OpenScape Business

How to: configure SIP Trunk for Verizon IP Trunk Services

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Table of History

Date	Version	Changes
10/05/2013	1.0	Converted to Unify Template
09/10/2024	1.1	Editorial changes

Note: The basis for this document is the current OpenScape Business at the time of certification. Since OpenScape Business is constantly developed, input masks and interfaces as well as requirements may change in the future. The settings and entries described here then apply accordingly.

Introduction

This application note highlights the use and setup for OpenScape Business with Verizon IP Trunks and Verizon IP Contact Center Trunk Services (IPCC). OpenScape Business became generally available on July 13, 2012. Verizon has certified the OpenScape Business for compatibility in accordance with their US Retail VoIP, EMEA Retail VoIP and IPCC Interoperability test plans.

A Note on Standards Compliance

Due to interpretation, conformance with standards does not automatically imply that products will properly interoperate. It is absolutely necessary to perform interoperability testing to insure expected results. Verizon has performed interoperability testing and certifies that the OpenScape Business system meets Verizon support expectations when implemented following the provisioning outlined in this brief with exceptions where noted.

Implementations of OpenScape Business using alternative provisioning or other software version or alternate Session Border Control (SBC) elements must be locally tested to insure interoperability. Project level support for these non-certified elements can be requested for via the customer's Verizon Account team.

OpenScape Business Reference Architecture

OpenScape Business may be designed in modular fashion from simple single server applications through active/active dual processing designs using geographically distributed processing nodes for high availability. With OpenScape Business, OpenScape software may run on industry standard server hardware or within IT environments employing virtual machines.

NOTE: Verizon certification lab setup with Virtual Farm 1 and Transformed Site with disaster recovery.



Verizon IP Trunk Service Highlight

Verizon IP Trunk Services simplify network management and drive operational efficiencies by enabling the convergence of voice and data traffic on the same access connection. Verizon provides native SIP trunks directly to OpenScape Business solution over Private IP or Internet Dedicated Access facilities. Due to the extensibility of the Verizon VoIP network, now OpenScape Business customers can consolidate suppliers and obtain local exchange services using Verizon IP Trunks.

Verizon IP trunks can be provisioned to provide outbound calls and direct inward dial (DID) calls.

Verizon Burstable Enterprise Shared Trunks (BEST) - Verizon's BEST is an IP trunk service billing feature that allows pooling of IP trunk sessions for multiple site customers. BEST services are applicable where Verizon IP trunks are delivered at each customer site vs. a central or regional trunk deployment model. BEST is an industry first and allows the customer to take advantage of IP trunk traffic engineering at the enterprise level. Traditional trunk services and competitive IP trunk service sessions are normally, engineered for peak calling times for each customer site. With Verizon BEST enabled, the customer's IP trunk sessions can be combined into an enterprise view which can result in significant reduction of IP call sessions (and costs) due to the typical over-subscription. No special OpenScape Business provisioning is required to take advantage of Verizon BEST features.

Verizon VoIP Enterprise Routing (VIPER) - Verizon's VIPER feature for IP trunks eliminates domestic and international per minute calling charges for business-to-business calls made between Verizon VoIP VIPER customers in the U.S. and Europe. Because the new service is enabled on the Verizon network, customers don't have to deploy any additional software or hardware. Customers only need to have VIPER feature enabled on their IP trunks to take advantage of free calling to other VIPER enabled accounts (no special OpenScape Business provisioning is required).

OpenScape Business Configuration – Verizon IP Trunks

This section will outline the steps for the Configuration of the Verizon SIP Trunks with the OpenScape Business system. The configuration requires the creation or modification of the database within the OpenScape Business system.

The documented steps assume that the system administrator is a certified technician on the OpenScape Business platform.

The configuration assumes that the Routing information has been completed to allow the OpenScape Business system to access the internet as well as the Verizon Registration destination.

Configure the OpenScape Business System Information

This section will provide the configuration steps for programming the OpenScape Business system.

Open the OpenScape Business System Administration tool

Launch the OpenScape Business Administration tool from your web browser. Once the portal is opened enter the user name and password to open the main menu screen.



Configure the system Parameters

Under the Expert > Basic Settings> System Flags menu insure that the following flags are enabled

Basic Settings	System Flags	
▼System	Edit System Flags	
System Flags	voice mail voice call number.	
Time Parameters		
Display	Call Pickup after automatic recall:	
DISA	Configurable CLIP:	
Intercept/Attendant/Hotline	Caller list at destination in case of Forward Ling.	
LDAP	Call forwarding after deflect call / single step transfer	
Texts		
Flexible menu	Follow call management in case or deflect call / single step transfer:	
Speed Dials	Calling number in pick-up groups / ringing groups / CFN (RNA: 🛛	
Service Codes	SPE support:	
Gateway	SPE advisory tone:	
DynDNS		
Quality of Service	SIP Prov. to SIP Prov. transit.	
Port Management	Transparent dialing of * and # on trunk interfaces:	
Call Charges		
Voicemail / Announcement Player	Open numbering scheme	
	active:	
	Node call number:	
	CO features (transfer/conf./drop)	
	CO features:	
	Transit permission	
	Feature transit.	
	Tie traffic trapsit: 🗹	
	External traffic transit : 📝	
	Restriction for UC calls	
	Restriction for UC calls:	

Setting Up the General SIP Trunk Parameters

From the main menu select Expert > Telephony server >Voice Gateway> SIP Parameters

Voice Gateway	SIP Parameters	
SIP Parameters	Edit SIP Parameters	
Codec Parameters		
Destination Codec Parameters	SIP Transport Protocol	
Internet Telephony Service Provider	SIP via TCP:	Yes
Networking	SIP via UDP:	
SIP Interconnection	SIP via TLS:	Yes
	SIP Registrar	web to
	Period of registration (sec):	120
	RFC 3261 Timer Values	
	Transaction Timeout (msec):	32000
	SIP Session Timer	
	RFC 4028 support:	
	Session Expires (sec):	1800
	Minimal SE (sec):	90
	Provider Calls	
	Maximum possible Provider Calls:	0

Insure that the SIP transport protocol flags are set as displayed above

Insure that the SIP Session Timer flags are set as displayed below

Press the Apply key to save your changes

Select Expert > Telephony server >Voice Gateway> Codec Parameters and select "Edit Codec Parameters"

Insure that the Codec Flags, the T.38 flags, Misc. flags and RFC 2833 flags are set as displayed.

Voice Gateway	Codec Parameters				
SIP Parameters		Edit Codec Paramete	rs		
Codec Parameters		3			
Destination Codec Parameters	Codec	Priority	Voice Activity Detection	Frame Size)
Internet Telephony Service Provider	G.711 A-law	Priority 3 💌	VAD:		20 💌 mse
Networking	G.711 µ-law	Priority 2 -	VAD:		20 • mse
SIP Interconnection	G.729A	Priority 1 -	VAD:		20 💌 mse
	G.729AB	not used 💌	VAD: 🗹		20 - mse
	Max. UDP Data Error Cor	Use FillBitRemoval: agram Size for T.38 Fax (bytes): rection Used for T.38 Fax (UDP)	V 1472 t38UDPRedundancy		
	RFC2833	ClearChannel:		Frame Size: 20 💌 mse	c
	Transmission of Fax/Mode	em Tones according to RFC2833:			
	Transmission of DTN	AF Tones according to RFC2833:			
		Payload Type for RFC2833:	98		
	Redundant Transmission of RFC28	33 Tones according to RFC2198:			

After making the changes press the Apply button to save your changes

Set STUN Configuration Parameter

Select Expert > Telephony server >Voice Gateway> Internet Telephony Service Provider > Edit STUN Configuration

Select —Use static IP|| as the STUN Mode Enter your Public IP address information

Leave the Public SIP Port of 5060 in place

After making the changes press the Apply button to save your changes

Please note that this setting is required even though STUN is not required by Verizon

Voice Gateway	Internet Telephony Service Provider		
SIP Parameters	Add Internet Telephony Service Provider	Edit STUN Configuration	Detect NAT Type
Codec Parameters			
Destination Codec Parameters	STL	JN Mode: Use static IP 💽	
Internet Telephony Service Provider			
Networking	Public IP Address: 67.100.100.100		
SIP Interconnection			
	Public SIP Port: 5060		
	Detected 1	Nat-Type: Error: Cannot reach STUN server	or server does not exist!

Define Special Phone Numbers and Primary Line Seizure Route Group

This **Special Phone Numbers** form allows you define special telephone numbers such as 911. Please note that 9C911 is a default entry. Press Ok & Next to move to the next step.

	Special phone numbers	
s		
ise make sure that all special call humbers are su	pported by the selected provider without fail.	
Special phone number	pported by the selected provider without fail. Dialed digits	
Special phone number	pported by the selected provider without fail. Dialed digits 9C911	
Special phone number	pported by the selected provider without fail. Dialed digits 9C911	
Special phone number 1 2 3	pported by the selected provider without tail. Dialed digits 9C911	

The **Prioritization for Exchange Line Seizure form** is used to define the Primary route group that will be selected when a user dials 9 to place an outbound call. Please insure that the Verizon group is the first entry. Press Ok & Next to move to the next step.

	Prioritization for Exchange Line Seizure
Exchange Line Seizure	Trunk Access Code 9
Prioritization for Exchange Line Se Try to get 'Outside line Seizure'	izure
First over	Verizon1 💌

The **Status for the ITSP** screen below will display the registration status of the Verizon ITSP. The orange indication indicates that a connection has not been established to the ITSP After revising the ITSP profile and adding the Internet Telephony numbers you will be able to redisplay the status to confirm the connection status of the group. Continue to press the Ok &Next button until the Finish button appears. Press the Finish button to complete this portion of the programming.

	Status for the Internet Telephor	ny Service Provider (I
Provider		User
Cbeyond	Disabled	
CenturyLink 1	Disabled	
CenturyLink 2	Disabled	
COLT UK & Europe	Disabled	
COLT VPN	Disabled	
Skype Connect	Disabled	
SoTel	Disabled	
SoTel with register	Disabled	
Verizon1	Enabled	
Windstream	Disabled	
XO	Disabled	

Revise the Verizon ITSP profile Domain and Proxy information

The profile for the Verizon SIP trunks has already been loaded into the OpenScape Business and contains the low level parameter settings. The domain information, proxy information and STN settings will need to be entered.

Select Setup > Wizards > Central Telephony > Internet Telephony to display the list of approved ITSPs.

Select the Edit Button associated with the Verizon1 provider

p - Wizards - (Central Telephony - Interne	et Telephony	
		Provider configuration and ac	tivation for Internet Telephony
		No call via Internet: Country specific view:	United States of America 💌
	Activate Provider		Internet Telephony Service Provide
Add		Other Provider	
Edit	E.	Cbeyond	
Edit		CenturyLink 1	
Edit	and the second s	CenturyLink 2	
Edit	100	COLT UK & Europe	
Edit		COLT VPN	
Edit		Skype Connect	
Edit		SoTel	
Edit		SoTel with register	
Edit	V	Verizon1	

In the Domain Name field you will need to enter the Domain Name or IP address information that is received from Verizon. This is because there are no fixed-public servers for Verizon and a private VPN will be used for connection to the Verizon services.

In the Provider Proxy group you will need to enter the Domain Name or IP address information and the port ID that is received from Verizon. The typical port ID is 5060.

In the Provider STUN group click on the Use STUN check box and enter a $-{\rm dummy}$ IP address such as 10.10.10.10

Click on the Apply button to write the information to the data base.

Setup - Wizards - Central Telephony - Internet Telephony	
Internet Telephor	y Service Provider
Provider Name:	Verizon1
Enable Provider:	
Domain Name:	Enter DNS or IP add
Use Registrar:	
IP Address / Host name:	0.0.0.0
Port	0
Reregistration Interval at Provider (sec)	120
Provider Proxy	
Port	Enter DNS or IP add
Provider Outbound Proxy	1 <u></u> 1
Use Outbound Proxy:	
IP Address / Host name:	0.0.0.0
Port:	0
Use Inbound Proxy:	
IP Address / Host name:	0.0.0.0
Port	3478
Provider STUN Use STUN	Ø
IP Address / Host name:	
	Enter a place holder IP A

After inputting the Domain Name, the Proxy Provider information, press the Ok button to accept the changes and then press Ok& Next to move to the next form.

Add the Main Internet Telephony Station Number

Insert the Main Telephone number in the Internet telephony station field. Please confirm with Verizon that an authorization name and password are not required. Press Ok to accept your input

		Internet Telephony	Station for Verizon1
		Internet telephony station:	9727289402
		Authorization name:	
		Password:	
		Confirm Password:	
		Internet	Telephony Phone Numb
Add	9727289402		

Add the Internet Telephony Phone Numbers

This step is used to add the set of call numbers received from Verizon that will be used for direct inward dialing to tour stations and groups.

		Internet Telephony	Station for Verizon
		Internet telephony station:	9727289402
		Authorization name:	
		Password:	
		Confirm Password:	
		Internet	Telephony Phone Numb
Add	9727289403		
Delete	15616723124		
Delete	9727289402		

In the Internet Telephony Phone Numbers section, enter the number input as the Main Internet Telephony Station number and press the Add button. In the example above the first number that should be entered is 9727289402.

Continue the above process to enter the balance of the Internet Telephony Station Numbers.

Press the OK & Next button

Associate the Internet Telephony Phone Numbers with the system users and groups

This section provides the information for assigning the Internet Telephony Phone Numbers to the system users and groups.

Using the list box associated with each of the Internet Telephony Phone Numbers, select a station or Group as the target destination for the selected telephone number.

You may select one of the numbers as the default entry. The selected telephone number will be displayed for all outbound calls placed by stations that are not assigned an Internet Telephony Phone Number.

Setup - Wizards - Central Telephony - In	ternet Telephony		
	Call Number Assignmer	t for Verizon1	
So that an internal participant or members of a with 'Use as PABX number for outgoing calls'.	call group can telephone via Internet without an "Internet	Telephony Phone Number", the "	Internet Telep
Name of Internet Telephony Station	Internet Telephony Phone Number	Internal Call Number	Use a
9727289402	15616723124	-	0
9727289402	9727289402	100 Bill 💌	0
9727289402	9727289403	101 Gilligan 💌	0

After assigning all of your numbers press the Ok &Next button to advance through the balance of the forms until the finish button appears. Press the Finish button to complete the ITSP programming.

Select the Verizon ITSP profile

Select Setup > Wizards > Central Telephony > Internet Telephony to display the list of approved ITSPs.

Home Administrators	Setup Expert mode Data Backup License Management Service Center
etup	
Wizards	Central Telephony
Basic Installation	
Telephones / Subscribers	Entit Internet Telephony
Central Telephony	Access parameters of the Internet Telephony Service Provider (ITSP), e.g., user account, password,
User Telephony	SIP station number
UC Suite	Edit Voicemail Access numbers for integrated voicemail. Set up of voicemail boxes
	Edit Phone Book / Speed Dialing Set up central speed-dial destinations for the system's internal phone book
	Edit Call Detail Recording Set up call detail recording connection parameters for call detail applications
	Edit Music on Hold / Announcements Record new melodies and announcements for Music on Hold and announcement before answering

Uncheck the —No call via Internet box to display the ITSPs for the United States Enable the Verizon ITSP entry and press Ok & Next to move to the next step

Setup - Wizards -	Central Telephony - Interne	et Telephony
		Provider configuration and activation for Internet Telephony
		No call via Internet: 🔲 Country specific view: United States of America 💌
	Activate Provider	Internet Telephony Service Provide
Add		Other Provider
Edit		Cbeyond
Edit	[207]	CenturyLink 1
Edit		CenturyLink 2
Edit		COLT UK & Europe
Edit		COLT VPN
Edit		Skype Connect
Edit		SoTel
Edit		SoTel with register
Edit		Verizon1

Uncheck the —No call via Internet box to display the ITSPs for the United States Enable the Verizon ITSP entry and press Ok & Next to move to the next step

Define the number of concurrent voice sessions

The **Settings for Internet Telephony** form is used to define the number of concurrent voice sessions that will be supported. The maximum number of sessions supported by the X3, X5 and X8 is 60. The maximum number of sessions supported by the Business S system is 120. This assumes that the internet medium is capable of handling the anticipated traffic

Settings for inte	Settings for Internet Telephony						
imultaneous Internet Calls							
Please enter in field "Upstream up to (Kbit/sec)" the Upstream of your Internet connection comm Jostream up to (Kbps) = 512	unicated by your Provider. You have typed in						
n the 'Change Feature> Internet Telephony' Assistant. This upstream allows you to conduct u cad, you will need to reduce this number of simultaneous calls.	p to 4 Internet phone calls simultaneously. If the call qual						
 the 'Change Feature> Internet Telephony' Assistant. This upstream allows you to conduct u oad, you will need to reduce this number of simultaneous calls. The number of simultaneous Internet Calls also depends on the licensing. 	${\sf p}$ to ${\bf 4}$ Internet phone calls simultaneously. If the call qual						
n the 'Change Feature> Internet Telephony' Assistant. This upstream allows you to conduct u oad, you will need to reduce this number of simultaneous calls. The number of simultaneous Internet Calls also depends on the licensing. Upstream up to (Kbps):	p to 4 Internet phone calls simultaneously. If the call qual						

Assuming 128kbps per call enter the upstream kbps size to calculate the number of concurrent sessions. In the example above entering a value of 512 kbps resulted in 4 concurrent voice sessions

Press Ok & Next to move to the next step

Please note that each ISP session will require an OSBiz S2M/SIP Trunk license. The licenses will have to be enabled under the License Management > CO Trunks

Home	Administrators	Setup	Expert mode	Data Backup	License Management	Service (Center
License Man	agement						
License in	formation	co	Trunks				
UNZJVAFYV	QL4QDVMA7L3LX2						
Local User	licenses	Access	to the Central Offic	e via Internet telepho	ny is licensed by CO trunk licens	ses	
Overview					Available licenses for SIF	trunks: 246	
IP User		SIP tri	unks				
Mobility U	ser			The configure for eac	ed number of simultaneous Interr h Internet Telephony Service Pro	net calls wider is: 4	
Deskshare	User	-		License number of	simultaneous Internet calls in th	is node: 4	
CO Trunks			License d	emand for number of	simultaneous Internet calls in th	is node:	4 -
System Lice	nses		2.00.000				

Confirm the Trunk Route Settings

Under the Expert Mode > Telephony Server > Trunks & Routing > Route select the Verizon1 trunk group and then select Change Route to confirm that the Gateway Location information entered during the original system setup is correct and the defined Seizure entered as part of the Wizard is accurate

Expert mode - relephony se	iver.			
Trunks/Routing	Route			
Trunks	Change Route	Change Routing Parameters		Special Parameter change
Route				
ISDN		Route Name:	Verizon1	
Trk Grp. 2		Seizure code:	856	
route 3		Colzare code.		
route 4		CO code (2nd trunk code):		
route 5	Gateway Location			
route 6		Country code:	1	
IC Suite		Local area code:	772	
route 9		Local area code.	112	
route 10		PABX number:	287	
route 11	PABX number-incoming			
route 12		Country code:		
Verizon1		Local area code:		
route 14		Eboar area ebao.		
route 15		PABX number:		
Networking		Location number:		
	PABX number-outgoing			
	and the second	Country code:		
		country code.		
		Local area code:		
		PABX number:		
			1	
		Suppress station number:		
	Overflow route			
		Overflow route :	None 💌	
	Digit transmission			
		Digit transmission:	en-bloc sendina 💌	

Select the Change Route Parameters button to display the form. Confirm or revise your entries to match the form displayed below.

Franks Nouting	Route		
Trunks	Change Route	Change Routing Paramete	rs
Route			_
0500	Analysis	of second dial tone / Trunk monitoring:	8
Trk Grp. 2		Intercent per direction:	m
route 3		intercept per unection.	
route 4		Over. service 3.1 kHz audio:	
route fi		Add direction prefix incoming:	
route 6		Add direction prefix outgoing:	
mane r		Ringback tone to CO:	
route 9		IN activated	
results 10		Litt activated.	
roate 11		Segmentation:	yes
route 12		deactivate UUS per route:	
Vermont		Always use DSP:	8
route 14			
route 15 Networklova		Analog trunk seizure:	no pause 💌
		Trunk call pause:	Pause 6 s 💌
		Type of seizure:	linear 💌
		Route type:	CO 💌
		No. and type, outgoing:	Unknown
		Call number type:	Direct inward dialing

Press the Apply button after making the changes

Confirm number of SIP sessions have been added the selected ITSP group

Under the Expert Mode > Telephony Server > Trunks & Routing >Trunks > LAN section you can select the "Port – ITSP" that is associated with the configured provider to confirm the number of channels added in the wizard have been added to the Verizon group. After confirming the quantity is correct exit the form to return to the main menu.

Trunks/Routing	Trunks					
Trunks		add line				
▼LAN						
▼Box: 1, Slot: 1	Trunk	Box-SI-Pt-Li	Code	Route	Status	
Port 3 Networking	Line 47	LAN 1-0-8-1	7847	Verizon1	active	ITSP 2
Port 4 SIP Interconnection 1	Line 48	LAN 1-0-8-2	7848	Verizon1	active	ITSP 2
Port 5 SIP Interconnection 2	Line 49	LAN 1-0-8-3	7849	Verizon1	active	ITSP 2
▼Port 7 ITSP 1	Line 50	LAN 1-0-8-4	7850	Verizon1	active	ITSP 2
Port 8 ITSP 2						
◆7847 0-8-47						
●7848 0-8-48						
◆7849 0-8-49						
◆7850 0-8-50						

Least Cost Routing Dial Plan

This step is required to allow the station user to dial a PSTN telephone number and have the outbound call route over the selected Verizon SIP trunk group.

Under the Expert Mode > Telephony Server >LCR >Dial Plan selection the entries for dial 9 access to be assigned to a Routing Table. In the example below the route table selected is 2. Press the blue arrow to the right of the route table entry to display the route table content.

LCR	^	Dial Plan			
Classes Of Service					
Dial Plan	E				
Routing table		Dial Plan	Name	Dialed digits	Routing Tabl
1 - Table		1	Emergency call	9C911	6 ▼ →
2 - Table		2			6 ▼ →
3 - Table		3			6 -
4 - Table		4			
5 - Table		5			
6 - Table		6			
7 - Table		-			6 💌 🤿
8 - Table		1			<u>6</u> →
9 - Table		8			6 💌 —>
10 - Table		9			6 💌 →
11 - Table		10			6 ▼ →
12 - Table		11			$6 \rightarrow$
13 - Table		12			6 ▼ →
14 - Table		13			
15 - Table		14		-	
16 - Table		45			
17 - Table		15			<u>6</u> →
18 - Table		16	Standard	9CNXX-NXX-XXXX	2 ▼ →
19 - Table		17	Standard	9C1-NXX-NXX-XXXX	2 🔹 🔿

The information under Expert Mode > Telephony Server >LCR >Routing Table will be displayed. In the example below the Verizon1 group is primary selection. The dial rule is SIP and the minimum COS is 15.

In applications where the My Fax application will be used the Min Cos must be set to 1.

LCR	· ·	Routi	uting Table								
Classes Of Service Dial Plan	2	Rout			Change Routing	g Table		-	-		
Routing table	1		Pouting Table: 2 an bloc conding								
1 - Table					Routing	Table.2		en-bioc	senting		
2 - Table		Index	Route		Dial Rule	min. COS	Warn	ing	Dedi Gate	cated eway	GW Nod
3 - Table 4 - Table		1	Verizon1 💌	SIP	•	15 💌	None		No		
5 - Table		2	Trk Grp. 12 🔹	SIP	-	15 💌	None	•	No	•	

The Dial Rule may be confirmed under Expert Mode > Telephony Server >LCR >Dial Rule. In the example below the out dial rule A will echo all digits to the PSTN after the access code "9".

Expert mode - Telephony Se	rver					
LCR	p	ial Rule				
Classes Of Service Dial Plan			Change Dial Ru	Rule		
Routing table		Rule Name	Dial rule format	Network access	Туре	
Dial rule	1	CO	A	Main network supplie 💌	Unknown	
	2	SIP	A	Main network supplie 💌	Unknown	

The LCR Out dial rule is used to define the digit string that will be sent to the PSTN. The system administrator uses a set of command codes to configure how much and which portions of the number that was dialed.

A dial string is created using field separators between dial pattern groups. The separator is either the letter "C" that will return dial tone or the character "-".

For example in the Dial plan string 9C1-NXX-XXX-XXXX The "9" is the LCR access code and is field 1 The "C" is a separator and will return simulated dial tone to the user The "1" is the entry in field one The "-"is a separator The "NXX" is the entry for field two The "-" is a separator The "XXX" is the entry for field three The "-" is a separator The "XXX" is the entry for field three The "-" is a separator

The command codes are

"A" = dial the entire string after field one or after a specified ECHO field."EX" = Echo the digits from a specific field. i.e. E2 = Dial the digits in field 2"D" = Insert a string of digits within the output. i.e. D408A

The Out dial rule for the SIP trunk call will be

Rule Name = Dial SIP Rule Format = A (echo all digits after the LCR access code) Procedure = Main Network Provider TON = Unknown

High Level Troubleshooting OpenScape Business and IP Trunks

Refer to the OpenScape Business Service Manual, Service Documentation, for OpenScape Business trouble shooting steps. The latest service documentation maybe found via "New Company" Business Area on-line web portal (SEBA).

Additional Documentation References

OpenScape Business General Information

http://wiki.unify.com/wiki/OpenScape Business

OpenScape Business and SIP Provider Information

http://wiki.unify.com/wiki/OpenScape_Business#Supported_VoIP_Provider

Network Configuration for VoIP Providers

http://wiki.Unify.com/wiki/Network_Configuration_for_VoIP_Providers List of Acronyms

Acronym	Description	Acronym	Description
B2BUA	Back-to-Back User Agent	NCS	Network Based Call Signaling Protocol
CCBS	Call Completion to Busy Subscriber	NE	Network Element
CCNR	Call Completion on No Reply	NNI	Network-Network Interface
CLIP	Calling Line Identification Presentation	OCSP	Online Certificate Status Protocol
CLIR	Calling Line Identification Presentation Restriction	PBX	Private Branch Exchange
COLP	Connected Line Identification Presentation	PPPoE	Point to Point Protocol over Ethernet
COLR	Connected Line Identification Presentation Restriction	PSAP	Public Safety Answering Point
CRL	Certificate Revocation List	PSTN	Public Switched Telephone Network
DID	Direct Inward Dialing	QoS	Quality of Service
DN	Directory Number	RFC	Request For Comments
DNS	Domain Name System	RTP	Real-time Transport Protocol
DNS	Domain Name Server	SBC	Session Border Controller
DSCP	Differentiated Services Code Point	SDP	Session Description Protocol
DSL	Digital Subscriber Line	SIP	Session Initiation Protocol
DSLAM	Digital Subscriber Line Access Multiplexer	SLA	Service Level Agreement
DTMF	Dual-Tone Multifrequency	SP	Service Provider

ENUM	Telephone Number Mapping	SSNE	SIP Signaling Network Element
ETSI	European Telecommunication Standardization Institute	ТСАР	Transaction Capabilities Application Part (SS7)
FQDN	Fully Qualified Domain Name	ТСР	Transmission Control Protocol
GWY	Gateway	TISPAN	Telecommunications & Internet Converged Services Networking
IP	Internet Protocol	UA	User Agent
ISUP	ISDN User Part (SS7)	UAC	User Agent Client
LIN	Location Identification Number	UAS	User Agent Server
MG	Media Gateway	URI	Uniform Resource Identifier
MGC	Media Gateway Controller	VCU	Video Conference Unit
MGCP	Media Gateway Control protocol	VM MS	Voice Mail/Media Server
МТР	Message Transfer Part (SS7)	V-MG	Video Media Gateway
NAPTR	Naming Authority Pointer Records	XML	Extensible Markup Language