Harmonize your enterprise

HiPath 3000 V9

Tutorial

Verizon IP Trunk and IP Contac Center Services

Version 1.0

HiPath 3000 Application Note

Verizon IP Trunk and IP Contact Center Services

HiPath 3000 Version 9.0

August 27, 2013

V1.0

Date	Version	Modified by	Section Affected	Comments
07/28/2013	Initial version.01		All	
08/9/2013	Initial version.02		All	
08/26/2013	Initial version .03		Added changes received on 8/20/2013	
08/27/2013	Initial version .04		Rewrote ITSP channel section	
08/27/2013	1.0			

Availability and technical specifications are subject to change without notice. The information provided in this document contains general descriptions or characteristics of performance which in case of actual use do not always apply as described or which may change as a result of further development of the products. An obligation to provide the respective characteristics shall only exist if expressly agreed in the terms of contract.

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1 Introduction

This application note highlights the use and setup for HiPath 3000 V9 with Verizon IP Trunks and Verizon IP Contact Center Trunk Services (IPCC). HiPath 3000 V9 became generally available on July 13, 2012. Verizon has certified the HiPath 3000 V9 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 a HiPath 3000 system.
HiPath 3000 and Verizon IP Trunk Services

HiPath 3000 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,
HiPath 3000 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. HiPath 3000 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 HiPath 3000 are the perfect solution for optimizing costs and business processes

The HiPath 3000 solution can provide optimum customer flexibility when combined with Verizon IP Trunk Services. The HiPath 3000 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 HiPath 3000 is able to take advantage of existing network services and drastically cut communication costs. HiPath 3000 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 HiPath 3000 V9 system meets Verizon support expectations when implemented following the provisioning outlined in this brief with exceptions where noted.

Implementations of HiPath 3000 V9 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 HiPath 3000 Voice solution components

HiPath 3000/5000 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. HiPath 3000 enables any combination of IP, analog and digital telephones, as well as PC clients and cordless telephones.

HiPath 3000 Model 3800
HiPath 3000 Model 3800 Base cabinet
HiPath 3000 Model 3800 HG1500 card for management of VoIP solutions and connectivity to ISP SIP Trunking resources
HiPath 3000 Model 3800 peripheral circuit cards for connectivity to TDM subscribers and trunk resources
Mains Power Cord, USA variant
OpenStage TDM and or IP telephones

HiPath 3000 Model 3500
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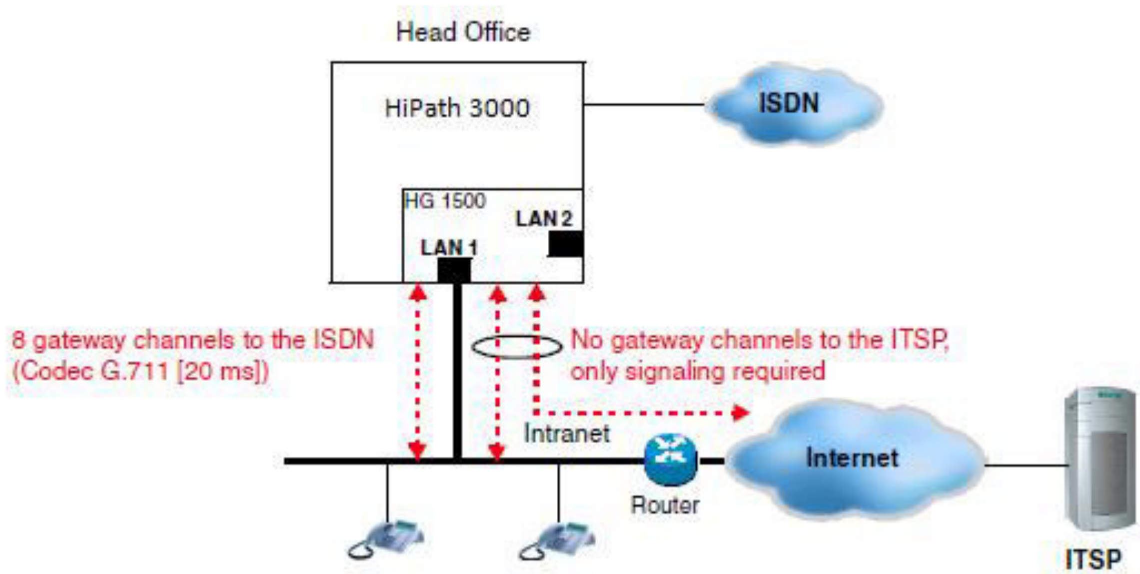
HiPath 3000 Model 3500 peripheral circuit cards for connectivity to TDM subscribers and trunk resources
Mains Power Cord, USA variant
OpenStage TDM and or IP telephones

5 HiPath 3000 Reference Architecture

HiPath 3000 V9 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 HiPath 3000 V9, 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.

Replace with HP3000 network architecture diagram



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 HiPath 3000 solution over Private IP or Internet Dedicated Access facilities. Due to the extensibility of the Verizon VoIP network, now HiPath 3000 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 HiPath 3000 V9 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 HiPath 3000 V9 provisioning is required).

7 Verizon IP Trunk Certification

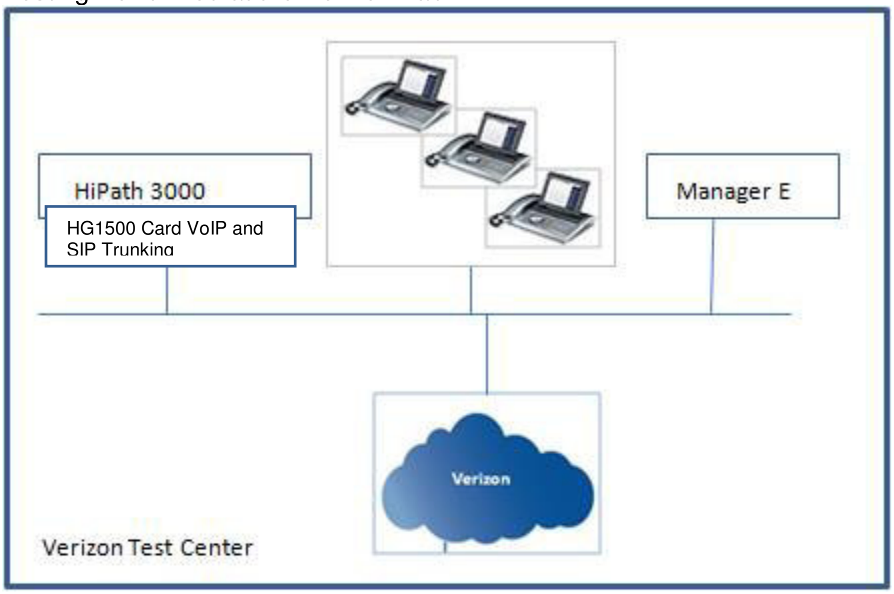
This section outlines the actual HiPath 3000 elements used in the Verizon IP Trunk Certification process for Verizon US IP Trunks, EMEA IP Trunks and IPCC Trunks:

Example HiPath 3000 Voice Bill of Materials (as certified):
Verizon HiPath 3000 Lab Bill of Materials

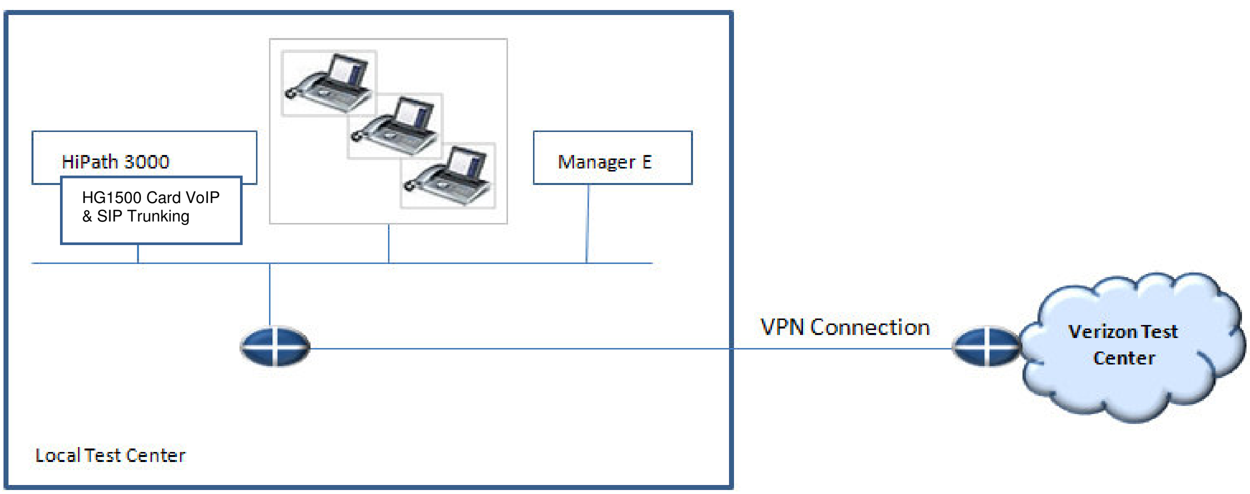
Order Number	Description	Quantity
L30251U0600G559	HP 3500 V9 SYSTEMBOX 2 S0 / 8 UP0/E/4A/B	1
L30251U0600A594	ANALOG TRUNK MODULE (TLANI4R) 3300/3500	1
L30250F0600C155	OPENSTAGE 40 - HFA LAVA	2
L30250F0600C175	OPENSTAGE 15 T - LAVA	1
L30251U0600A741	HG1500 V3.0 PKG W8 BCHN FOR 3300/3500 V9	1

7.1 HiPath 3000 High Level Configuration (as certified):

Testing Performed at the Verizon Lab



Testing performed by Local North American and European Labs



8 LOAD ISSUES/CONCERNS:

The following issues were identified during CPE Interop testing.

The HiPath 3000 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 HiPath 3000 V9 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 HiPath 3000 V9 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.

The HiPath 3000 V9 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 HiPath 3000) 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 HiPath 3000 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 the HiPath 3000 V9.

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. <u>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.</u>

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. <u>However</u>, 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 HiPath 3000 Configuration – Verizon IP Trunks

This section will outline the steps for the Configuration of the Verizon SIP Trunks with the HiPath 3000 Version 9 system. The configuration requires the creation or modification of the database within the HiPath 3000 system and the HG1500 VoIP Interface card.

The documented steps assume that the system administrator is a certified technician on the HiPath 3000 platform.

The configuration information assumes that the system connection to the Internet using LAN Port 1 on the HG1500 card has been established.

11.1 Configure the HiPath 3000 System Information

This section will provide the configuration steps for programming the HiPath 3000 system

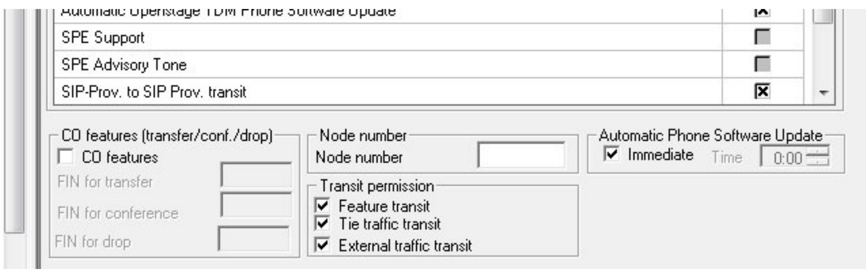
11.1.1 Transfer database to the Manager E tool from the system

After completing the programming for the HG1500 card, using the Manager E tool, transfer a new database from the system to the Manager E tool. Select File > Transfer to access the File transfer form.

Enter the IP address of the HiPath 3000 system and select the System > PC option.

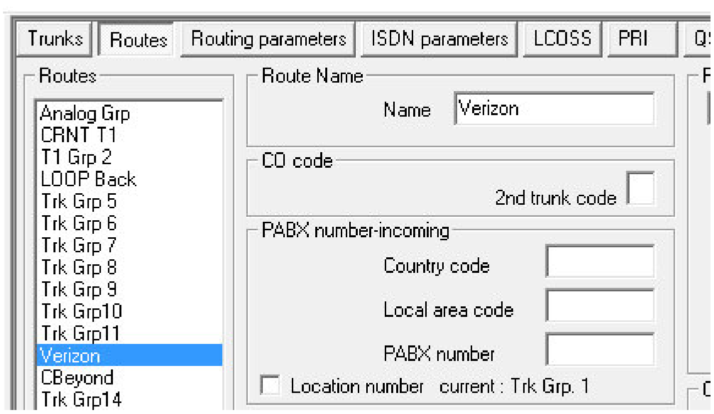
11.1.2 Configure the system Parameters

Under the system parameters > Flags menu insure that the following flags are enabled



11.1.3 Configure the Trunk Route Settings

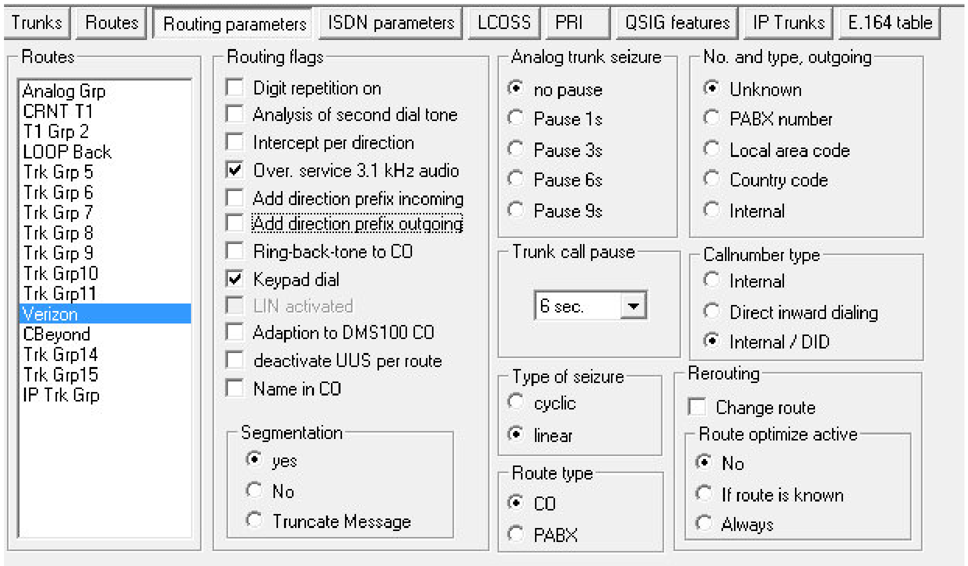
Under the Lines and Networking > Routes menu select Trunk Grp. 12 and change the name of the Route to Verizon.



Press the Apply button after making the changes

11.1.4 Configure the Trunk Route Parameters

Under the Lines and Networking > Routing Parameters menu insure that the displayed settings are in place
Press the Apply button after making the changes



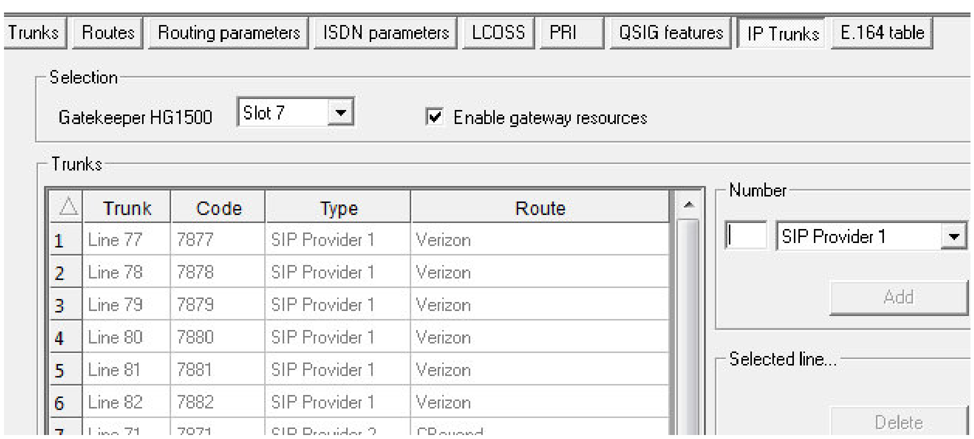
11.1.5 Configure the IP Trunk Channel Settings

Under the Lines and networking > IP Trunks confirm the quantity of ITSP/SIP Trunks for the Verizon IP trunk group is correct or needs to be modified. An ITSP trunk is required for every ITSP call.

If you need to add to the displayed qty for SIP provider 1, enter the additional number of channels in the Number entry field and press the Add button.

If you need to delete channels, highlight the channels in the table and press the Delete button.

After making any changes to the configuration press the Apply button to save your changes.



Additional Notes

A SIP call placed from a TDM digital phone to the PSTN over an ITSP/SIP trunk would require 1 channel.

An incoming call from the PSTN on a SIP trunk to the integrated voicemail card, analog port or TDM digital phone would require 1 channel.

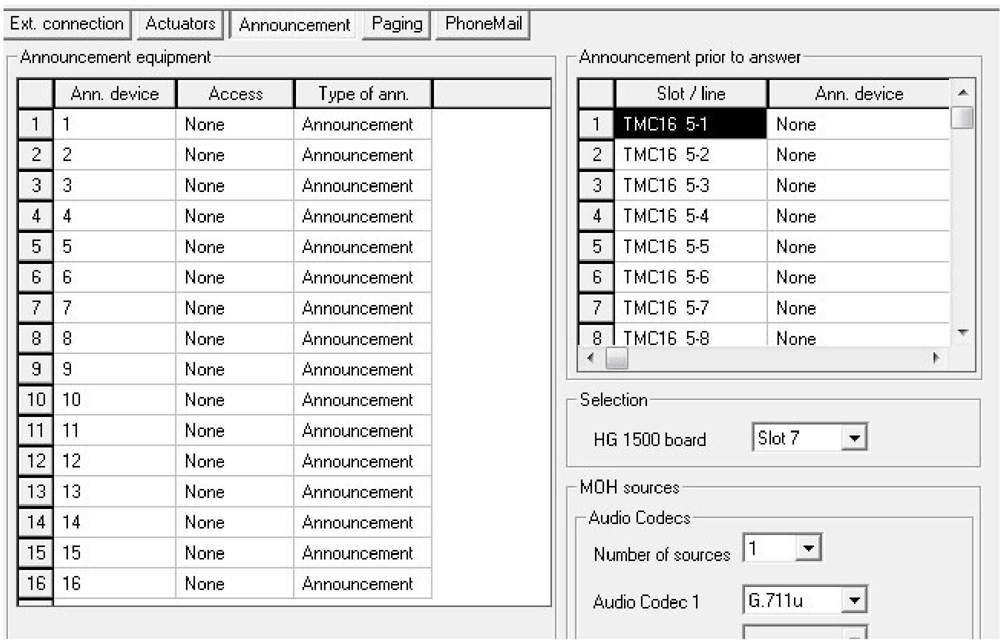
A call by an IP phone from SIP trunk or to the SIP trunk would not require a channel.

11.1.6 Configure channel(s) to provide MOH to SIP trunk calls that are placed on hold

Under the Auxiliary Equipment > Announcement menu select the slot number where the HG1500 card is seated
Under the MOH sources > Audio Codecs select the number of sources from 1 to 5. Each source will support up to 10 media stream sessions for MOH

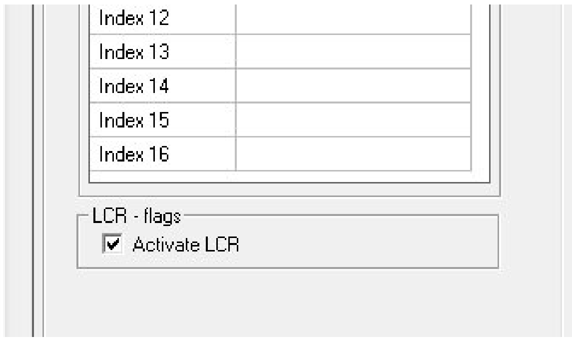
Leave the Audio Codec 1 set for G.711u

After making any changes to the configuration press the Apply button to save your changes.



11.1.7 Enable Least Cost Routing (LCR)

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 Least Cost Routing > Flags and COS > LCR flags, insure that the “Activate LCR” check box is enabled.
After making any changes to the configuration press the Apply button to save your changes.



11.1.8 Create or confirm that an LCR Out dial rule is configured

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 “-“.

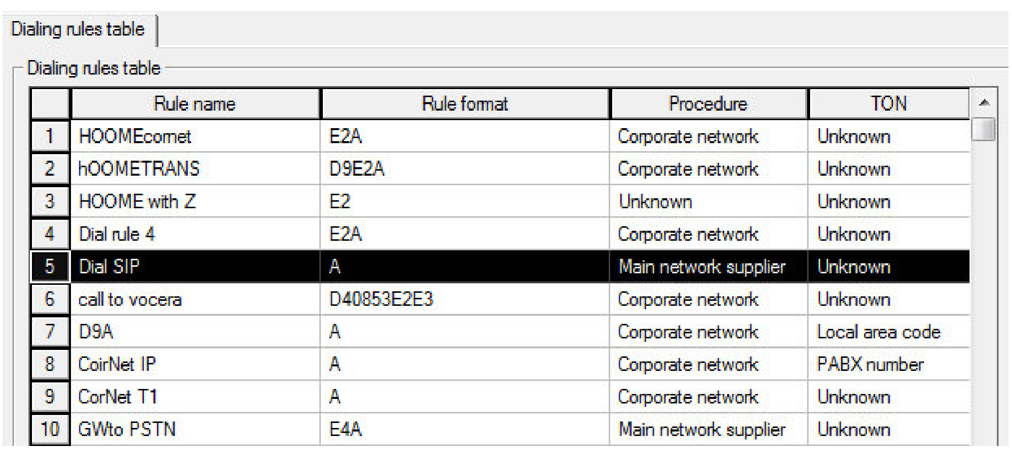
For example in the Dial plan string 9C1-NXX-XXX-XXXX
The “9” is the LCR access code and is field 1
The “C” is a separator and will return simulated dial tone to the user
The “1” is the entry in field one
The “-“is a separator
The “NXX” is the entry for field two
The “-“is a separator
The “XXX” is the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

Under the Least Cost Routing > Dial plan > Dialing Rules table, add the new rule outlined above or confirm that a default out dial rule is in place.

If the out dial rule is in place press the Apply button and then close button.

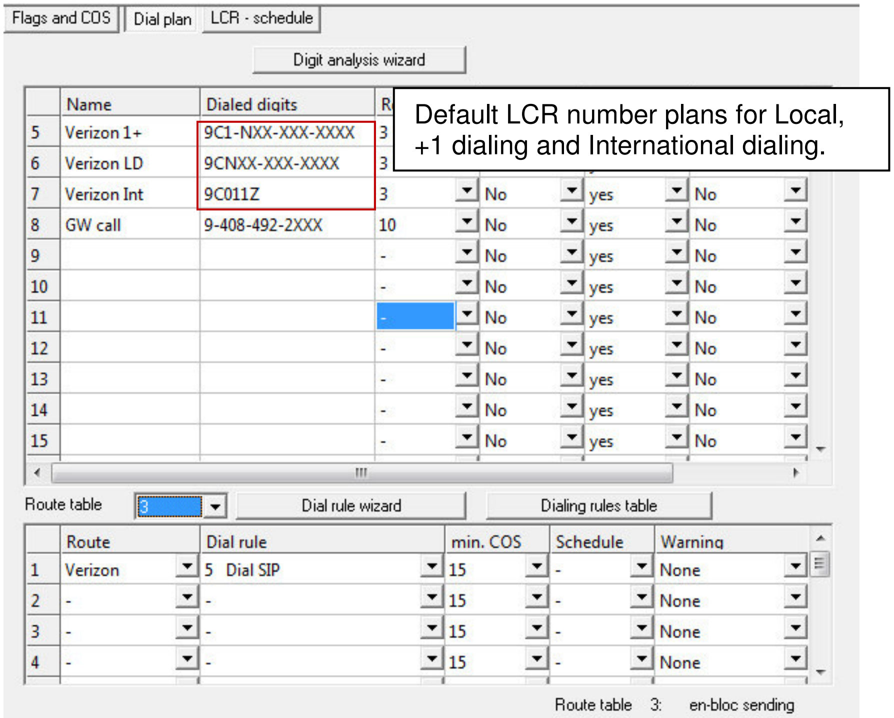

Dialing rules table

Dialing rules table

	Rule name	Rule format	Procedure	TON
1	HOOMEcomet	E2A	Corporate network	Unknown
2	hOOMETRANS	D9E2A	Corporate network	Unknown
3	HOOME with Z	E2	Unknown	Unknown
4	Dial rule 4	E2A	Corporate network	Unknown
5	Dial SIP	A	Main network supplier	Unknown
6	call to vocera	D40853E2E3	Corporate network	Unknown
7	D9A	A	Corporate network	Local area code
8	CoirNet IP	A	Corporate network	PABX number
9	CorNet T1	A	Corporate network	Unknown
10	GWto PSTN	E4A	Main network supplier	Unknown

11.1.9 Create LCR Dial plan and LCR Route Group

Under the Least Cost Routing > Dialplan confirm the default number plan entries are present. If the default entries are not present then they would need to be added to the LCR Dial plan table.



The information in the name column should be entered by the system administrator for identification purposes.

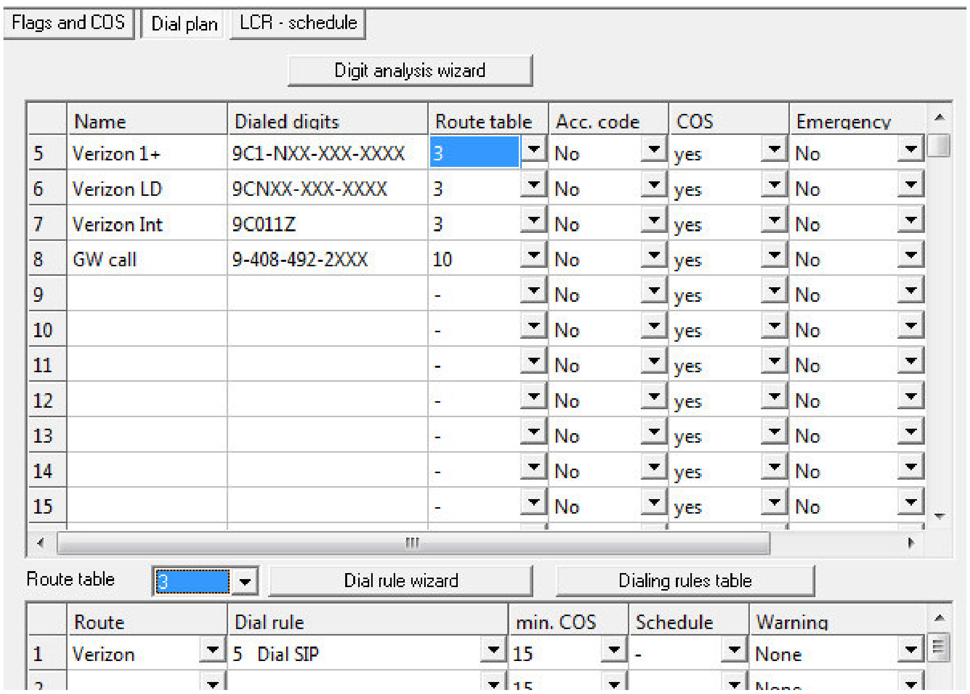
Please note that if the system is to be connected to an OpenScape Office HX server and the customer will be using the Fax application supported by the UC solution, the Minimum LCR COS level must be set to 1.

After entering the dial plan information select an unused Route table to define the trunk route group and out dial rule. In the example, Route Table 3 has been selected.

In the lower portion of the form select Route Table 3 from the Route Table list box

In selection 1 > Route of Route Table 3 enter the name of Route Table defined in step 11.2.3. In the example below the name assigned is Verizon.

In selection 1 > Dial Rule, enter the name of the out dial rule. In the example below the Dial Rule name is Dial SIP.



	Name	Dialed digits	Route table	Acc. code	COS	Emergency
5	Verizon 1+	9C1-NXX-XXX-XXXX	3	No	yes	No
6	Verizon LD	9CNXX-XXX-XXXX	3	No	yes	No
7	Verizon Int	9C011Z	3	No	yes	No
8	GW call	9-408-492-2XXX	10	No	yes	No
9			-	No	yes	No
10			-	No	yes	No
11			-	No	yes	No
12			-	No	yes	No
13			-	No	yes	No
14			-	No	yes	No
15			-	No	yes	No

	Route	Dial rule	min. COS	Schedule	Warning
1	Verizon	5 Dial SIP	15	-	None

11.2 HG1500 VoIP GW Configuration

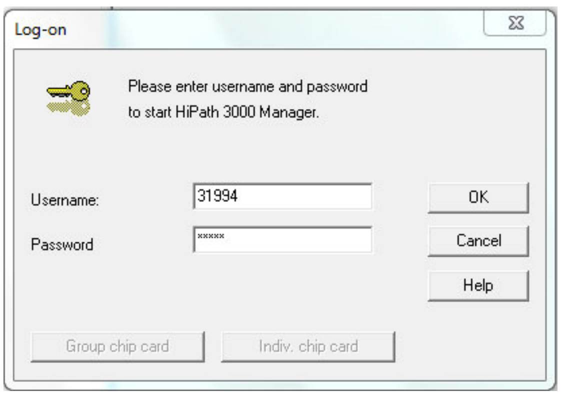
11.2.1 Connections to the Verizon Network

Prior to configuring the HG1500 card a VPN connection will need to be established from the customer's premise to the Verizon network. This will require that the settings for the default Router, DNS and Static STUN settings are configured within the HG1500 card for proper operation with the local VPN router.

Please note that only LAN Port 1 on the HG1500 card may be used to connect to the Gateway router to support the ITSP/SIP connections. LAN port 1 is also used to support connectivity for VoIP solutions as well.

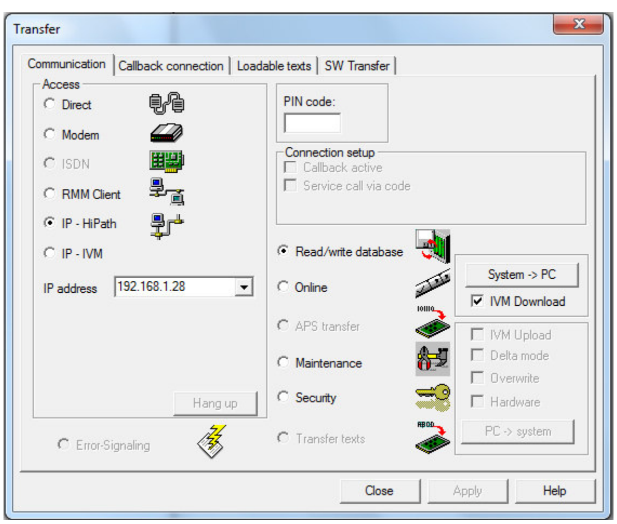
11.2.2 Log on to the system management tool

Load the Manager E program and log in using the default user name 31994 and the default password 31994



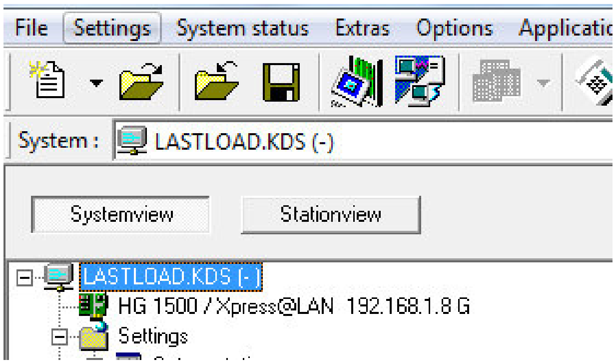
11.2.3 Transfer a copy of the database file to the Manager E program

Select File > Transfer to access the File transfer form. Enter the IP address of the HiPath 3000 system and select the System →PC option.



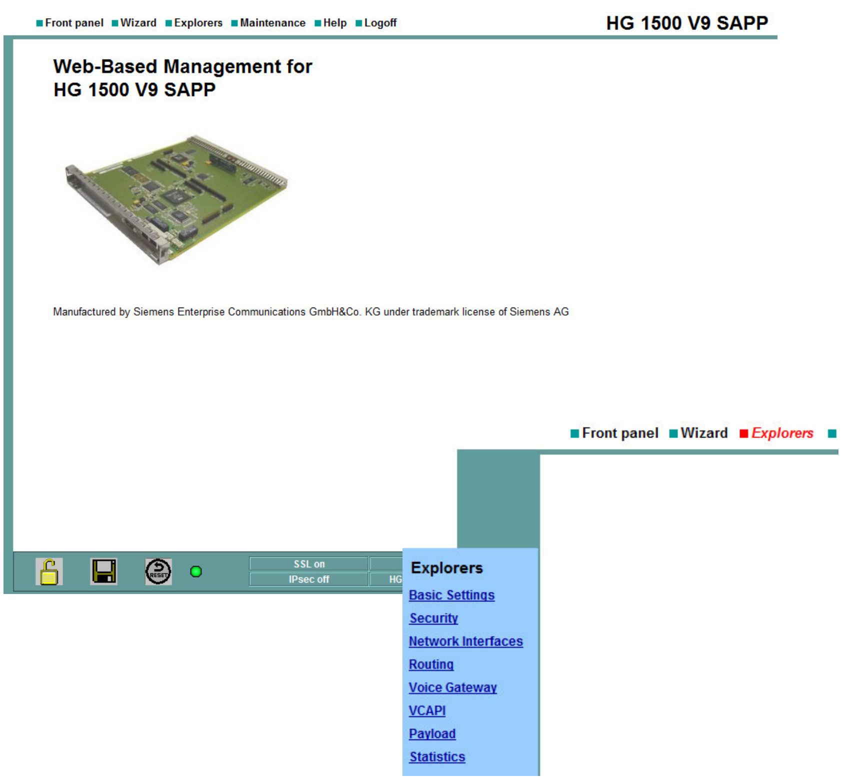
11.2.4 Access the Web based tool for the HG1500 VoIP GW card

Once the customer database file is loaded, press the HG1500/Xpress@Lan link to access the HG1500 card.
Please note that the IP address for the HiPath system used in the lab is 192.168.1.28. The IP address of the HG1500 card used in the lab is 192.168.1.8.



11.2.5 Open Up the HG1500 Main Menu

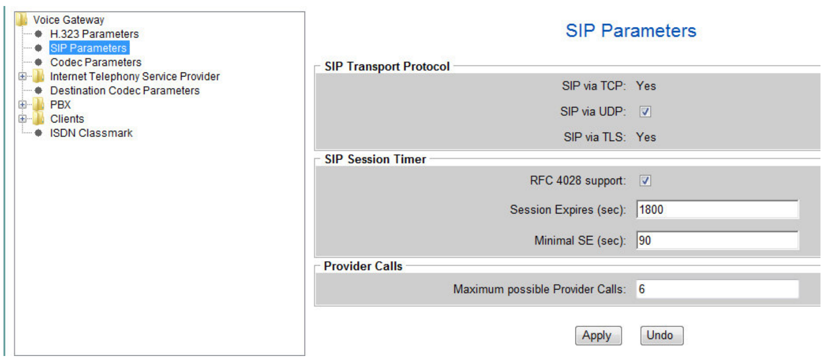
When the HG1500 page appears, press the Explorer Menu choice to open the menu selections. Once the menu is displayed press the Voice Gateway selection from the menu to begin the configuration. Please note that any time the diskette icon is red, this means that the enter information should be saved by pressing the diskette icon.



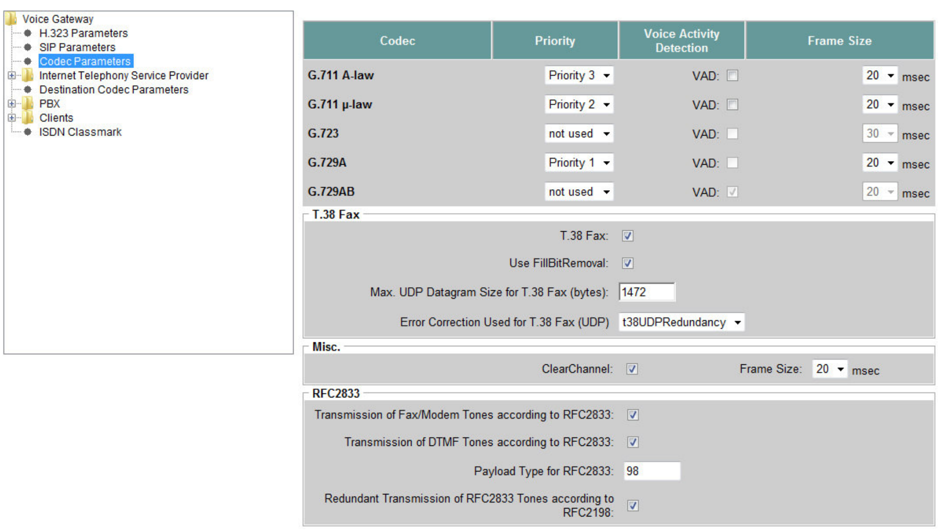
11.2.6 Setting Up the General SIP Trunk Parameters

When the sub menu appears, select the Voice Gateway > then right click SIP Parameters selection and select “edit SIP parameters.

Insure that the SIP transport protocol flags are set as displayed below
Insure that the SIP Session Timer flags are set as displayed below
Under the Provider Calls section, enter the qty of concurrent calls that will be supported by the session. The example below shows that 6 simultaneous calls will be supported. The maximum number of sessions is 32. After making the changes press the Apply button to write the changes to the database <u>and then press the SAVE button at the bottom of the page.</u>



Right Click on the 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 write the changes to the database <u>and then press the SAVE button at the bottom of the page.</u>

11.2.7 Set STUN Configuration Parameter

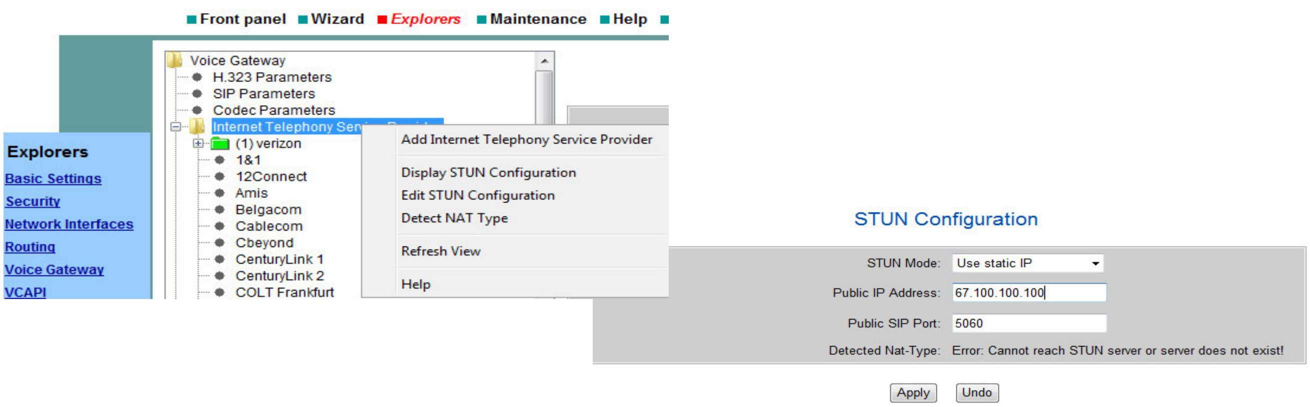
Right click in the Internet Telephony Service Provider menu and select "Edit Stun Configuration "

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

Leave the Public SIP Port of 5060 in place

Click on the Apply button to write the information to the data base and then click on the "disk icon" to save the data base.

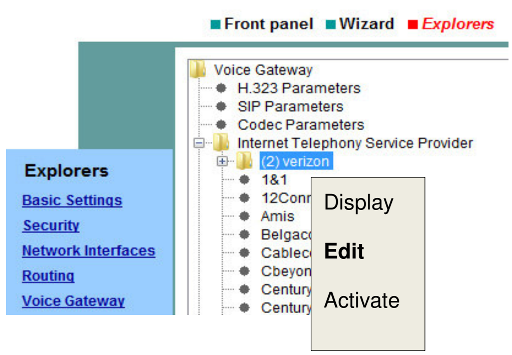
Please note that this setting is required even though STUN is not required by Verizon



11.2.8 Select the Verizon ITSP profile

Click on the plus sign next to the Internet Telephony Service Provider Menu choice to display the list of ITSP profiles that have been preloaded in the database.

Select and right click on the Verizon profile and select the EDIT prompt to view and edit the profile information



11.2.9 Revise the Verizon ITSP profile

The profile for the Verizon SIP trunks has already been loaded into the HG1500 card and contains the low level parameter settings. The domain information, proxy information and STN settings will need to be entered.

Click on the Enable Provider check box to enable the Verizon profile

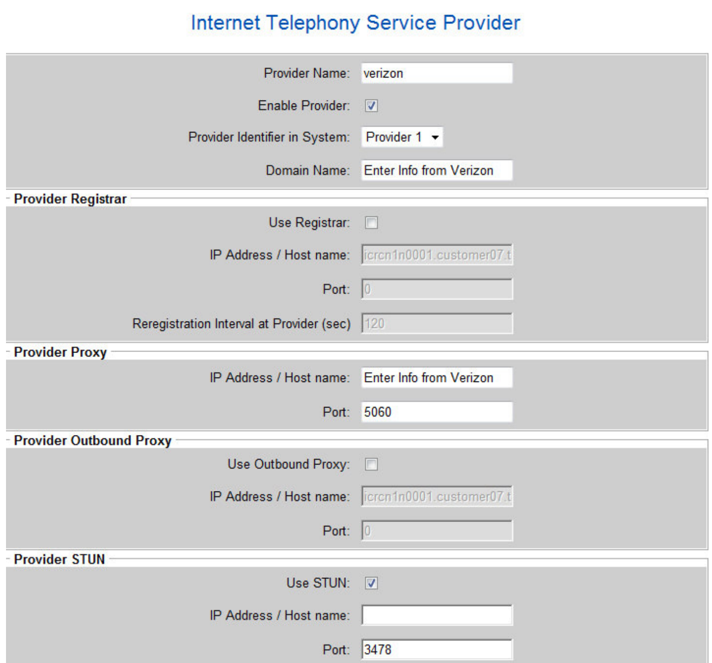
Select the Provider Identifier (Trunk Route Group) that will be assigned to the Provider. Typically Provider 1 should be selected unless the system will be configured with multiple ITSP providers. By Selecting the Provider Identifier the system will create a Trunk Route Group within the Manage E database that will be required when completing the Least Cost Routing configuration step.

In the Domain 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.

Click on the Apply button to write the information to the data base and then click on the “disk icon” to save the data base.

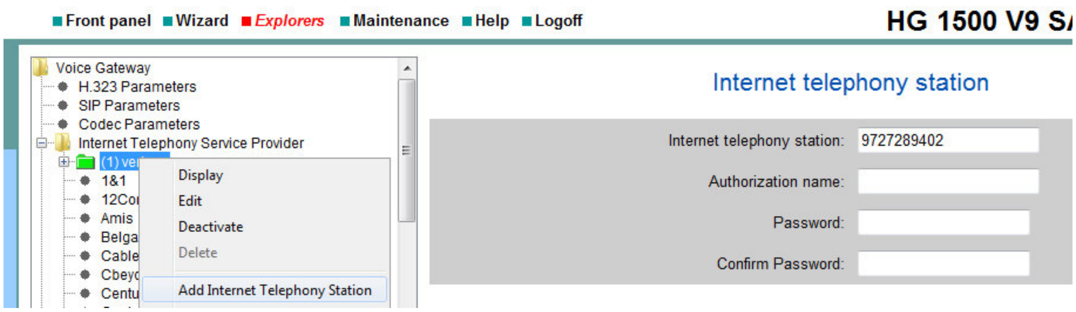


11.2.10 Add the Internet Telephony Station Number

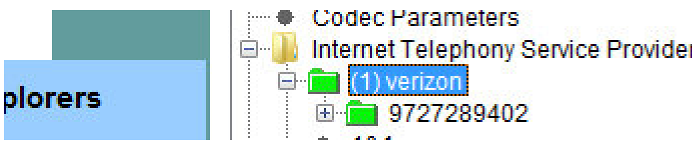
Right Click on the Internet Telephony Service Provider menu and select "Add Internet Telephony Station" from the menu.

Insert the Main Telephone number in the Internet telephony station field.

Click on the Apply button to write the information to the data base and then click on the "disk icon" to save the data base.



Once the Internet Telephony Station Number has been added click on the Plus sign to confirm the number has been added to the Verizon profile.



11.2.11 Add the MSNs (Main Subscriber Numbers) to the database

Right Click on the plus sign next to the Internet Telephony Station number to display the MSN menu item

Right click on the MSN menu and select Add MSN

Fill in the 10 or 11 digit number for the first MSN provided by Verizon. The MSN represents the series of telephone numbers that will be used for DID and outgoing identification purposes. Typically the first MSN number is the same as the Internet Telephony Station Number.

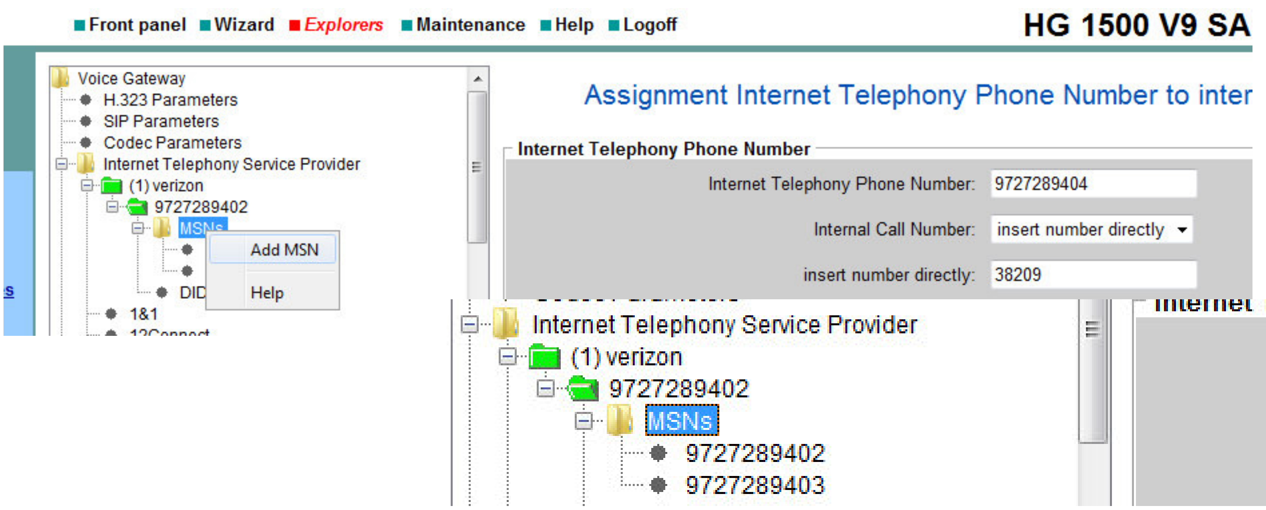
Select the “insert number directly” choice from the Internal Call Number list box

Enter a valid internal station number in the “Insert number directly” field

Check the “default entry” check box if the entered telephone number will be displayed for stations not assigned an MSN number on outbound calls.

Click on the Apply button to write the information to the data base and then click on the “disk icon” to save the data base if the disk icon is red.

Repeat the process to assign the balance of the MSN numbers to the system.



Please note that as an alternative a DID range may be created to assigned the MSN to the station number. MSN creation and number assignments can be configured to be done automatically when creating a DID range.

After confirming that the MSN numbers have been configured correctly click on the “disk icon” to save the configuration

After completing the HG1500 portion of the configuration the HiPath 3000 system should be restarted. This step will also restart the HG1500 card.

12 High Level Troubleshooting HiPath 3000 and IP Trunks

Refer to the HiPath 3000, Service Manual, Service Documentation, for HiPath 3000 trouble shooting steps. The latest service documentation maybe found via on-line web portal (SEBA).

13 Additional Documentation References

HiPath 3000 General Information
http://wiki.siemens-enterprise.com/wiki/HiPath_3000

HiPath 3000 and SIP Provider Information
http://wiki.siemens-enterprise.com/wiki/Collaboration_with_VoIP_Providers

Network Configuration for VoIP Providers
http://wiki.siemens-enterprise.com/wiki/Network_Configuration_for_VoIP_Providers

From the Electronic Documentation Site
HiPath 3000/5000 V9, Service Documentation, Issue 8 (HTML) or PDF
HiPath 3000/5000 V9, HG 1500 V3.0, Administrator Documentation, Issue 7 (HTML) or PDF
HiPath 3000/5000 V9, Manager E, Administrator Documentation, Issue 9 (HTML) or PDF

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

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