OpenScape Business V1

Tutorial

Configuration of Verizon IP Trunk Services for the OpenScape Business V1 system

Version 1.0

Definitions

How To

A How To describes the configuration of an feature within the administration of the system. It addresses primarily trained administrators.

Tutorial

Within the tutorials procedures for installation, administration and operation of specific devices, applications or 3rd party systems, which are connected to the system, are described. The tutorial addresses primarily trained administrators.

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Table of History

Date	Version	Modified by	Section Affected	Comments
09/18/2013	Initial version.01		All	
09/19/2013	Version .02		Section 5 Section 7.1 Section 11.1.6 Section 11.1.8	Added diagrams Added diagrams Comment on lic. Comments on STUN
09/19/2013	Version .03		Section 11.1.6	Added screen shot
09/20/2013	Version .04			Moved sections 11.1.5 & 11.1.6 to end of configuration steps. Deleted reference to Click to save button.
09/30/2013	Version .05			Replaced diagram in section 11.1.11
10/05/2013	Version 1.0			Converted to Unify Template

1. 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.

2. Who should read this document?

This brief is written for Resellers and Direct Channel support teams that are installing the Verizon SIP services on an OpenScape Business system.

OpenScape Business and Verizon IP Trunk Services

OpenScape Business is a powerful, reliable communication platform for every sector of industry. It offers you the variety of services of classic telephony, combined with state-of the-art solutions for Unified Communications. And all in one single, flexible and cost-saving configuration. As a modular communication platform,

OpenScape Business is able to satisfy the requirements of companies with stringent demands. It is a flexible and scalable solution that can be combined with an incredibly broad range of applications and features and coordinated with the individual requirements of your company. OpenScape Business is an innovative and flexible converged platform that perfectly adapts communications to the company structured medium-sized business. Whether your aim is to enhance growth or seamlessly integrate branch offices or mobile staff, the three expansion stages of the OpenScape Business are the perfect solution for optimizing costs and business processes

The OpenScape Business solution can provide optimum customer flexibility when combined with Verizon IP Trunk Services. The OpenScape Business is a small medium enterprise VoIP communication platform that may use Internet Session Initiated Protocol SIP based trunks or Private IP networks to connect IP or TDM terminals (voice or video), software clients to the PSTN for voice, fax and video applications. The system will support a maximum of 32 concurrent voice sessions that may be shared by up to 4 SIP trunk groups.

Nowadays, there are more network providers offering telephony services than ever before. As the de-facto standard for Internet telephony, induces Internet telephony service providers (ITSP) to provide attractive applications and business models. With its SIP interfaces, the OpenScape Business is able to take advantage of existing network services and drastically cut communication costs. OpenScape Business already supports new SIP options, including SIP phones or user and system connections for Internet telephony.

3. 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.

4. OpenScape Business Voice solution components

OpenScape Business is a high-performance, reliable communication platform for medium-sized enterprises. It offers the wide range of functionality of a traditional telephony system, coupled with modern UC communication solutions. The system is suited to both packet-switched (LAN/WAN) and line-switched (ISDN) environments, or a combination of the two. OpenScape Business enables any combination of IP, analog and digital telephones, as well as PC clients and cordless telephones.

OpenScape Business Model X8

OpenScape Business Model X8 Base cabinet

OpenScape Business Model X8 peripheral circuit cards for connectivity to TDM subscribers and trunk resources

Mains Power Cord, USA variant

OpenStage TDM and or IP telephones

OpenScape Business Model X5R

OpenScape Business Model X5 Base cabinet

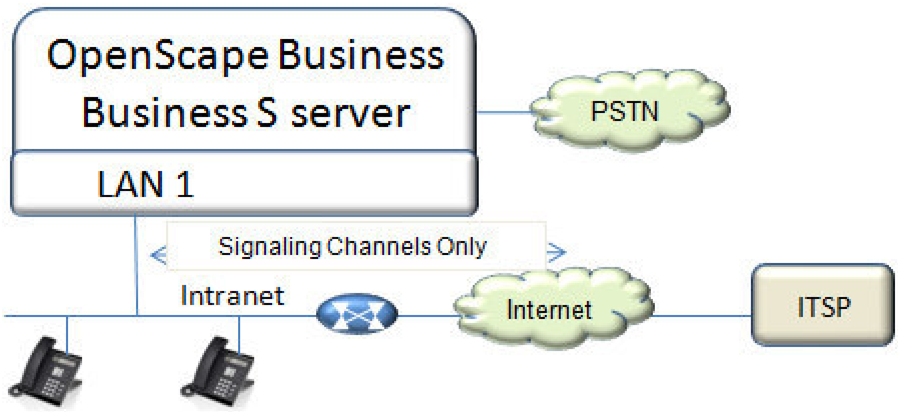
OpenScape Business Model X5 peripheral circuit cards for connectivity to TDM subscribers and trunk resources

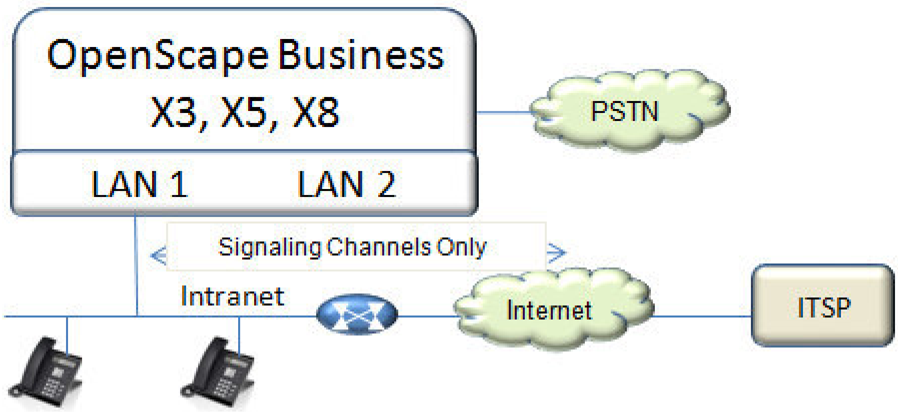
Mains Power Cord, USA variant

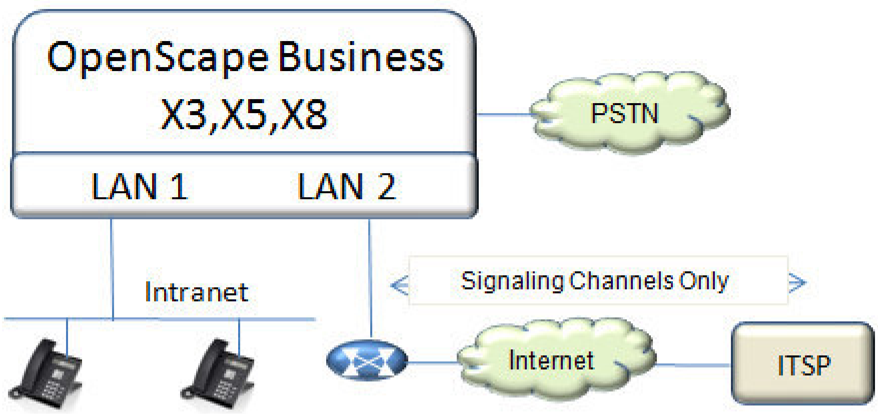
OpenStage TDM and or IP telephones

5. 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.







6. 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).

7. Verizon IP Trunk Certification

This section outlines the actual OpenScape Business elements used in the Verizon IP Trunk Certification process for Verizon US IP Trunks, EMEA IP Trunks and IPCC Trunks:

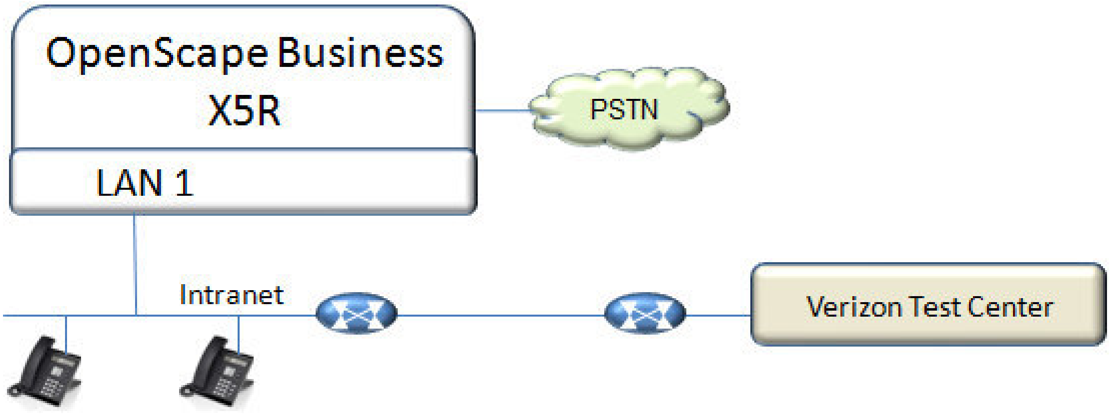
Example OpenScape Business Voice Bill of Materials (as certified):

Verizon OpenScape Business Lab Bill of Materials

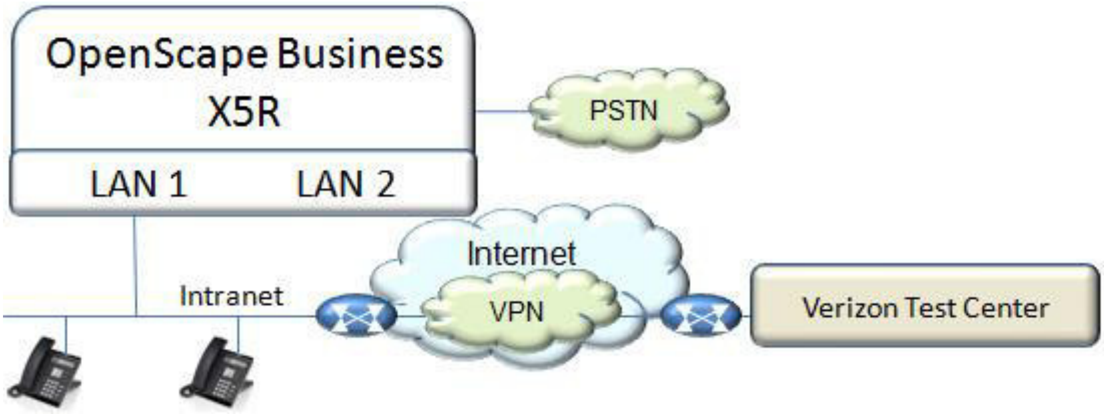
Qty	Part Number	Description
OpenScape Business X5R/X3R HW		
1	L30251U0600G611	OSBiz V1 X5R Base sys,2S0/8UP0/4ab
1	L30251U0600A594	ANALOG TRUNK MODULE (TLANI4R) 3300/3500
OpenScape Business Common Licenses		
4	L30250U0622B571	OSBiz V1 IP User
4	L30250U0622B572	OSBiz V1 TDM User
4	L30250U0622B581	OSBiz V1 Voicemail
4	L30250U0622B574	OSBiz V1 S2M/SIP Trunks
4	L30250U0622B594	OpenScape Business V1 myAttendant
WorkPoints		
4	L30250F0600C151	OPENSTAGE 40 T - LAVA

7.1. OpenScape Business High Level Configuration (as certified):

Testing Performed at the Verizon Lab



Testing performed by Local North American and European Labs



8. Load Issues and Concerns:

The following issues were identified during CPE Interop testing.

The OpenScape Business does not support REFER or REFER with Replaces; Unscreened ANI; generation of RTCP packets; it does not generate “183 Session Progress” responses (only “180 Ringing”); and, always sends RTP packets even when the system places a call on “Hold”.

The following issues were identified during Delta Lab testing.

The OpenScape Business critical defect that was previously identified has been corrected with a number of changes to the system configuration, re-tested, and found to handle SIP UDP fragmentation correctly and the critical defect has been closed.

The OpenScape Business provides unacceptable facsimile completion ratios with the Verizon Network Gateway types. Completion ratios with the various Network Gateway and VSP types, as well as, SONUS Gateway ranged between 90 and 94 percent and this is considered unacceptable.

OpenScape Business utilizes INVITE / Re-INVITE methods for call-forwarding but retains the media stream instead of releasing it to reduce bandwidth usage, and, the wrong Caller-ID is delivered to the terminating phone. *This will be corrected with the release of the OpenScape Business Platform (Next Generation of the OpenScape Business) scheduled for release in the 4th Calendar quarter of 2013.*

On calls where codecs are re-negotiated the OptiPoint 420 Standard phone plays-out DTMF digits using RFC2833/4733 rtp events even when this capability was never offered or negotiated. Please note that the OptiPoint 420 is a manufacturer discontinued device that may be present on some OpenScape Business customer locations. Reseller must notify the customer about the results of the testing mentioned in the sentence above... Note that the other phones tested (OpenStage 20T and OpenStage 30T) do not exhibit this defect and correctly play out DTMF digits using inband DTMF for re-negotiated codec calls.

EXCLUSIONS:

During CPE Interop testing 2CPE and IPCC capabilities were not required by the Verizon VTM and were not tested. During Delta Lab testing H.264 video codec and G.722 codec support were not tested as none of the phones supplied and tested support these capabilities. Also, during Delta Lab testing the call-forwarding via REFER method was not tested as this method is not supported by OpenScape Business.

Interoperability and Delta Testing was performed using the current Verizon production software releases. Testing was performed using the requirements and assumptions as provided in the CPE Interop IP Trunking Test Plan v1.4 and CPE Interop IP Trunking Delta Test Plan v1.2.

9. Test Results

The following Verizon IP Trunk services and specific test cases have been certified with any exceptions noted below:

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
	Security			
TC1	Layer 2 IPSec Authentication	Pass		
	DNS SRV			
TC2	Service Protocols/Port Adherence	N/A		This capability will be tested during the Delta Testing phase of certification.
	Inbound			
TC3	Inbound Call Loop Avoidance	Pass		
TC4	Inbound call with originator (PSTN) release	Pass		
TC5	Inbound call with terminator (CPE) release	Pass		
TC6	Inbound call - Hang-up during Ring phase	Pass		
TC7	Inbound Call - vendor phone not registered/online	Pass		
TC8	Inbound Calling Line Identification (Caller-ID)	Pass	Caller ID displayed correctly on the IP-PBX phone.	
TC9	Inbound Call Waiting	Pass		
TC10	Inbound G.711 Fax	Pass		
TC11	Inbound T.38 Fax	Pass		

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC12	Inbound Call from PSTN with Privacy Restricted	Pass	Caller ID on the PBX phone display "Anonymous and Number Unknown".	
	Outbound			
TC13	Unscreened ANI using Diversion Header	N/A	Siemens HiPath 3000 does not support this capability.	
TC14	Unscreened ANI using P-Asserted Identity	N/A	Siemens HiPath 3000 does not support this capability.	
TC15	Outbound call with Originator (CPE) release	Pass		
TC16	Outbound call with Terminator (PSTN) release	Pass		
TC17	Outbound call - Hangup during ring phase	Pass		
TC18	Outbound 1+10digit call	Pass		
TC19	Outbound International Call	Pass		
TC20	Outbound 311 Non-Emergency call	Pass		
TC21	Outbound 555-1212 Directory Assistance	Pass		
TC22	Outbound 411 Directory Assistance	N/A	Dialed 411 (without "1") and received intercept saying "1" was needed.	Not supported from 972-728-xxxx DIDs. Capture provided by vendor shows responses expected.
TC23	Outbound 1411 Directory Assistance	Pass		
TC24	Outbound 711 Telephone Relay Services (Hearing Impaired)	Pass		
TC25	911 Emergency Service	Pass		
TC26	Outbound 511 Information Line	Pass	Received recording that 511 number is not in service	Capture provided by vendor shows expected response that number is not in service – "511" not supported by the 972-728-xxxx DIDs.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC27	Outbound Toll-Free Call	Pass		
TC28	Operator assistance (0+ Local)	Pass		
TC29	Operator assistance (0+ Toll)	Pass		
TC30	Operator assistance (0 Minus)	Pass		
TC31	Operator assistance (00 Minus)	Pass		
TC32	Operator assistance (01+ international)	Pass		
TC33	Outbound G.711 Fax	Pass		
TC34	Outbound T.38 Fax	Pass		
TC35	Outbound Calling Line Identifier (Caller ID)	Pass		Was correct Calling number displayed at termination – if so - Pass?
TC36	Outbound Fast Answer	Pass		
TC37	Outbound Call to PSTN with Privacy Requested	Pass		Vendor provided second capture that now includes the "privacy:id" field and the CLI information is withheld at the terminating telephone.
TC38	Calling Party Number not provisioned	Pass		
	Protocols			
TC39	UDP for SIP	Pass		
TC40	SDP support (RFC 2327)	Pass		
TC41	RTP and RTCP support (RFC 3550)	Pass	System does not support generating RTCP packets	
TC42	SIP Headers	Pass		
TC43	18x Behavior	Pass	System does not support generating 183 responses	
TC44	302 Behavior	Pass		

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC45	Diversion Header	Pass		The transfer to another IP-PBX line occurs within the IP-PBX
TC46	DTMF RFC 2833—Outbound	Pass		
TC47	DTMF RFC 2833—Inbound	Pass		
TC48	Offer/Answer with SDP (RFC3264)	Pass		
TC49	Call Hold (RFC 3264)	Pass		
TC50	Media Inactivity	Fail		The IP-PBX never stops sending RTP packets.
TC51	FQDN	Blocked		Only tested for 2CPE testing configurations.
	Media			
TC52	G.711 ulaw	Pass		
TC53	G.729 and G.729a	Pass		
TC54	Codec Negotiation	Pass		
TC55	Early Media Support	Pass		
	Diffserv			
TC56	RTP	Pass		
TC57	SIP	Pass		
	Attended Call Transfer Re-INVITE Method			
TC58	IPPBX-PSTN-IPPBX	Pass		The transfer is performed within the IP-PBX.
TC59	IPPBX-PSTN-PSTN	Pass		
TC60	PSTN-IPPBX-IPPBX	Pass		The transfer is performed within the IP-PBX.
TC61	PSTN-IPPBX-PSTN	Pass		Vendor provided capture (re-test) verified to be correct.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
	Semi-Attended Call Transfer Re-INVITE Method			
TC62	IPPBX-PSTN-IPPBX	Pass		The transfer is performed within the IP-PBX.
TC63	IPPBX-PSTN-PSTN	Pass		
TC64	PSTN-IPPBX-IPPBX	Pass		The transfer is performed within the IP-PBX.
TC65	PSTN-IPPBX-PSTN	Pass		
	Blind Call Transfer Re-INVITE Method			
TC66	IPPBX-PSTN-IPPBX	Pass		The transfer is performed within the IP-PBX.
TC67	IPPBX-PSTN-PSTN	Pass		
TC68	PSTN-IPPBX-IPPBX	Pass		The transfer is performed within the IP-PBX.
TC69	PSTN-IPPBX-PSTN	Pass		
	Attended Call Transfer REFER Method			
TC70	IPPBX-PSTN-IPPBX	N/A	The Siemens HiPath 3000 does not support REFER	
TC71	IPPBX-PSTN-PSTN	N/A	The Siemens HiPath 3000 does not support REFER	
TC72	PSTN-IPPBX-IPPBX	N/A	The Siemens HiPath 3000 does not support REFER	
TC73	PSTN-IPPBX-PSTN	N/A	The Siemens HiPath 3000 does not support REFER	
	Semi-Attended Call Transfer REFER Method			
TC74	IPPBX-PSTN-IPPBX	N/A	The Siemens HiPath 3000 does not support REFER	

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC75	IPPBX-PSTN-PSTN	N/A	The Siemens HiPath 3000 does not support REFER	
TC76	PSTN-IPPBX-IPPBX	N/A	The Siemens HiPath 3000 does not support REFER	
TC77	PSTN-IPPBX-PSTN	N/A	The Siemens HiPath 3000 does not support REFER	
	Blind Call Transfer REFER Method			
TC78	IPPBX-PSTN-IPPBX	N/A	The Siemens HiPath 3000 does not support REFER	
TC79	IPPBX-PSTN-PSTN	N/A	The Siemens HiPath 3000 does not support REFER	
TC80	PSTN-IPPBX-IPPBX	N/A	The Siemens HiPath 3000 does not support REFER	
TC81	PSTN-IPPBX-PSTN	N/A	The Siemens HiPath 3000 does not support REFER	
	Call Conference			
TC82	IPPBX-PSTN-IPPBX	Pass		
TC83	IPPBX-PSTN-PSTN	Pass		
TC84	PSTN-IPPBX-IPPBX	Pass		
TC85	PSTN-IPPBX-PSTN	Pass		
	CPE Failover Behavior			
TC86	Options method request and response	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC87	Round-Robin (Load share 50/50 between the two CPEs	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC88	Primary/Secondary failover (Hunt)	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC89	Both CPE Fail behavior	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC90	Verizon Alternate Route using DNS/SRV query	N/A		This is a future requirement that is not supported at this time.
TC91	Verizon Alternate Route using IP:port assignment	N/A		This is a future requirement that is not supported at this time.
	Ambient Noise			
TC92	Ambient Noise – CPE to PSTN	Pass		
TC93	Ambient Noise – PSTN to CPE	Pass		
	EMEA Retail Interop			
	Inbound – Calls From Verizon PSTN to the Vendor VoIP			
TC94	Inbound Fax	Pass		
TC95	Inbound - G.711 CODEC Negotiation	Pass		
TC96	Inbound - G.729 CODEC Negotiation	Pass		
	Outbound – Vendor VOIP TO Verizon PSTN CALL DIRECTION			
TC97	Outbound - FAX	Pass		
TC98	Outbound - G711 CODEC Negotiation	Pass		

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC99	Outbound - G729 CODEC Negotiation	Pass		
TC100	Outbound - Call Redial	Pass		
	Re-Invite Call Test Cases			
	Attended Call Transfers			
TC101	IP-PBX calls PSTN attended transfer to IP-PBX	Pass		
TC102	IP-PBX calls PSTN attended transfer to PSTN	Pass		
TC103	PSTN calls IP-PBX attended transfer to IP-PBX	Pass		
TC104	PSTN calls IP-PBX attended transfer to PSTN	Pass		
	Semi-Attended Call Transfers			
TC105	IP-PBX calls PSTN semi-attended transfer to IP-PBX	Pass		
TC106	IP-PBX calls PSTN semi-attended transfer to PSTN	Pass		
TC107	PSTN calls IP-PBX semi-attended transfer to IP-PBX	Pass		
TC108	PSTN calls IP-PBX semi-attended transfer to PSTN	Pass		
	Blind Call Transfers			
TC109	IP-PBX calls PSTN with blind transfer to IP-PBX	Pass		The transfer is performed within the IP-PBX.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC110	IP-PBX calls PSTN with blind transfer to PSTN	Pass		
TC111	PSTN calls IP-PBX with blind transfer to IP-PBX	Pass		The transfer is performed within the IP-PBX.
TC112	PSTN calls IP-PBX with blind transfer to PSTN	Pass		
	REFER Call Transfer Test Cases			
	Attended Call Transfers			
TC113	IP-PBX calls PSTN attended transfer to IP-PBX	N/A	The Siemens HiPath 3000 does not support REFER	
TC114	IP-PBX calls PSTN attended transfer to PSTN	N/A	The Siemens HiPath 3000 does not support REFER	
TC115	PSTN calls IP-PBX attended transfer to IP-PBX	N/A	The Siemens HiPath 3000 does not support REFER	
TC116	PSTN calls IP-PBX attended transfer to PSTN	N/A	The Siemens HiPath 3000 does not support REFER	
	Semi-Attended Call Transfers			
TC117	IP-PBX calls PSTN semi-attended transfer to IP-PBX	N/A	The Siemens HiPath 3000 does not support REFER	
TC118	IP-PBX calls PSTN semi-attended transfer to PSTN	N/A	The Siemens HiPath 3000 does not support REFER	
TC119	PSTN calls IP-PBX semi-attended transfer to IP-PBX	N/A	The Siemens HiPath 3000 does not support REFER	
TC120	PSTN calls IP-PBX semi-attended transfer to PSTN	N/A	The Siemens HiPath 3000 does not support REFER	
	Blind Call Transfers			

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC121	IP-PBX calls PSTN with blind transfer to IP-PBX	N/A	The Siemens HiPath 3000 does not support REFER	
TC122	IP-PBX calls PSTN with blind transfer to PSTN	N/A	The Siemens HiPath 3000 does not support REFER	
TC123	PSTN calls IP-PBX with blind transfer to IP-PBX	N/A	The Siemens HiPath 3000 does not support REFER	
TC124	PSTN calls IP-PBX with blind transfer to PSTN	N/A	The Siemens HiPath 3000 does not support REFER	
	Conference Call Test Cases			
TC125	IP-PBX calls PSTN conference to IP-PBX	Pass		The conference is performed within the IP-PBX.
TC126	IP-PBX calls PSTN conference to PSTN	Pass		
TC127	PSTN calls IP-PBX conference to IP-PBX	Pass		
TC128	PSTN calls IP-PBX conference to PSTN	Pass		
	IPCC Testing			
	IPCC with PIP Simulation			
TC129	OPTIONS Method Request and Response	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
	IP Toll Free			
	Inbound Calls (Verizon Business PSTN to Vendor)			
TC130	Inbound Calls with Request-URI Set to Vendor's Provisioned URL Address	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC131	Inbound Call with Originator (PSTN) Release	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC132	Inbound Call with Terminator (SIP) Release	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC133	Inbound Call with Proprietary Headers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
	Network Call Redirect (NCR) Testing with no Enhanced Transfer			
TC134	Inbound Call with NCR with Answer	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC135	Inbound Call with NCR with Ring No Answer	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC136	Inbound Call with NCR with User Busy	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
	Outbound Calls (Vendor to Verizon Business)			
TC137	Outbound Call with CPN Allowed – Privacy Null	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC138	Supported SIP Methods (RFC 3261)	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC139	UDP for SIP and Long Message Support	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
	IP TF Transfer (Basic) - Blind REFER			
TC140	PSTN to SIP UA to SIP UA	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC141	PSTN to SIP UA to PSTN	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC142	PSTN to SIP UA to SIP UA – Party C Unavailable	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC143	PSTN to SIP UA to PSTN – Party C Unavailable	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC144	PSTN to SIP UA to SIP UA - Party A Disconnects Before C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC145	PSTN to SIP UA to PSTN - Party A Disconnects Before C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC146	PSTN to SIP UA to SIP UA - Party B Waits, Party C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC147	PSTN to SIP UA to PSTN - Party B Waits, Party C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC148	PSTN to SIP UA to SIP UA - Party C Unavailable	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC149	PSTN to SIP UA to PSTN - Party C Unavailable	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC150	PSTN to SIP UA to SIP UA - Party A Disconnects Before C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC151	PSTN to SIP UA to PSTN - Party A Disconnects Before C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
	IP TF Transfer (Basic) - Attended – REFER with Replaces			
TC152	PSTN to SIP UA to SIP UA - Party B Sends Immediate BYE, Party C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC153	PSTN to SIP UA to PSTN – Not Supported	Not tested.		This test case is not supported at this time and is a place holder for possible future testing
TC154	PSTN to SIP UA to SIP UA - Party B Waits, Party C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC155	Party A Disconnects Before B Sends REFER with Replaces	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC156	Party B Disconnects Without Sending REFER with Replaces	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
	IP TF Transfer (Enhanced Transfer) - Blind – DTMF			
TC157	PSTN to SIP UA to SIP UA - Party C Answers – Party C has Enhanced Transfer Capabilities	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC158	PSTN to SIP UA to SIP UA - Party C Answers – Party C Does Not Have Enhanced Transfer Capabilities	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
	IP TF Transfer (Enhanced Transfer) - Attended – DTMF			
TC159	PSTN to SIP UA to SIP UA - Party C Answers – Party C has Enhanced Transfer Capabilities	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC160	PSTN to SIP UA to SIP UA - Party C Answers – Party C Does Not Have Enhanced Transfer Capabilities	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
	IP IVR			
	Inbound Calls (Verizon Business PSTN to Vendor)			
TC161	Inbound Call with Originator (PSTN) Release	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC162	Inbound Call with Terminator (SIP) Release	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC163	Inbound Call with Disconnect During Ring Phase (Cancel Call)	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC164	Inbound Call with Vendor Phone Not Registered with SIP PBX	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC165	Inbound Call with Ring No Answer Timer Expire	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC166	Inbound Call with User Busy	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC167	Inbound Call with CPN Allowed – Privacy Null	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC168	Inbound Call with CPN Restricted – Privacy "id"	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC169	Inbound Call with Long Duration	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
	Network Call Redirect (NCR) Testing			
TC170	Inbound Call with NCR with Answer	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC171	Inbound Call with NCR with Ring No Answer	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC172	Inbound Call with NCR with User Busy	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC173	Inbound Call with Release Link Trunking (RLT)	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC174	Call Hold	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC175	Media Inactivity (Call Hold Long Duration)	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
	IP IVR Custom Transfers (Blind – REFER)			
TC176	PSTN to SIP UA to SIP UA - Party B Sends Immediate BYE, Party C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC177	PSTN to SIP UA to PSTN - Party B Sends Immediate BYE, Party C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC178	PSTN to SIP UA to SIP UA - Party C Unavailable	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC179	PSTN to SIP UA to PSTN - Party C Unavailable	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC180	PSTN to SIP UA to SIP UA - Party A Disconnects Before C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC181	PSTN to SIP UA to PSTN - Party A Disconnects Before C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC182	PSTN to SIP UA to SIP UA - Party B Waits, Party C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC183	PSTN to SIP UA to PSTN - Party B Waits, Party C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC184	PSTN to SIP UA to SIP UA - Party C Unavailable	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC185	PSTN to SIP UA to PSTN - Party C Unavailable	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC186	PSTN to SIP UA to SIP UA - Party A Disconnects Before C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC187	PSTN to SIP UA to PSTN - Party A Disconnects Before C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
	IP IVR Custom Transfers (Blind – DTMF)			
TC188	PSTN to SIP UA to SIP UA	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC189	PSTN to SIP UA to PSTN	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
	IP IVR Custom Transfers (Attended – REFER with REPLACE)			

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC190	PSTN to SIP UA to SIP UA - Party B Sends Immediate BYE, Party C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC191	PSTN to SIP UA to PSTN – Not Supported	N/A		This test case is not supported at this time and is a place holder for possible future testing
TC192	PSTN-SIP UA-SIP UA - Party B Waits, Party C Answers	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC193	Party A Disconnects Before B Sends REFER with Replaces	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC194	Party B Disconnects Without Sending REFER with Replaces.	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
	IP IVR Custom Transfers (Attended – DTMF)			
TC195	PSTN to SIP UA to SIP UA	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC196	PSTN to SIP UA to PSTN	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
	Media			
TC197	Verizon Business SDP Offer – Vendor SDP Answer	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.
TC198	Vendor SDP Offer – Verizon Business SDP Answer	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC199	Verifying RTP – Phone on Mute	Not tested.		This capability was not tested as it was deemed to be not required by Verizon VTM and Verizon Marketing.

10. CPE Interop IP Trunking Delta test cases

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
	UDP Transport for SIP for PIP Access Test Cases			
TC1	UDP Transport for SIP for PIP Access - Fragmented UDP	~~Fail –~~ **PASS** **Critical defect Closed** ~~written~~	Per Siemens network expert from the development team, the current status is that they do support fragmentation; there is no specific code component for that, however, it is done automatically by the IP Stack. If a packet's size is greater than 1500 bytes then it is automatically fragmented.	**UPDATE:** On 6/4/2013 Siemens support personnel participated in a WebEx session with Vz to review the Siemens HiPath 3000 V9 configuration settings. It was determined that the STUN configuration settings (multiple locations) were incorrectly set and Siemens made a number of changes. After saving the changes and re-setting the system the SIP UDP fragmentation tests were re-ran and the results showed that the Siemens HiPath 3000 V9, if configured correctly, would correctly process SIP UDP packets that were fragmented, and, would fragment SIP UDP packets inbound and outbound as necessary. Wireshark captures and associated "screenshots" were captured and saved showing successful results for these tests that were conducted with Vz ITP testing platform. As a result, the "critical defect #32" that was written against the Siemens HiPath 3000 V9, will be closed and a revision to the final report will be issued reflecting these changes.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC2	UDP Transport for SIP for PIP Access - Not to exceed the Path MTU	Pass		Siemens HiPath 3000 V9 supports a mechanism for pre-configuring the maximum packet size below the MTU. This can be done by changing the value in the HG1500 / STMI WBM under menu Explorers > Network Interfaces > Edit LAN1. So, if the SIP message is more than 1500 bytes then the message will be fragmented. UPDATE: On 6/4/2013 Verizon verified this functionality in the Siemens HiPath 3000 V9 - see TC1.
TC3	UDP Transport for SIP for PIP Access - Message Size	Pass	Siemens HiPath 3000 V9.0 does not have a specific mechanism to limit SIP message size to less than 16 kb.	As indicated for TC2 above, the Siemens HiPath 3000 has a mechanism to limit SIP message size to less than 1500 bytes, therefore, no SIP message would exceed 16 KB. Any SIP message that would exceed 1500 bytes should automatically be fragmented. UPDATE: On 6/4/2013 Verizon verified this functionality in the Siemens HiPath 3000 V9 - see TC1.
	Signaling CLI Test Cases			
TC4	Outbound-Proxy	Pass		Siemens HiPath 3000 V9.0 supports formatting of From, P-Asserted-ID, and Diversion headers and uses either the CPE_FQDN or CPE_IPADDR when the call originates from one of its subscribers or is forwarded from one of its subscribers. HiPath 3000 does not support the use of Remote-Party-ID headers.
	Locating SIP Servers Test Cases			

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC5	Locating SIP Servers - DNS TTL	Pass		Siemens HiPath 3000 V9.0 will do a DNS SRV query prior to TTL expiration if a call is attempted, or, will continuously do DNS SRV queries at 75 second intervals if no call is attempted or system response necessary.
TC6	Locating SIP Servers - Proxy Failover	Pass		Siemens HiPath 3000 V9.0 fully supports DNS SRV records and attempts to contact the secondary proxy upon detection of a call failure or no response from the first proxy.
	DNS Priorities, Weights, and Ports			
TC7	Locating SIP Servers - IP Address and Port	N/A		Siemens HiPath 3000 V9.0 fully supports DNS resolution and, as a result, this test is not applicable.
TC8	Locating SIP Servers - DNS SRV Records	Pass		Siemens HiPath 3000 V9.0 supports DNS SRV queries (multiple) with priorities, weights, and ports correctly, and supports A Records of the DNS SRV responses to determine primary and secondary targets as defined in RFC2782.
TC9	Locating SIP Servers - Call Setup Timer Expiry	Fail		The Siemens HiPath 3000 V9.0 does not support a configurable timer for initial call setup failure. However, the HiPath 3000 utilizes the general SIP timeout = 32 seconds. After this time with no response, the HiPath 3000 triggers a rerouting or failover to a secondary route should it exist. The HiPath 3000 also triggers a rerouting to any secondary proxy when the HiPath 3000 receives a SIP failure message.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC10	Locating SIP Servers - DNS SRV failover reporting	Pass		Siemens HiPath 3000 V9.0 utilizes the general SIP timeout = 32 seconds. After this time with no response, the HiPath 3000 triggers a rerouting or failover to a secondary route should it exist. The HiPath 3000 also triggers a rerouting to any secondary proxy when the HiPath 3000 receives a SIP failure message.
TC11	Locating SIP Servers - Call Setup Failure Tracking	Fail		Siemens HiPath 3000 V9.0 does not support the capability (reports, records, or logs) to track call setup failures.
	FQDN and IP Addresses (PIP)			
TC12	FQDN and IP Addresses - Private IP	Pass		Siemens HiPath 3000 V9.0 uses IP addresses routable from Verizon Business' PIP network interface to the customer when communicating with the Verizon VoIP service.
TC13	FQDN and IP Addresses - Private IP and NAT	Pass		Any NAT for Private IP addresses is done in the Siemens HiPath 3000 V9.0 devices.
TC14	Unknown and Proprietary Headers	Pass		Siemens HiPath 3000 V9.0 ignores any unknown or proprietary headers that it does not understand as required by RFC3261.
	DTMF Payload			
TC15	DMTF Payload (Inbound and RFC 2833)	Pass		
TC16	DMTF Payload (Transmit)	Pass		
TC17	DMTF Payload (Receive)	Pass		

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC18	DMTF Payload (SLGs)	Pass		Siemens HiPath 3000 phones and system exhibit the following performance: - Fixed digit duration of 90 ms regardless of actual keypresses. - Minimum interdigit gap of 350 ms is enforced. - Time delta between start and end events varies between 90 - 150 ms.
TC19	DTMF Payload - Minimum Digit Duration	Pass		
TC20	DTMF Payload - Long Digit Duration	Fail	Please note that the OptiPoint 420 is manufacturer discontinued product and is no longer available for new system sales.	With the Siemens HiPath 3000 the phones provided and tested (OptiPoint 420 Standard, OpenStage 20T, OpenStage 30T) provide fixed length DTMF digits and are encoded with fixed digit durations of 90 ms regardless of keypress. Note that these are digital phones and this is typical behavior.
TC21	DTMF Payload - Minimum Interdigit Time	Pass		Siemens HiPath 3000 will enforce a minimum interdigit gap time of 350 ms.
TC22	DTMF Payload - Long Interdigit Time	Pass		
TC23	DTMF Payload - Signal Velocity	Pass		
	AVT Payload Type			

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC24	Verifying Up Speed Renegotiation with DMTF after Re-invite	Pass Major defect written!	Please note that the OptiPoint 420 is manufacturer discontinued product and is no longer available for new system sales.	Two of the three Siemens phones tested (OpenStage 20T, OpenStage 30T, and, OptiPoint 420 Standard) performed this test correctly and sent the DTMF digits inband via the G.711 codec. Because of this, this test case will be marked as "pass"; **however there will be a major defect written against the system and the OptiPoint 420 Standard phone should NOT BE USED until this defect is corrected.** When re-negotiating a g.711 codec with the Siemens HiPath 3000 the only codec negotiated is an G.711 "0" codec, however, the Siemens OptiPoint 420 Standard phone plays out the digits from the system using RFC2833 "101" which was never negotiated! This violates RFC4733 which replaces RFC2833 and no digits are received at the termination. If this were an IVR system, it is likely that the call would not be successful as the IVR menu could not be navigated as there would be no digits sent to the IVR system.
TC25	Verifying Up Speed Renegotiation with DMTF after Re-invite - G729 w/ RFC 2833	Pass		

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC26	Silence Suppression	Pass		Siemens HiPath 3000 V9.0 does not use silence suppression (continues sending RTP even when "muted") and does not use "13" (comfort noise) for Hold or in SDP messages. G.711 calls were also tested and SDP did not contain "13" silence suppression and continued sending RTP even when "muted". MOH was also transmitted and used a=sendonly / a=sendrecv to place call on-hold and take the call off-hold.
	Facsimile			
TC27	Facsimile - Network Gateway support	Pass		Siemens HiPath 3000 supported facsimile with all gateway types.
TC28	Facsimile – Echo Canceller State V17 Fax	Pass		
TC29	Facsimile – Echo Canceller State V34 Fax	Pass		
TC30	Facsimile - 95 percent completion rate minimum	Fail		Siemens HiPath 3000 V9.0 provided unacceptable facsimile completion ratios with the Verizon Network Gateway types. Completion ratios with the various Network Gateway and VSP types, as well as, SONUS Gateway ranged between 90 and 94 percent and this is considered unacceptable.
TC31	Facsimile - Fall back from Super Group 3 to Group 3 FAX	Pass		Siemens HiPath 3000 supports fall back from Super Group 3 FAX to Group 3 FAX and detects Group 3 fax and successfully completes.

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC32	T.38 Redundancy Level	Pass		05-14-2013 - UPDATE - Siemens told us what option to change in the configuration to enable T.38 udp redundancy – we changed the option and re-tested. Siemens HiPath 3000 now supports T.38 and also supports T38udpredunancy level 2.
TC33	CPE To/From VoIP Gateway Fax Call Initiated over G.729– No Support for T.38 on Terminating Gateway	Pass		
TC34	CPE To/From VoIP Gateway Fax Call Initiated over G.729– No Support for T.38 on Originating Gateway	Pass		
TC35	CPE To/From VoIP Gateway Fax Call Initiated over G.711– No Support for T.38 on Terminating Gateway	Pass		
TC36	CPE To/From VoIP Gateway Fax Call Initiated over G.711– No Support for T.38 on Originating Gateway	Pass		
	Voice Quality			
TC37	Voice Quality - G.711	Pass		Siemens HiPath 3000 supports a minimum PESQ score of at least 4.0 with the G.711 codec.
TC38	Voice Quality - G.729	Pass		Siemens HiPath 3000 supports a minimum PESQ score of at least 3.5 with the G.729 codec.
	Call Forward			

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC39	Call Forward - INVITE	Fail Minor defect written!		Siemens HiPath 3000 V9.0 supports Call Forwarding via INVITES / Re_INVITES, however, the Caller ID is not correctly presented (Call ID delivered to Party C is "forwarded line number" rather than the "originating line number"). This call was from a PSTN caller (A) to the IP-PBX line (B) which is forwarded to another PSTN line (C). Signalling is anchored at the IP-PBX and the Media is also anchored at the IP-PBX -- This is considered a "failed test case" as the media stream should be released from the IP-PBX and the wrong Call ID is delivered to the terminating phone.
TC40	Call Forward - REFER	N/A		Siemens HiPath 3000 V9.0 does not support REFER method for call-forwarding. REFER is used to invoke transfers. OpenScape Voice Server interworks REFER request and does not send REFER towards OS SBC / Verizon on a SIP trunk. Call-forwarding is accomplished using INVITE / Re-INVITE methods.
	Mid-Call Codec Renegotiation			

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC41	Mid-call codec renegotiation against multiple network gateways	Pass		Siemens HiPath 3000 V9.0 all test calls completed successfully to and from the PSTN using each of the Network Gateways (except the Sonus test calls which was only performed for outbound from the IP-PBX since Sonus is not used for inbound calls); voice path in both directions was verified for all calls at various times during the extended call times. In all calls the FAX simulation tone did cause renegotiation from G.729 codecs to G.711 codecs. Note, however, that the HiPath 3000 choose G.711 a-law codec rather than G.711 u-law codec for some reason – but – both codecs were in INVITES and re-negotiations.
	Codecs			
TC42	Codecs - G.729	Pass		Siemens HiPath 3000 V9.0 continues sending RTP even when "muted" or "on-hold" and does not use "13" (comfort noise) in SDP messages. a=sendonly used for call hold.
TC43	Codecs - G.722	N/A		Siemens HiPath 3000 V9.0 does not support G.722 codecs.
TC44	Codecs - H.264	Not Tested		Siemens OpenStage phones do not support H.264 video and H.264 video was not tested with the Siemens HiPath 3000 V9.0 configuration. Siemens indicated that they have an OpenScape Desktop Soft Client Personal Edition (ODC-PE) but it was not tested during Interop or Delta testing efforts. Siemens indicated that the OpenScape Desktop Client Personal Edition (ODC-PE) softclient supports H.264 video.
	CNAME			

Test Case ID	Requirement	Result	Vendor Comments	Verizon Comments
TC45	CNAME	Pass		Siemens HiPath 3000 V9.0 supports delivery of CNAME in the display name portion of the From header; however, the Siemens terminating phone sets tested (OptiPoint 420. OpenStage 20T and OpenStage 30T) only display the calling number and do not seem capable of delivering or displaying the Calling station information.

11. 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.

11.1. Configure the OpenScape Business System Information

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

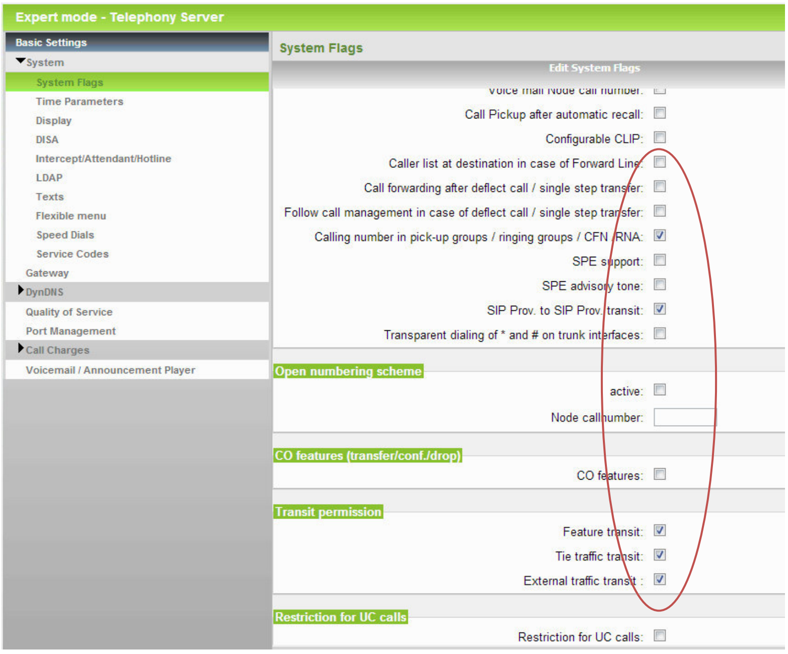
11.1.1. Open the OpenScape Business System Administration tool

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.

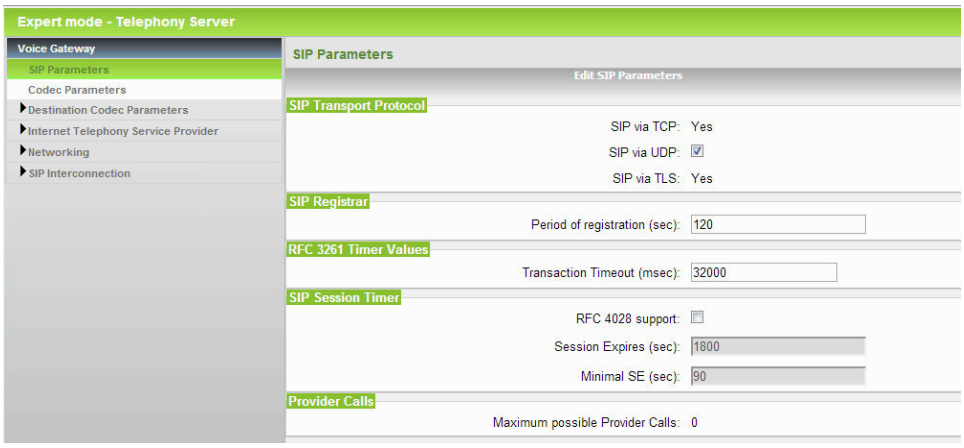
11.1.2. Configure the system Parameters

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



11.1.3. Setting Up the General SIP Trunk Parameters

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



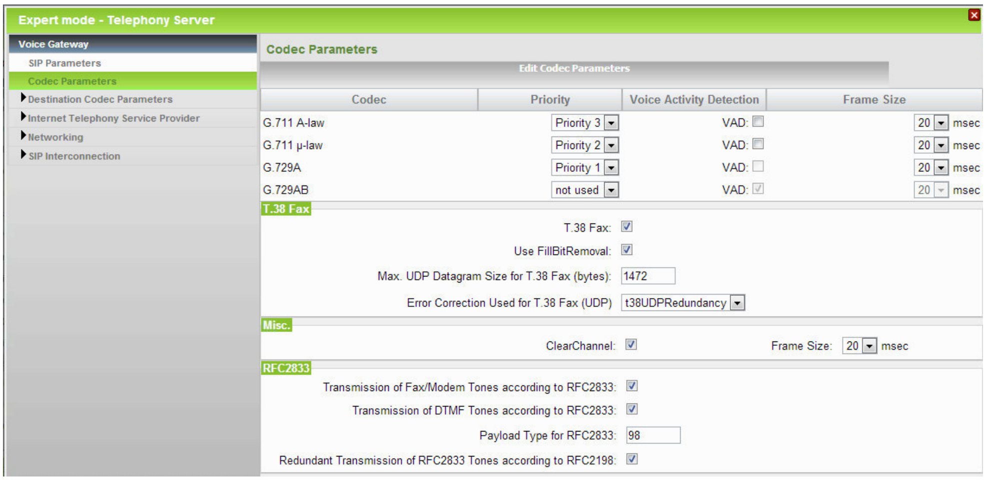
Insure that the SIP transport protocol flags are set as displayed above

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.



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

11.1.4. Set STUN Configuration Parameter

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

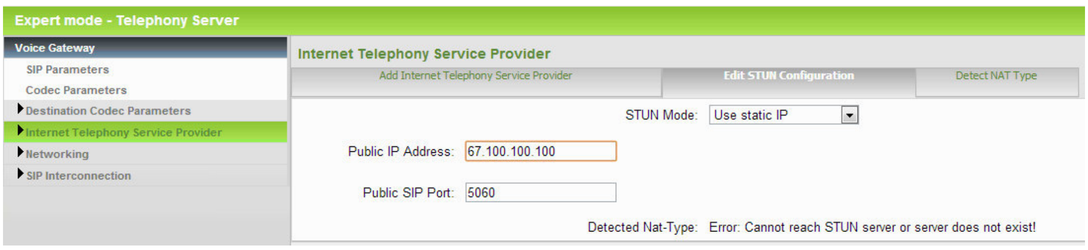
Select "Use static IP" as the STUN Mode

Enter your Public IP address information

Leave the Public SIP Port of 5060 in place

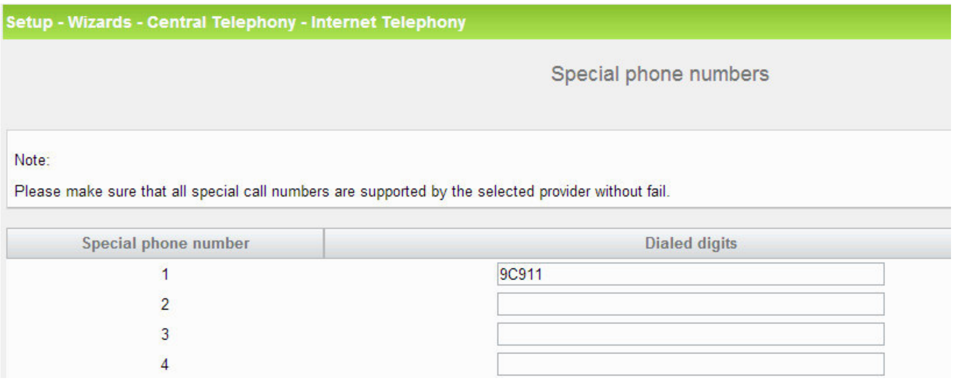
After making the changes press the Apply button to save your changes

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

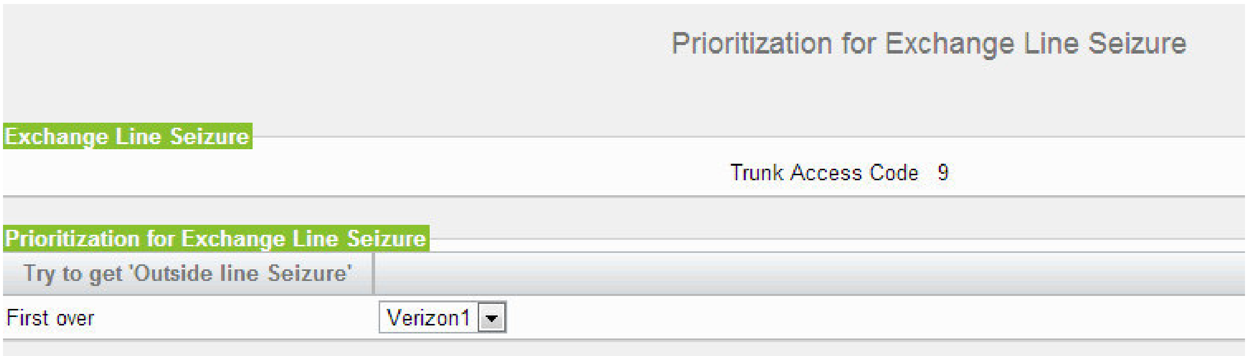


11.1.5. Define Special Phone Numbers and Primary Line Seizure Route Group

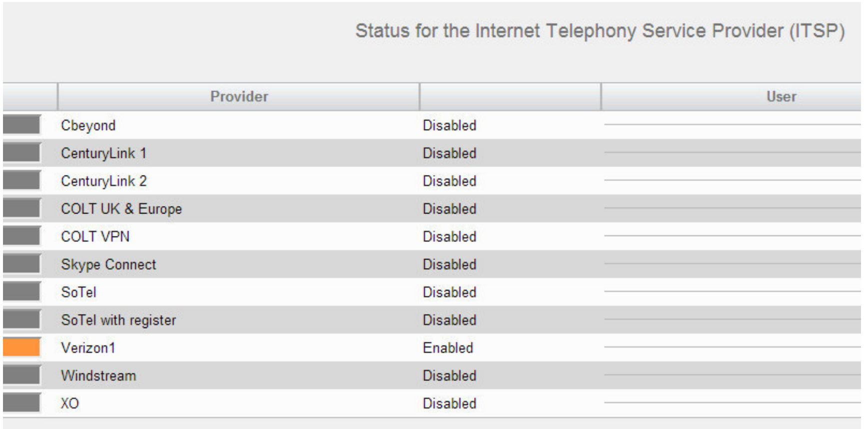
This **Special Phone Numbers** form allows you define special telephone numbers such as 911. Please note that 9C911 is a default entry. Press Ok & Next to move to the next step



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



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 re-display 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.

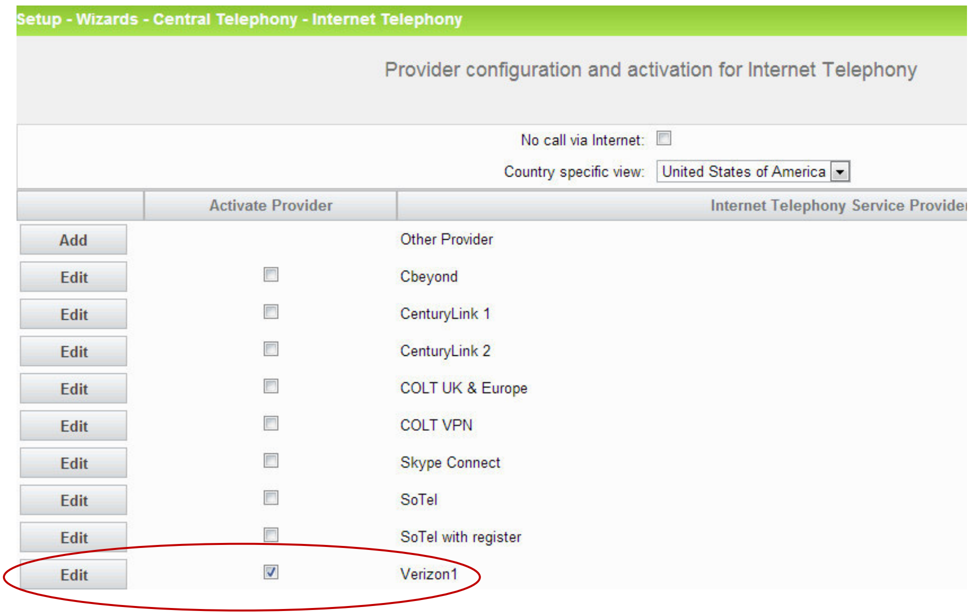


11.1.6. Revise the Verizon ITSP profile Domain and Proxy information

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

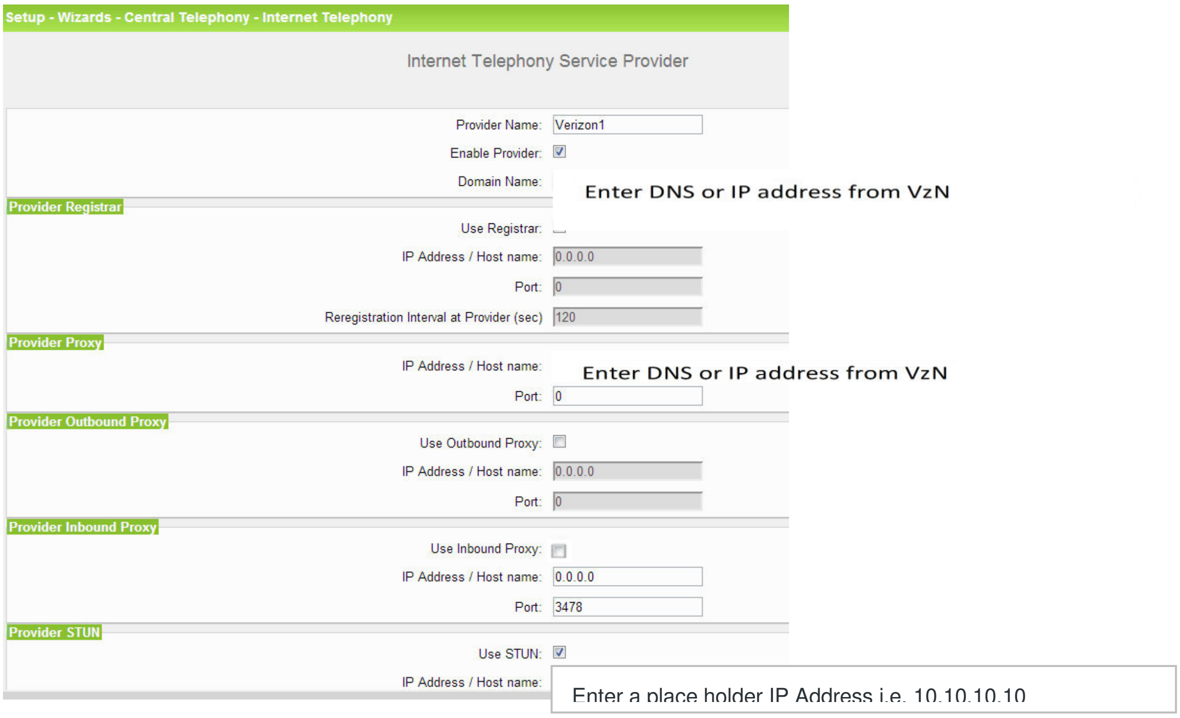


In the Domain Name field you will need to enter the Domain Name or IP address information that is received from Verizon. This is because there are no fixed-public servers for Verizon and a private VPN will be used for connection to the Verizon services.

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 “dummy” IP address such as 10.10.10.10

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



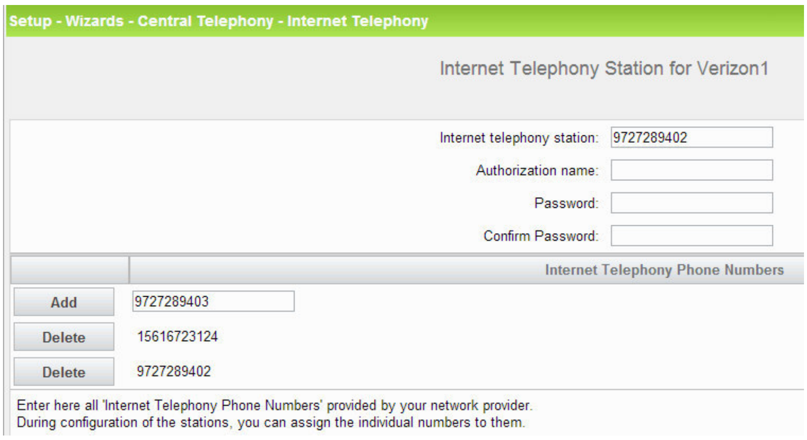
After inputting the Domain Name, the Proxy Provider information, press the Ok button to accept the changes and then press Ok& Next to move to the next form.

11.1.7. 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

11.1.8. 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.



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.

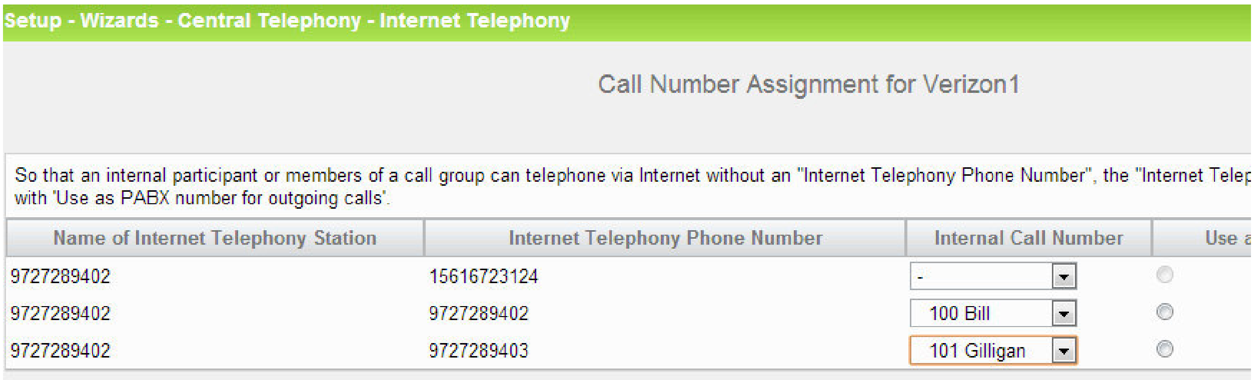
Press the OK & Next button

11.1.9. 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.

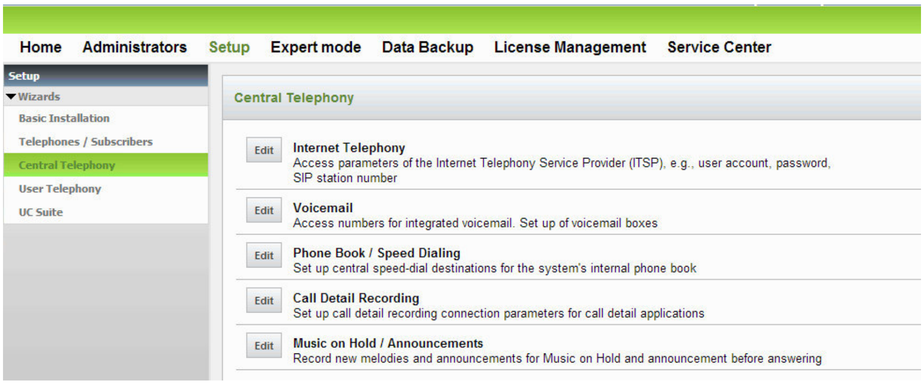
You may select one of the numbers as the default entry. The selected telephone number will be displayed for all outbound calls placed by stations that are not assigned an Internet Telephony Phone Number.



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.

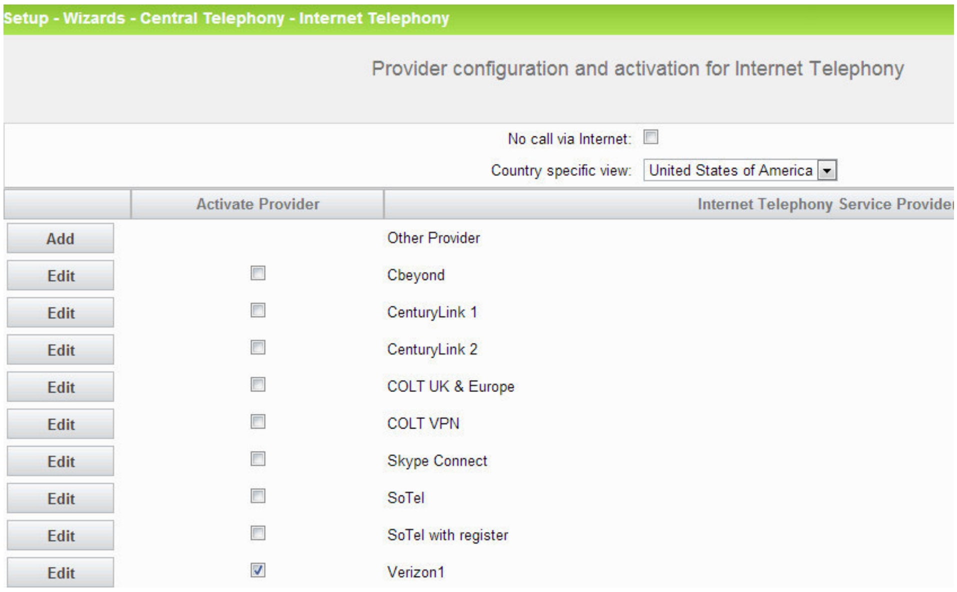
11.1.10. Select the Verizon ITSP profile

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



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

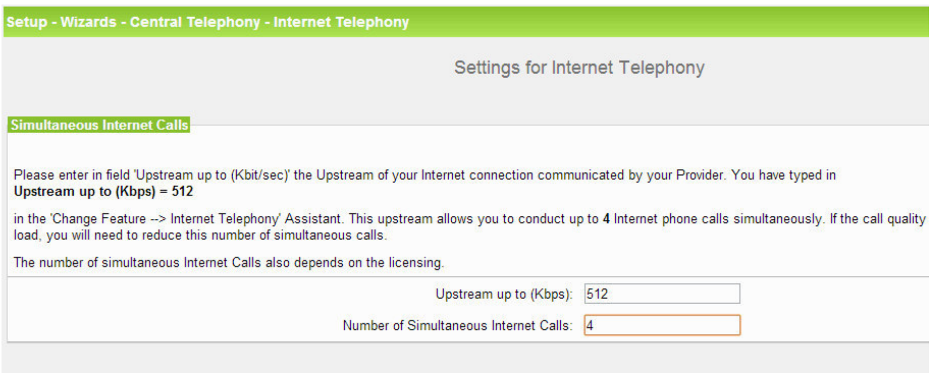


Uncheck the “No call via Internet” box to display the ITSPs for the United States

Enable the Verizon ITSP entry and press Ok & Next to move to the next step

11.1.11. Define the number of concurrent voice sessions

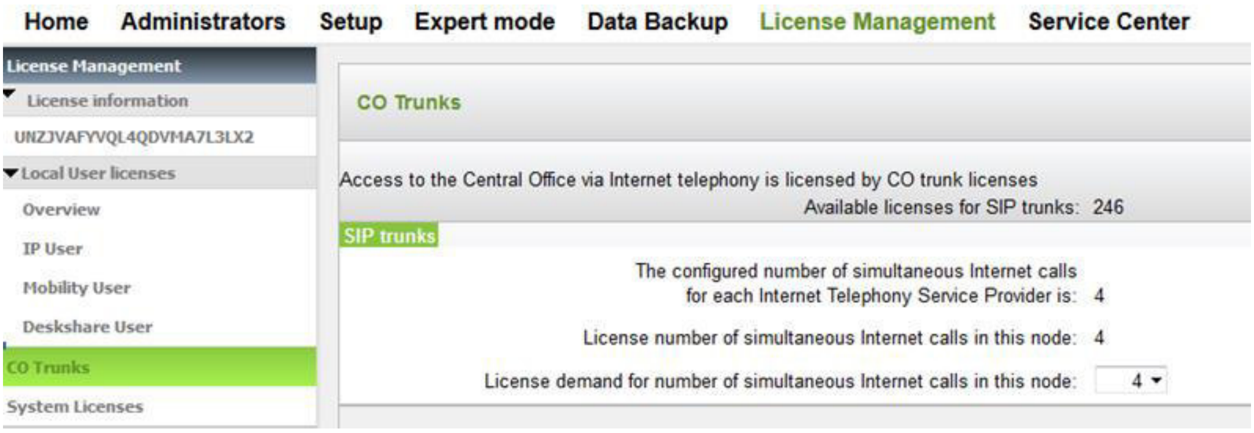
The **Settings for Internet Telephony** form is used to define the number of concurrent voice sessions that will be supported. The maximum number of sessions supported by the X3, X5 and X8 is 60. The maximum number of sessions supported by the Business S system is 120. This assumes that the internet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



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

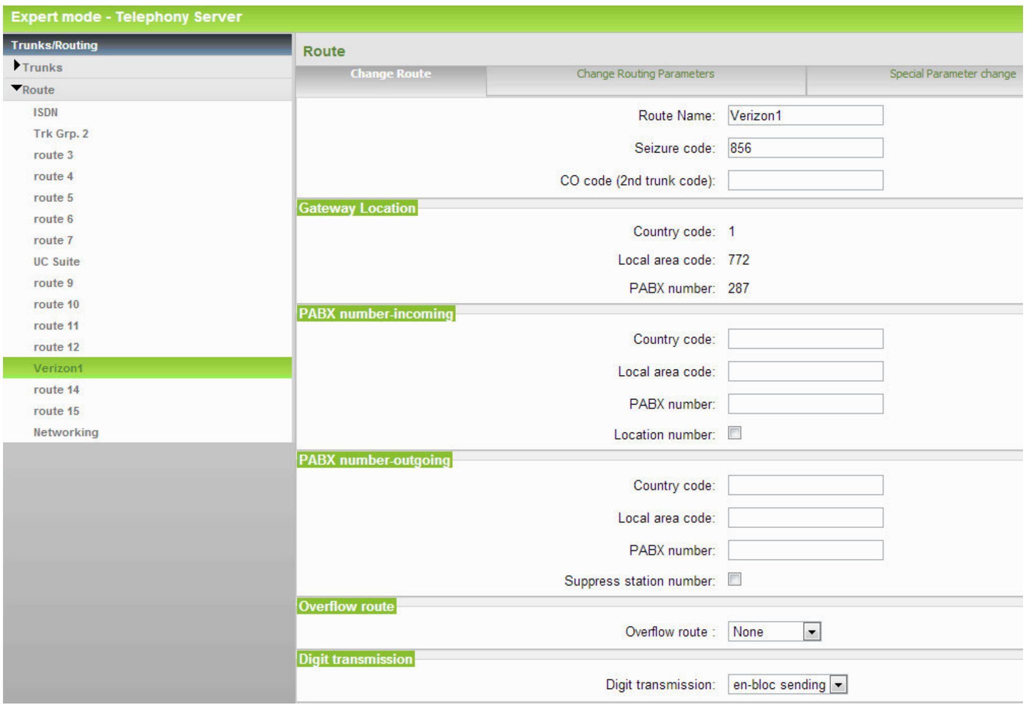
Press Ok & Next to move to the next step

Please note that each ISP session will require an OSBiz V1 S2M/SIP Trunk license. The licenses will have to be enabled under the License Management > CO Trunks

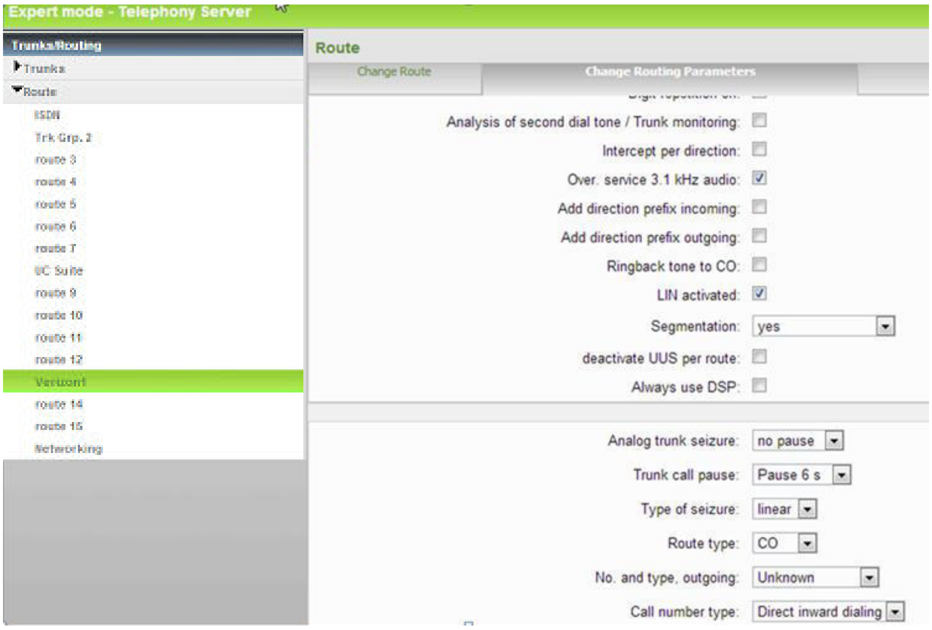


11.1.12. Confirm the Trunk Route Settings

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



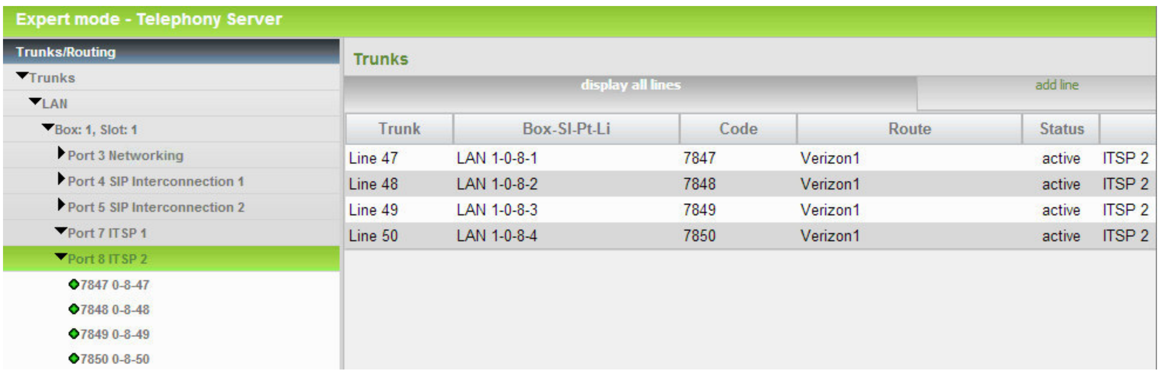
Select the Change Route Parameters button to display the form. Confirm or revise your entries to match the form displayed below.



Press the Apply button after making the changes

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

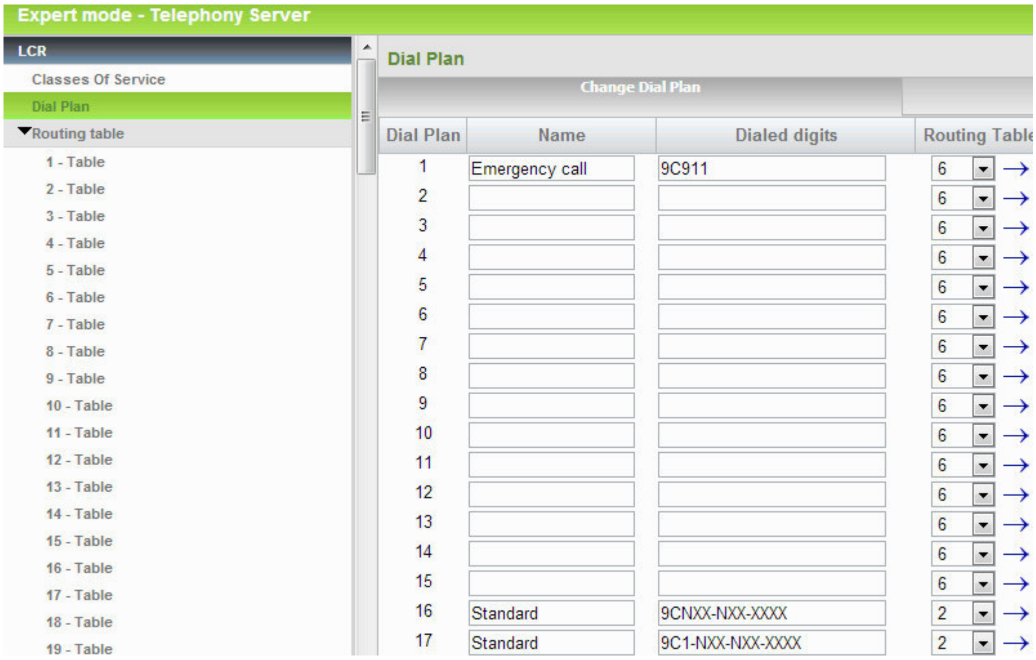
Under the Expert Mode > Telephony Server > Trunks & Routing >Trunks > LAN section you can select the “Port – ITSP” that is associated with the configured provider to confirm the number of channels added in the wizard have been added to the Verizon group. After confirming the quantity is correct exit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



11.1.14. Least Cost Routing Dial Plan

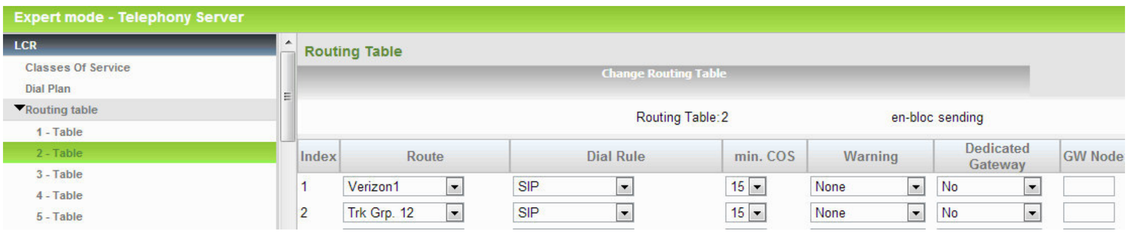
This step is required to allow the station user to dial a PSTN telephone number and have the outbound call route over the selected Verizon SIP trunk group.

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.

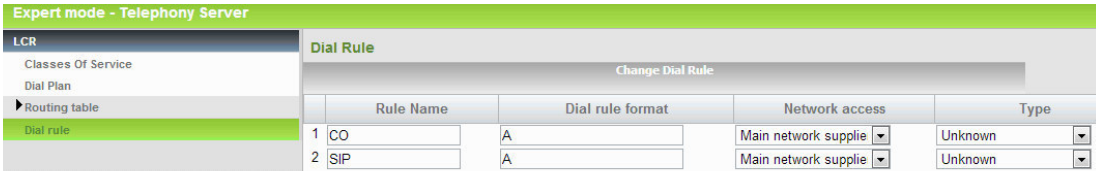


The information under Expert Mode > Telephony Server >LCR >Routing Table will be displayed. In the example below the Verizon1 group is primary selection. The dial rule is SIP and the minimum COS is 15.

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



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”



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 “XXXX” is the entry for field four

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

12. 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).

13. 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	**TCAP**	Transaction Capabilities Application Part (SS7)
FQDN	Fully Qualified Domain Name	**TCP**	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
MTP	Message Transfer Part (SS7)	**V-MG**	Video Media Gateway
NAPTR	Naming Authority Pointer Records	**XML**	Extensible Markup Language

About Unify

Unify is one of the world's leading communications software and services firms, providing integrated communications solutions for approximately 75 percent of the Fortune Global 500. Our solutions unify multiple networks, devices and applications into one easy-to-use platform that allows teams to engage in rich and meaningful conversations. The result is a transformation of how the enterprise communicates and collaborates that amplifies collective effort, energizes the business, and enhances business performance. Unify has a strong heritage of product reliability, innovation, open standards and security.

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