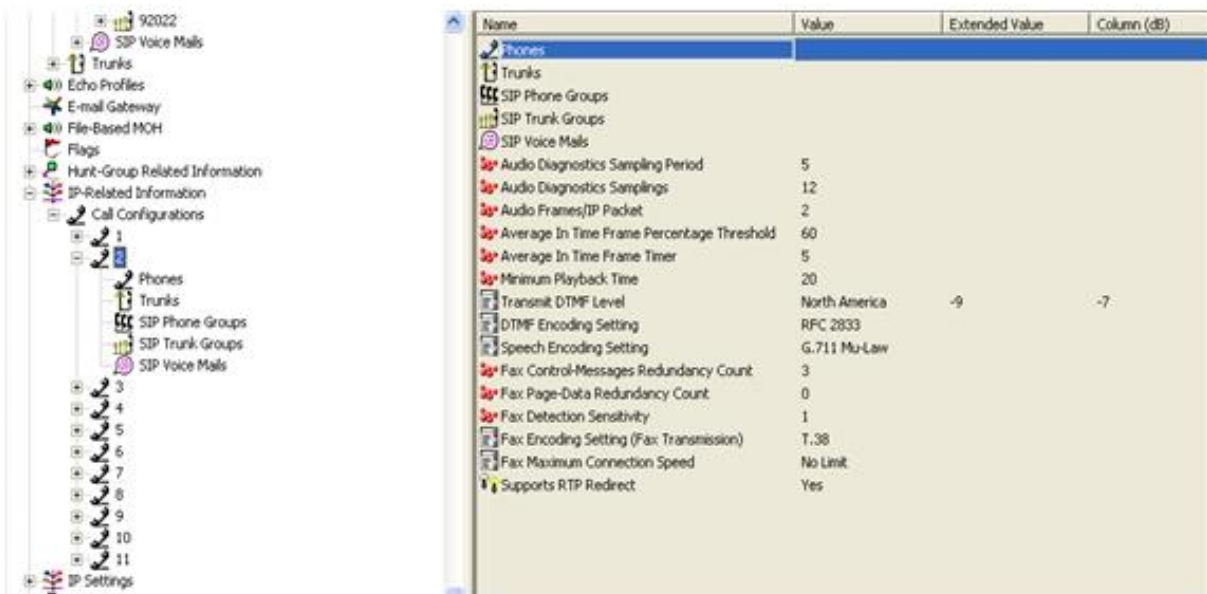


Figure 8: Call Configuration Options

To view a list of IP endpoints that are currently assigned to the call configuration:

- Select System - IP Related Information - Call Configurations - **Local** (or **Remote**).
- Double-click **Endpoints**.
-

To view IP trunks that are currently assigned to a Call Configuration:

- Select System - IP Related Information - Call Configurations - **Local** (or **Remote**).
- Double-click **Trunks**.

Summary Table:

[TemplateInfo]

ExportTemplateName=ExportTemplateSIPTranslator

ExportedFromVersion=4.0.3.67

ExportedDate=[2010-10-19 13:10:34]

[IPCallConfigurations]:PROFILE:call_configuration_id

description=8:Remote

audio_frames_ip_packet=17:2

transmit_dtmf_level=17:0

dtmf_level_column=2:-7

dtmf_level_row=2:-9

dtmf_encoding_setting=17:5

echo_suppression=11:TRUE

echo_suppression_sensitivity_level=2:2

echo_saturation_blocker=11:TRUE

minimum_playback_time=2:20

speech_encoding_setting=17:0

frame_percentage_threshold=17:60

frame_timer=17:5

digital_gain_level=17:0

audio_diagnostics_sampling_period=2:5

audio_diagnostics_samplings=2:12

fax_encoding_setting=17:9

fax_detection_sensitivity=3:1

fax_encoding_setting_detection=17:5

fax_data_page_redundancy_count=17:0

fax_control_data_redundancy_count=17:3

fax_maximum_connection_speed=17:0

supports_rtp_redirect_flag=11:TRUE

[SIPTranslators]

ip_address=8:255.255.255.255

listening_port=3:5060

fqdn=8:proxy1.integravoip.net

call_configuration_id=17:2

call_failure_threshold=2:6

keep_alive_ping_enabled=11:TRUE

keep_alive_ping_interval=2:60

keep_alive_ping_failure_threshold=17:1

nat_address_type=17:0

use_e_164_format=11:TRUE

session_refresh_timer=2:0

sip_t1_timer=2:1000

sip_t2_timer=2:4000

sip_hold_sdp_type=17:0

message_waiting_indication_enabled=11:FALSE

require_registration=11:TRUE

registrar_ip_address=8:255.255.255.255

registrar_fqdn=8:

registrar_ip_port=3:5060

registration_interval=2:1800

registration_initial_delay_interval=2:60

registration_maximum_retries=2:5

update_method_supported=11:FALSE

min_session_refresh_timer=2:120

early_media_supported=11:TRUE

campon_flag=11:TRUE

sip_message_timer=2:16000

refer_method_supported=11:FALSE

refer_with_replaces_supported=11:FALSE

[COTrunkGroups]

send_station_cid_to_attached_pbx_flag=11:TRUE

propagate_orig_cid_for_unanswered_calls_flag=11:TRUE

calling_party_name=8:

calling_party_number=8:13608529775

force_cotg_calling_party_name_number=11:TRUE

[SIPAlternateIPAddressFQDN]:LIST

ip_address_fqdn=8:192.168.101.205

