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VOCAL Test Plan

<u>Monet Project</u>

Rev	Date	Originator	Comments
1.0	7/20/2000	Testing&Integration Team	Draft, Based on PRD Rev.1.2
1.1	7/25/2000	Testing&Integration Team	Revised after review meeting, based on PRD Rev.1.4
1.2	7/31/2000	Larry Trovinger	Incorporate CL and CE comments.
1.3	7/31/2000	Anthony Nguyen	added 4.4, and updates to 4.5, 4.7
1.4	8/2/2000	Larry Trovinger	Cisco comments incorporated.
1.5	8/2/2000	Enlai Chu	Modified section 3 after review meeting
1.6	8/3/2000	Oscar Thomas	Modified section 5 after review meeting
1.7	8/3/2000	Anthony Nguyen	Modified section 4, add software versions to 1.3
1.8	8/3/2000	Tina Zhang	Modified section 6, add two feature matrixes, and add software versions to 1.8
1.9	8/3/2000	Enlai Chu	Modified section 3 after review meeting
1.10	8/4/2000	Oscar Thomas	Modified section 5 after review meeting
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1.12	8/4/2000	Larry Trovinger	Added negative test cases.
1.13	8/5/2000	Enlai Chu	Removed redundant test case from section 3.
1.14	8/8/2000	Tina Zhang	Updated section 6 after Luan's review.

Table of Contents

1 Product Overview

1.1 Definitions

This list of definitions and acronyms contains a limited number of vendor and customer specific entries.

Term	Definition
UA	User Agent
CPS	Calls Per Second
BHCA	Busy Hour Call Attempts
MTBF	Mean Time Before Failure
Transient Calls	Initial invite sent, but voice path not yet connected
RS	Redirect Server
MS	Marshal Server
FS	Feature Server
MGCP	MGCP Translator
MIND	MIND Billing Server
VM	Voice Mail
CDR	Call Detail Record Server
SNMP GUI/OSS	
PRD	Product Release Document

1.2 Master Solution Test Plan

Figure 1. Redundant System Configuration

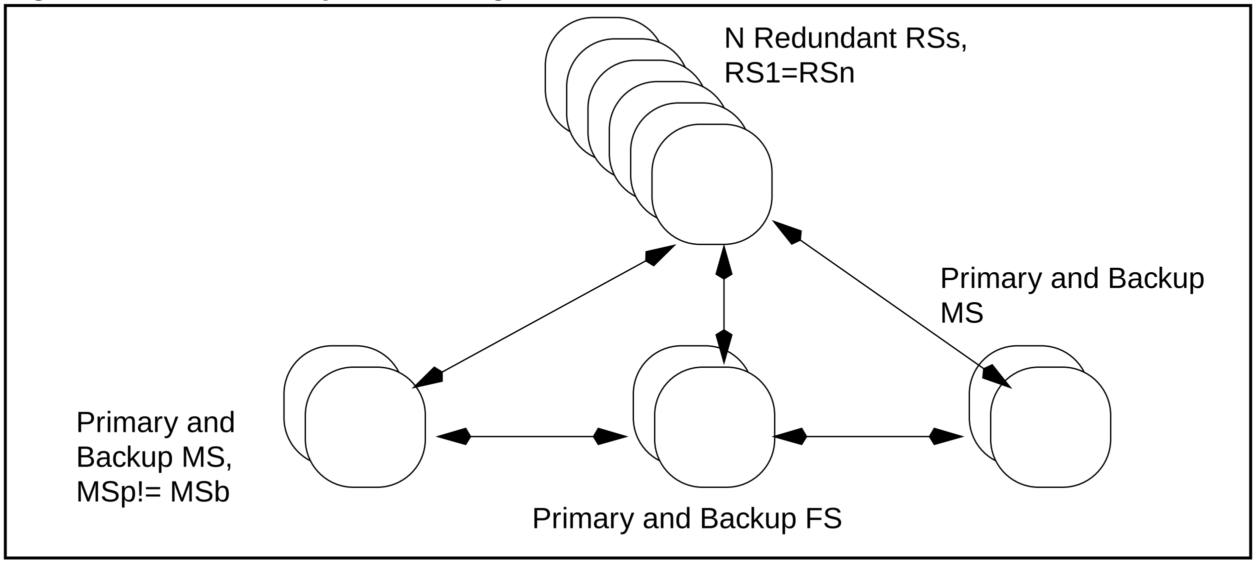
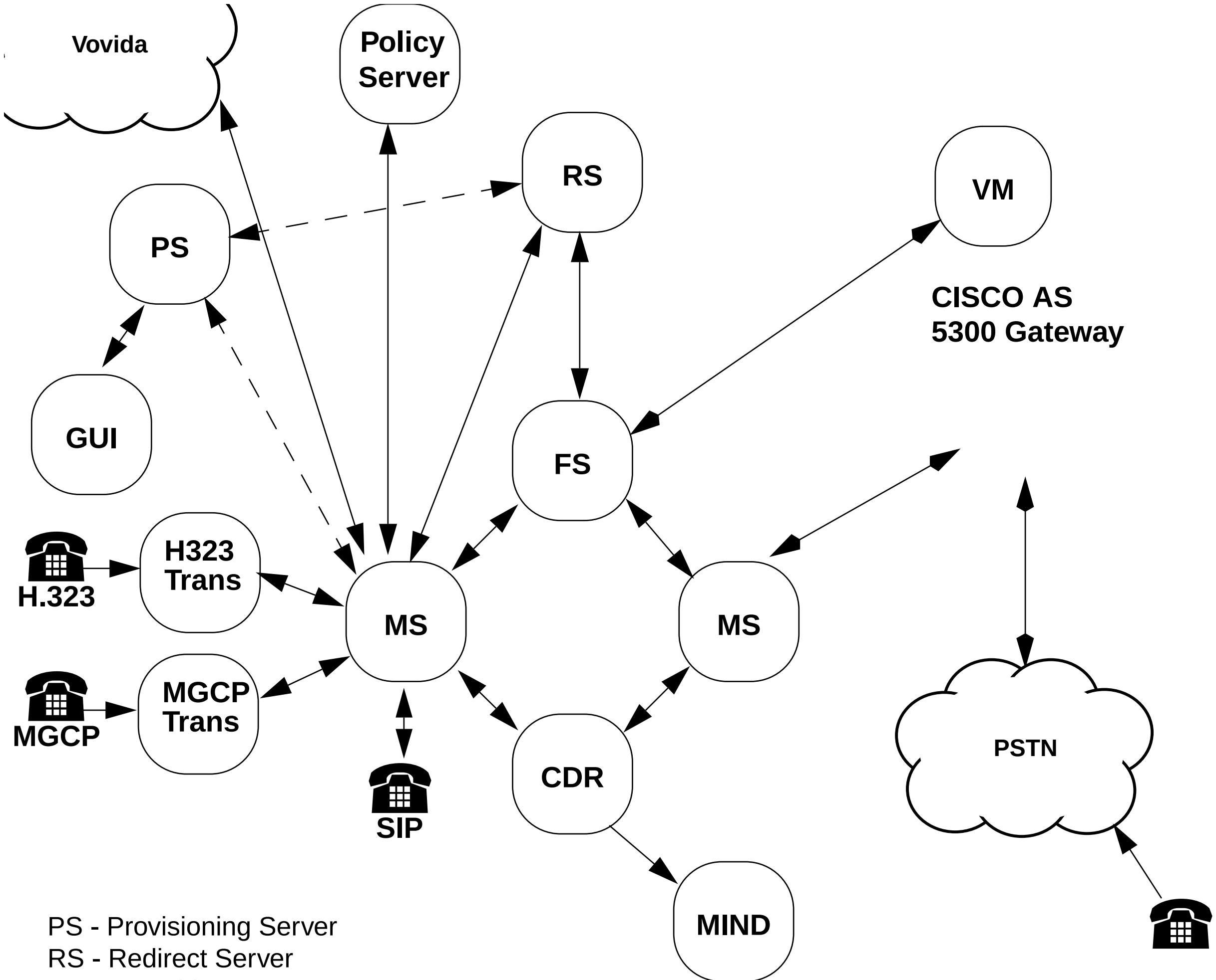


Figure 2. Customer Network Configuration



PS - Provisioning Server
RS - Redirect Server
GUI - Graphical User Interface
MS - Marshal Server
FS - Feature Server
CDR - Call Detail Record Server
VM - Voice Mail
PSTN - Public Switched Telephone Network
H.323 Trans - H.323 Translator
MGCP Trans - MGCP (Media Gateway Control Protocol) Translator

1.3 Testbed Equipment Requirements

The following equipment is required...

Test Equipment	Qty	Details
Reference H/W Platform	2	1 each: HP Netserver Lpr and Large Scale Pogo System. HP Netservers: 700Mhz PIII, 375RAM. Pogos: Dual PIII,
Vovida Vocal Soft-Switch		Version 1.0
Red Hat Linux		Release 6.2 (Zoot), Kernel 2.214
Microsoft Windows NT		Version 4.00.1381
Microsoft Internet Explorer 5.0		Version 5.00.2014.0216
Microsoft Netmeeting		Version 3.01 (4.4.3388)
Netscape Communicator		Version 4.72
Mind-iPhonEX		Version 3.10.040.5
Cisco 5300 Gateway		IOS Version 12.1.3T
Cisco 2600 Gateway		IOS Version 12.1.3T
Cisco Telecaster		Application Load ID: P0S3Z333 Boot Load ID: PC03K030
Telogy		
Voyant Innovox MCU		Voyant ReadiVoice Conferencing System, v2-00-0

1.4 Test Tool Requirements

The following test tools are required.

Test Tools	Qty	Details
Call Generator	1	Cisco AS5300, IOS image containing Callgen
Proxy Tool	1	To generate users
Vovida Load Generators	1	UAs

2 System Requirements

2.1 Scalability

2.1.1 Capacity Requirement Tests (Reference to PRD-1.4, Sec-2.1.1)

Rls 1.0 must support up to 56 calls per second busy hour traffic and 100,000 endpoints. Assume an average 3 minutes per call on 10 percent of the endpoints (.10*100,000/180=56cps).

2.1.1.1 Use Vovida Load Generator

Action:	• Generate 56 cps.
Result:	• Calls will be allowed to complete. Call-setup/teardown timing will remain acceptable.
Test Equipment for this test case: Vovida UAs in loadgen mode	
Tested by Vovida.	

2.1.1.2 Use Callgen

Action:	• Generate 8 cps.
Result:	• Calls will be allowed to complete. Call-setup/teardown timing will remain acceptable.
Test Equipment for this test case: Callgen to generate 8 cps	
Tested by Cisco.	

2.1.1.3 Medium Number of Endpoints

Action:	• Configure 20,000 endpoints • Change UA to answer random user selection or pick 5 - 10
Result:	• Calls will be allowed to complete. Call-setup/teardown timing will remain acceptable.
Test Equipment for this test case: Proxy tool to generate users Vovida UAs	
Tested by Vovida and Cisco.	

2.1.1.4 Large Number of Endpoints

Action:	• Configure 100,000 endpoints • Change UAs to answer random user selection or pick 5 - 10
Result:	• Calls will be allowed to complete. Call-setup/teardown timing will remain acceptable.
Test Equipment for this test case: Proxy tool to generate users	
Tested by Vovida.	

2.1.2 Capacity Failure Test (Reference to PRD-1.4, Sec-2.1.2)

The system must be capacity tested to failure on a reference hardware architecture (HP LPr and POGO) to determine actual system capacity under load.

Action:	• Generate 30 cps, 40 cps and so on.
Result:	• Record cps capacity at which system fails. Record nature of failure; i.e., which server.
Test Equipment for this test case: Use both Reference H/W platform and Large Scale System Use Vovida load generators	
Tested by Vovida.	

2.1.3 Capacity Failure Test Single System

Determine capacity failure points and characteristics on a basic system.

Action:	• Generate 10 cps, 20 cps and so on.
Result:	• Record cps capacity at which system fails. Record nature of failure; i.e., which server.
Test Equipment for this test case: Use Single System - 8 HP servers Use Vovida load generators	
Tested by Vovida.	

2.2 Reliability

2.2.1 Call Duration Test (Reference to PRD-1.4)

Action:	• Generate 8 cps for 24 hours.
Result:	• Calls will be allowed to complete.
Test Equipment for this test case: Use Vovida load generators	
Tested by Vovida.	

2.2.2 Memory Usage

Action:	• Generate 8cps for 24 hours. • Examine system memory usage
Result:	• Calls will be allowed to complete.
Test Equipment for this test case: Use Vovida load generators	
Tested by Vovida.	

2.3 No Single Point of Failure Test (Reference to PRD-1.4, Sec-2.2.1)

The system must support dual redundancy on all core components. (Only applied to Provisioning Server, CDR Server, Redirecting Server, SIP and Gateway Marshal Server, Stateless Feature Server). Reference Failure Flow Chart. For the test cases in this section, to bring a server down remove its ethernet connection.

2.3.1 CDR Dual-Write Redundancy (Reference to PRD-3.4.5)

The system must have the ability to dual-write CDR records to redundant (internal) CDR servers.

Action:	• Bring up two CDR servers with the V.O.C.A.L. switch • Make calls • Bring down first CDR server • Make calls • Bring up first CDR server and bring down second CDR server • Make calls • Examine call detail record
Result:	• Check that both CDR logs contain identical call records for the calls made when both CDR servers are up • When first CDR server is down, only the second CDR server will have CDR logs generated • When the second CDR server is down and the first up, only the first CDR server will have logs generated • When both CDR server are up again, both will have identiacl call records from the time that the servers are active.

Test Equipment for this test case: Redundant System. Billing (MIND) server
Tested by Vovida.

2.3.2 Redundant Redirect Server Test

Action:	• While calls are being generated bring Redirect Server down • Make a Telecaster call • Bring Redirect Server up • Make a Telecaster call
Result:	• Redundant Redirect Server will activate. Established calls will be unaffected.
Test Equipment for this test case: Redundant System.	
Tested by Vovida.	

2.3.3 Redundant SIP Marshal Server Test

Action:	• Bring SIP Marshal Server down • Make a Telecaster call • Bring SIP Marshal Server up • Make a Telecaster call
Result:	• Backup SIP Marshal Server will activate. Call processing will continue.
Test Equipment for this test case: Redundant System.	
Tested by Vovida.	

2.3.4 Redundant Gateway Marshal Server Test

Action:	• Bring Gateway Marshal Server down • Make a Telecaster call to PSTN • Bring Gateway Marshal Server up • Make a Telecaster call to PSTN
Result:	• Backup Gateway Marshal Server will activate. Call processing will continue.
Test Equipment for this test case: Redundant System.	
Tested by Vovida.	

2.3.5 Redundant Feature Server Test

Action:	• Bring FNA Feature Server down • Make a Telecaster call • Bring FNA Feature Server up • Make a Telecaster call
Result:	• Backup Feature Server will activate. Call processing will continue.
Test Equipment for this test case: Redundant System.	
Tested by Vovida.	

2.3.6 Redundant Provisioning Server Test

Action:	• Using two provisioning servers - PS1 and PS2. • Provision FNA on PS1. • Verify FNA works. • Bring down PS1 • Verify FNA still works. • Bring up PS1. • Bring down PS2. • Verify FNA works.
Result:	• FNA will work in all states. Call processing will continue.
Test Equipment for this test case: Redundant System.	
Tested by Vovida.	

2.3.6.1 Boot Time Test (Reference to PRD-1.4, Sec-2.2.2)

Action:	• Power down system. • Power up system.
Result:	• Time to call processing.
Test Equipment for this test case: Redundant System.	
Tested by Vovida and Cisco.	

2.4 Redundancy

Must be architected such that multiple Redirect servers within one softswitch may be set up to provide redundant call processing capabilities.

2.4.1 Redundant and Monitoring Test (Reference to PRD-1.4, Sec-2.3.1)

A software or hardware fault on one of the active servers must transfer call processing to one of the redundant servers and must trigger an alarm or warning on the SNMP GUI server.

Action:	• Power down system with active Redirect Server. Examine SNMP GUI for alarm or warning.
Result:	• Call processing will be transferred to redundant server and warning or alarm will be triggered on SNMP GUI.
Test Equipment for this test case: Redundant System, SNMP access.	
Tested by Vovida.	

2.4.2 Server Switchover Test

Describe Switchover Time. Use Netmgnt GUI or Log messages to monitor switchover (Reference to PRD-1.4, Sec-2.3.2 and Sec-2.3.3).

Action:	• Kill call processing servers • Examine for detection of failure • Examine that switchover is completed within 2 seconds
Result:	• The log should show the details • Netmgnt GUI should show the failure and recovery of system
Test Equipment for this test case: Redundant System	
Tested by Vovida.	

2.4.3 Marshal Failure Test (Reference to PRD-1.4, Sec-2.3.4)

Failure of any marshal must be detected within 5 seconds.

Action:	• Unplug ethernet to marshal. Check for detection of failure.
Result:	• Failure will be detected within 5 seconds and displayed on SNMP log.
Test Equipment for this test case: Redundant System.	
Tested by Vovida.	

2.5 Authentication

Demonstrate two types of authentication: Digest with Telecaster and Access List for Gateway. (Reference to PRD-1.4, Sec-2.4)

2.5.1 Digest with Telecaster

2.5.1.1 Successful Authentication

Action:	• Make sure Telecaster password is correct. • Make a call to another Telecaster.
Result:	• Call is completed successfully.
Test Equipment for this test case: Reference System. 2 Cisco Telecasters.	
Tested by Vovida.	

2.5.1.2 Unsuccessful Authentication

Action:	• Make sure Telecaster password is incorrect. • Make a call to another Telecaster.
Result:	• The call fails.
Test Equipment for this test case: Reference System. 2 Cisco Telecasters.	
Tested by Vovida.	

2.5.2 Access List for Gateway

2.5.2.1 Authorized IP Address

Action:	• Provision Marshal with the correct IP address of 5300 GW. • Make a call from PSTN to a 5300 GW.
Result:	• Call is completed successfully.
Test Equipment for this test case: Reference System. 2 Cisco Telecasters. Cisco 5300.	
Tested by Vovida.	

2.5.2.2 Unauthorized IP Address

Action:	• Provision the Marshal with the incorrect IP address of the 5300 GW. • Make a call from PSTN to a 5300 GW.
Result:	• The call fails.
Test Equipment for this test case: Reference System. 2 Cisco Telecasters. Cisco 5300.	
Tested by Vovida.	

2.6 Open Platform

System will be based on the Linux operating system and will be hardware independent.

2.6.1 Installation Test, HP Server (Reference to PRD-1.4, Sec-2.5)

Action:	• Install system on HP Netserver System running Linux operating system.
Result:	• System will install and operate.
Test Equipment for this test case: HP Netserver Lpr system.	
Tested by Vovida.	

2.6.2 Installation Test, POGO Server

Action:	• Install system on POGO System running Linux operating system.
Result:	• System will install and operate.
Test Equipment for this test case: POGO System.	
Tested by Vovida.	

2.7 Negative Test Cases

Netcat scripts will be used to generate invalid or illegal requests. Since all SIP stack applications (MS, RS, FS, SUA, Voicemail, JTAPI) use the same stack the Marshal Server can be used as a representative for testing.

2.7.1 Minimal Invite

Action:	• Send minimal INV with Call-ID, Content-Length, Content-Type, CSeq, From and To fields
Result:	• Server will pass along message.
Test Equipment for this test case: Netcat for message generation	
Tested by Vovida.	

2.7.2 Send Invite With All Fields

Action:	• Send INV with all fields
Result:	• Server will pass along message.
Test Equipment for this test case: Netcat for message generation	
Tested by Vovida.	

2.7.3 Send Message With Missing To

Action:	• Send message with missing To field.
Result:	• Server will drop message without crashing.
Test Equipment for this test case: Netcat for message generation	
Tested by Vovida.	

2.7.4 Send Message With Unreachable To

Action:	• Send message with unreachable To field.
Result:	• Server will drop message without crashing.
Test Equipment for this test case: Netcat for message generation	
Tested by Vovida.	

2.7.5 Send Message With Incorrect From

Action:	• Send message with incorrect From field.
Result:	• Server will drop message without crashing.
Test Equipment for this test case: Netcat for message generation	
Tested by Vovida.	

2.7.6 Send Message With Invite Misspelled Invite

Action:	• Send message with misspelled Invite.
Result:	• Server will drop message without crashing.
Test Equipment for this test case: Netcat for message generation	
Tested by Vovida.	

2.7.7 Send 1K UDP Payload

Action:	• Send message with 1K UDP payload.
Result:	• Server will pass along message without crashing.
Test Equipment for this test case: Netcat for message generation	
Tested by Vovida.	

2.7.8 Send Message With Varied Case

Action:	• Send message with varied case.
Result:	• Server will drop message without crashing.
Test Equipment for this test case: Netcat for message generation	
Tested by Vovida.	

2.7.9 Send Empty Message

Action:	• Send empty message.
Result:	• Server will drop message without crashing.
Test Equipment for this test case: Netcat for message generation	
Tested by Vovida.	

2.7.10 Send Message With Branch In VIA

Action:	• Send message with branch in VIA.
Result:	• Server will drop message without crashing.
Test Equipment for this test case: Netcat for message generation	
Tested by Vovida.	

2.7.11 Send ACK To A Message That Was Not Sent

Action:	• Send ACK to a message that was not sent.
Result:	• Server will drop message without crashing.
Test Equipment for this test case: Netcat for message generation	
Tested by Vovida.	

2.7.12 Send 200 Message To Message That Was Not Sent

Action:	• Send 200 message to message that was not sent.
Result:	• Server will drop message without crashing.
Test Equipment for this test case: Netcat for message generation	
Tested by Vovida.	

2.7.13 Send Cancel To Message That Was Not Sent

Action:	• Send cancel message to message that was not sent.
Result:	• Server will drop message without crashing.
Test Equipment for this test case: Netcat for message generation	
Tested by Vovida.	

3 Interface Requirements

3.1 Interfaces Support for Multiple Protocols

3.1.1 MGCP Gateway and Endpoint Support via an MGCP / NCS Translator / Marshal (Reference to PRD-3.1.4)

3.1.1.1 MGCP to SIP

Action:	• Make a call from an analog phone connected to a MGCP Cisco 2600 to a Cisco SIP Telecaster.
Result:	• Call goes go through.
Test Equipment for this test case MGCP Cisco 2600 and analog phone Cisco Telecaster	
Tested by Vovida.	

3.1.1.2 SIP to MGCP

Action:	• Make a call from a Cisco SIP Telecaster to an analog phone connected to an MGCP Cisco 2600.
Result:	• Call goes through.
Test Equipment for this test case MGCP Cisco 2600 and analog phone Cisco Telecaster	
Tested by Vovida.	

3.1.1.3 MGCP to PSTN

Action:	• Make a call from an analog phone connected to an MGCP Cisco 2600 to the PSTN via a 5300 (dial 9 to get out).
Result:	• Call goes through.
Test Equipment for this test case MGCP Cisco 2600 and analog phone Cisco 5300 PSTN phone	
Tested by Vovida.	

3.1.1.4 PSTN to MGCP

Action:	• Make a call from a PSTN phone to an analog phone connected to an MGCP Cisco 2600.
Result:	• Call goes through.
Test Equipment for this test case MGCP Cisco 2600 and analog phone Cisco 5300 PSTN phone	
Tested by Vovida.	

3.1.1.5 MGCP to MGCP

Action:	• Make a call from an analog phone connected to a MGCP Cisco 2600 to another phone hooked up to the same MGCP Cisco 2600.
Result:	• Both calls go through.
Test Equipment for this test case 1 MGCP Cisco 2600 2 analog phones	
Tested by Vovida.	

3.1.1.6 MGCP to NCS Telogy Box

Action:	• Make a call from an analog phone connected to a MGCP Cisco 2600 to an analog phone connected to an NCS Telogy Box.
Result:	• Call goes through.
Test Equipment for this test case MGCP Cisco 2600 and analog phone NCS Telogy box and analog phone	
Tested by Vovida.	

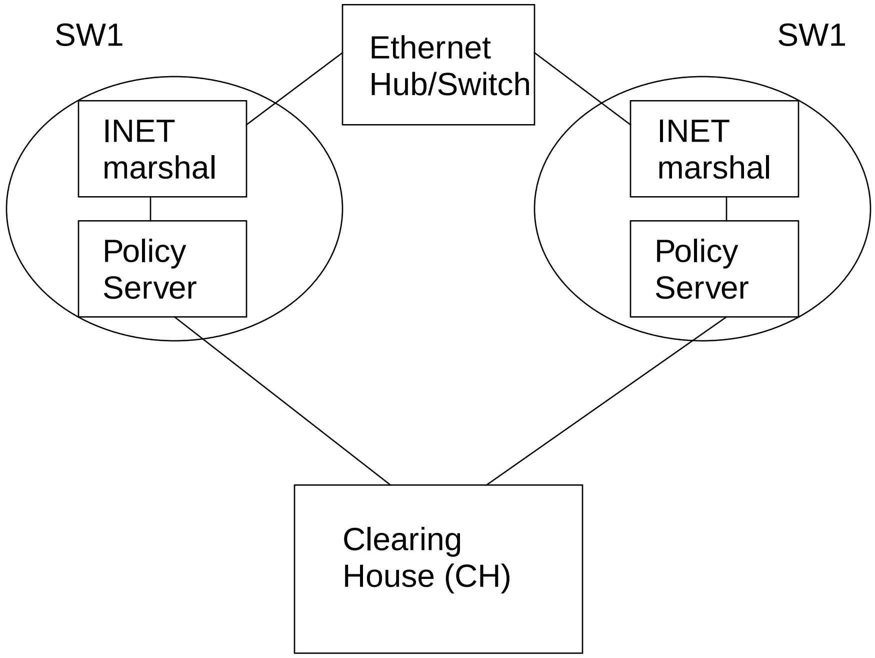
3.1.1.7 NCS Telogy Box to MGCP

Action:	• Make a call from a phone connected to an NCS Telogy Box to an analog phone connected to a MGCP Cisco 2600
Result:	• Call goes through.
Test Equipment for this test case MGCP Cisco 2600 and analog phone NCS Telogy box and analog phone	
Tested by Vovida.	

3.1.2 H.323 Gateway and Endpoint Interface (Reference to PRD-3.1.5)

Action:	• Make a call from Microsoft Netmeeting to a Cisco Telecaster • Make a call from a Cisco Telecaster to Microsoft Netmeeting • Make a call from Microsoft Netmeeting to another Microsoft Netmeeting client. • Make a call from Microsoft Netmeeting to the PSTN via a 5300 • Make a call from a PSTN phone to Netmeeting via a 5300
Result:	• Calls should complete as expected
Test Equipment for this test case 2 Microsoft Netmeeting Clients (version 3.01) Cisco Telecaster phone Cisco 5300 and corresponding PSTN analog phone.	
Tested by Vovida.	

3.1.3 System Using COPS to Communicate with Network Elements for QoS (Reference to PRD-3.1.6)



3.1.3.1 COPS with OSP

Action:	• Setup - Two VOCAL softswitches SW1 and SW2 running INET marshals. SW1 is an all-in-one system and SW2 is a Reference System. Also, ensure that PolicyServer is enrolled with the CH(Clearing House) • Make a call to user@SW2 from user@SW1 • Make a call to user@SW1 from user@SW2
Result:	• Check the output of Policy Server of SW1 to see if it did the authentication using COPS->OSP->Clearing House. • Once the call enters SW2, check POlicyServer output at SW2 to see if it did the token validation. • Token authentication/validation succeeds, call goes through. • Hang up the callee, check PolicyServer window to verify that usage is sent out to CH. • Hangup the caller, and check the PolicyServer log to verify that usage is sent out to Clearing House
Test Equipment for this test case 2 Cisco Telecaster phones All-in-One system Reference System	
Tested by Vovida.	

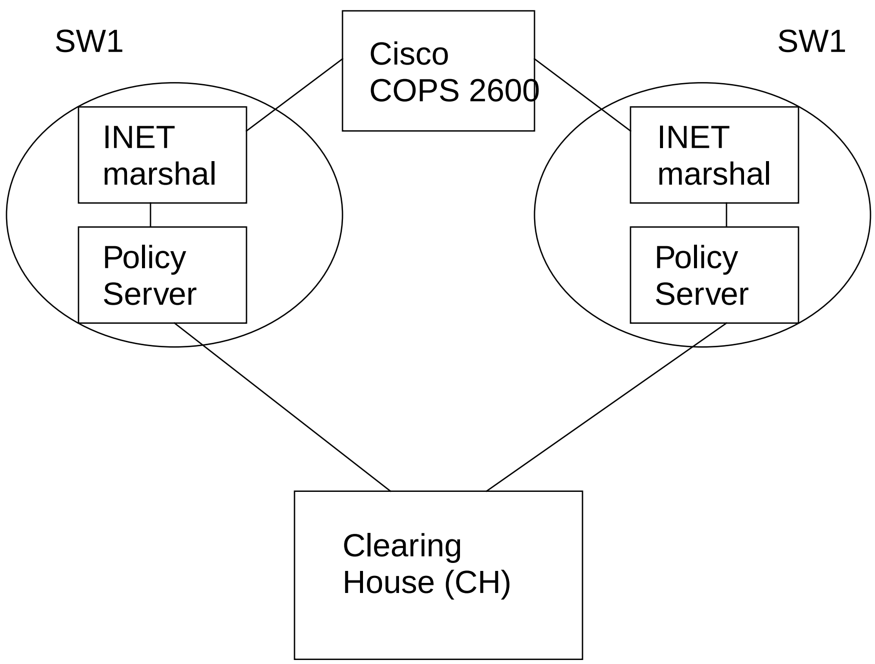
3.1.3.2 COPS with OSP -- Authentication Error

Action:	• Shutdown the PolicyServer on SW1 so only the PolicyServer on SW2 is running. • Call phone on SW2 from phone on SW1. • Restart Policy Server on SW1 (if not running), shutdown PolicyServer on SW2. • Call phone on SW1 from phone on SW2.
Result:	• Calls fail with authentication error
Test Equipment for this test case 2 Cisco Telecaster phones Reference System All-in-one system	
Tested by Vovida.	

3.1.3.3 COPS with OSP -- Validation Error

Action:	• Shutdown the PolicyServer on SW1 and restart the PolicyServer on SW2. • Call phone on SW2 from phone on SW1. • Restart Policy Server on SW1 (if not running), shutdown PolicyServer on SW2. • Call phone on SW2 from phone on SW1
Result:	• Calls fail because of validation error
Test Equipment for this test case COPS 2600 Router 2 Cisco Telecaster phones All-in-one System Reference System	
Tested by Vovida.	

3.1.3.4 COPS with RSVP



Action:	• Vovida Sip User Agent (SUA) on SW1 is talking to SUA on Athena via a Cisco COPS 2600. The test will work only with SUA's and not with Telecasters, as SUA's do the RSVP request. • Enable COPS/RSVP debugging on the COPS 2600 terminal using following command cops2600>enable cops2600#debug IP rsvp policy cops2600#debug cops • Verify that the rsvp daemon is running on both SW1 and SW2. • Make a call from SW1's SUA to SW2's SUA • Repeat test for SW2's SUA calling SW1's SUA
Result:	• Check cops2600 terminal for COPS-RSVP messages for PATH and RESV request. • Check SUA log to verify COPS request for Enable QoS. • Check PolicyServer log to see Enable QoS request. • Hang up the call. • Check PolicyServer and SUA logfiles to see Diable QoS message. • Check COPS 2600 terminal log to see RSVP Path-Tear message

Test Equipment for this test case
COPS 2600 Router
2 Vovida SIP User Agents
All-in-one System
Reference System
Tested by Vovida.

3.1.3.5 OSP (Open Settlement Protocol) Client Interface to OSP Clearing Houses (Reference to PRD-3.4.1)

Action:	• Make a call between two soft switches connected via the TransNexus clear IP demo clearing house
Result:	• Verify that inter-softswitch calls are authorized via the outgoing Inet Marshal, auth. token is carried over in the SIP INVITE as a multi-part MIME message to the incoming INet Marshal. Token is extracted and validated and call cuts through.
Test Equipment for this test case	
2 V.O.C.A.L. switches with one Cisco Telecaster connected to each switch	
TransNexus OSP Clearing House, no SSL	
Tested by Vovida.	

3.2 Programming Interfaces

3.2.1 JTAPI Application Programming Interface (Reference to PRD-3.2.1)

Action:	• Use the JAVA Dialpad application to make a call from a Vovida SIP User Agent to another Vovida SIP User Agent. • Enter the caller's phone number to log into the system • Enter the callee's phone number • Select the "Call Now" button
Result:	• The application first rings the caller's phone. • Once the caller picks up, the callee's phone starts to ring. • When the callee picks up, the caller and callee are connected in a normal phone call. • NOTE: Telecasters do not currently support Vovida's Transfer implementation so caller has to be a Vovida SIP User Agent
Test Equipment for this test case	
2 Vovida SIP User Agents	
JAVA Dialpad application created using JTAPI API	
Tested by Vovida.	

3.3 Database Interfaces (Reference to PRD-3.3)

3.3.1 Importing Data from LDAP Compliant Databases

Action:	• Create users in Vovida's LDAP database, which include data in the following fields: vocalIP, vocalFNA, vocalCFB, vocalPhonenumber • Import users from LDAP database into V.O.C.A.L. provisioning database
Result:	• Check that users have been correctly added to V.O.C.A.L. provisioning database from LDAP database by looking at the records via the V.O.C.A.L. provisioning GUI
Test Equipment for this test case V.O.C.A.L. switch software Populated LDAP database V.O.C.A.L. provisioning GUI	
Tested by Vovida.	

3.4 Billing Interface (CDR Server)

3.4.1 CDR Record Timestamp Precision of 100ms (Reference to PRD-3.4.2)

Action:	• Configure system to pass records to MIND billing system using RADIUS server • Make sure the cdrserv executable is running and view its logfiles in real time (using tail -f if necessary) • Make a call • End the call
Result:	• When call starts, watch billing executable timestamp the call in its logfile • When call ends, watch billing executable timestamp the call in its logfile • Make sure billing timestamps' numerical precision are 100ms. • Look at the MIND billing system's records to verify that the timestamps are correct.
Test Equipment for this test case V.O.C.A.L. switch 2 Cisco Telecaster phones connected to switch RADIUS server	
Tested by Vovida.	

3.4.2 CDR Record-Keeping Reliability (Reference to PRD-3.4.3)

If CDR server is not present, we can configure the marshal to disallow or allow calls.

3.4.2.1 Marshal Configured to Allow Calls

Action:	• Using the provisioning GUI, configure the marshal to allow calls if CDR server is not present • Disconnect the CDR Server from the network • Make a call
Result:	• Calls should proceed as normal
Test Equipment for this test case V.O.C.A.L. switch 2 Cisco Telecaster phones connected to switch	
Tested by Vovida.	

3.4.2.2 Marshal Configured to Disallow Calls

Action:	• Using the provisioning GUI, configure the marshal to disallow calls if CDR server is not present • Disconnect the CDR Server from the network • Make a call • Reprovision the marshal to allow calls if CDR is not present. • Make a call.
Result:	• When marshal is provisioned to disallow calls if CDR is not present, caller should receive a busy signal and call will not go through. • When marshal is provisioned to allow calls if CDR is not present, call will go through.
Test Equipment for this test case V.O.C.A.L. switch 2 Cisco Telecaster phones connected to switch	
Tested by Vovida.	

3.4.3 72-hour CDR Record queue (Reference to PRD-3.4.4)

The system must provide the capability to queue CDR records for up to 72 hours before the records must be passed to a 3rd party billing system or discarded.

Action:	• Assuming 8 calls per second, 8 x 3600 x 72 = ~2 million records are generated within a 72 hour period. • Generate 2 million CDR call records. • Send the CDR call records to the MIND billing system.
Result:	• Check that all records are passed to MIND system.
Test Equipment for this test case V.O.C.A.L. switch CDR record generator MIND billing system	
Tested by Vovida.	

3.4.4 CDR Record Disk Usage

Action:	• Generate 2 million CDR records.
Result:	• Ensure that there is sufficient disk space to accommodate 2 million records.
Test Equipment for this test case V.O.C.A.L. switch CDR record generator	
Tested by Vovida.	

3.5 Internetworking with Other Products

3.5.1 PSTN Gateways (Reference to PRD-3.6.2)

3.5.1.1 SIP Telecaster to SIP Cisco 2600 FXS (Reference to PRD-3.6.2.3)

Action:	• Make a call from a Cisco Telecaster to a Cisco 2600 FXS connected to an analog phone.
Result:	• Call should complete as expected
Test Equipment for this test case 1 Telecaster phone V.O.C.A.L. switch Cisco 2600 with FXS card and one analog phone	
Tested by Vovida.	

3.5.1.2 SIP Cisco 2600 FXS to SIP Telecaster (Reference to PRD-3.6.2.3)

Action:	• Make a call from a Cisco 2600 FXS connected an analog phone to a Cisco Telecaster.
Result:	• Call should complete as expected
Test Equipment for this test case 1 Telecaster phone V.O.C.A.L. switch Cisco 2600 with FXS card and one analog phone	
Tested by Vovida.	

3.5.1.3 SIP Telecaster to SIP Cisco 2600 FXO (Reference to PRD-3.6.2.3)

Action:	• Make a call from a Cisco Telecaster to a Cisco 2600 FXO connected to the PSTN.
Result:	• Call should complete as expected
Test Equipment for this test case 1 Telecaster phone V.O.C.A.L. switch SIP Cisco 2600 with FXO card connected to the PSTN	
Tested by Vovida.	

3.5.2 Voice Mail / Integrated Messaging (Reference to PRD-3.6.3)

3.5.2.1 Vovida Integrated Messaging System.

Action:	• Configure the callee with CFNA to voice mail. • Make sure the callee's voice mail and e-mail account are set up. • Make a call from a Cisco Telecaster to another Cisco Telecaster but let the callee ring until the Vovida voice mail system picks up. • Leave a message
Result:	• Check that the message is sent via e-mail to the phone user's account • Via the attached voice message in the e-mail received, listen to the voice message to verify the sound quality.
Test Equipment for this test case 2 Telecaster phones E-mail client and computer with speakers V.O.C.A.L. switch with configured voice mail	
Tested by Vovida.	

3.5.3 Multiport Conference Bridge (MCB) (Reference to PRD-3.6.4)

MCB is supported by Vovida SIP User Agents and MGCP endpoints only.

3.5.3.1 Meet Me Conference Call Using Cisco Telecaster Phones

Action:	• Create a three-way conference situation by having the three Cisco Telecasters call the MCB.
Result:	• Conference call completes normally.
Test Equipment for this test case 3 Cisco Telecaster phones V.O.C.A.L. switch Multi-Conference Bridge (MCB) Server	
Tested by Vovida.	

3.5.3.2 Meet Me Conference Call Using Various Phones

Action:	• Create a three-way conference situation by having one Cisco Telecaster, one Cisco MGCP 2600 hooked up to a phone, and one Telogy NCS box hooked up to a phone call into the MCB.
Result:	• Conference call completes normally.
Test Equipment for this test case 1 Cisco Telecaster phone Cisco MGCP 2600 and phone Telogy NCS and phone V.O.C.A.L. switch Multi-Conference Bridge (MCB) Server	
Tested by Vovida.	

3.5.3.3 Ad Hoc Conference Call Using Vovida SIP User Agents

Action:	• Configure one Vovida SIP User Agent to use conferencing. • From the configured SIP User Agent, call the second SIP User Agent. • Flash hook while the second SIP User Agent stays on the line, and call the third SIP User Agent. • Flash hook again to have all three user agents in a conference call.
Result:	• Conference call completes normally via the MCB.
Test Equipment for this test case 3 VOVIDA SIP User Agents and phones V.O.C.A.L switch Multi-Conference Bridge (MCB) Server	
Tested by Vovida.	

3.5.3.4 Ad Hoc Conference Call Using Vovida SIP User Agent and Various Phones

Action:	• Configure one Vovida SIP User Agent to use conferencing. • From the configured SIP User Agent, call the phone connected to the Cisco MGCP 2600. • Flash hook while the phone connected to the Cisco 2600 stays on the line, and call the phone connected to the Telogy NCS box. • Flash hook again to have all three phones in a conference call.
Result:	• Conference call completes normally via the MCB.
Test Equipment for this test case VOVIDA SIP User Agent and phone Cisco MGCP 2600 Telogy NCS and phone V.O.C.A.L switch Multi-Conference Bridge (MCB) Server	
Tested by Vovida.	

4 Routing Requirements

4.1 Route TDM based voice calls via a CAS trunk gateway (Reference to PRD-4.3)

Action:	• Use the 5300 gateway that has a CAS line connected to it • Make a phone call from a PSTN line to an on-net phone (i.e. Telecaster)
Result:	• Call goes through the 5300 • TDM based voice call is setup, and speech path is clear
Test Equipment for this test case One PSTN phone, 5300 gateway	
Tested by Vovida.	

4.2 Route 1-800, 1-877, 1-888, 1-900, and similar types of calls (Reference to PRD-4.6)

Action:	• Make a phone call dialing 1-800-ALL-NEWS (1-800-255-6397) • Make a phone call dialing 1-877-381-7287 (UReach) • Make a phone call dialing 1-888-NIH_NIDA (1-888-644-6432) • Make a phone call dialing 1-900-740-1000 (Big Brother Viewer Poll, .99/ min)
Result:	• Each call is setup, and speech path is clear
Test Equipment for this test case One Telecaster, 5300 gateway	
Tested by Vovida.	

5 Operational Requirements

5.1 Test Cases for Operation, Service and Support

5.1.1 OSS should provide a web based GUI for ease of use (Reference to PRD-5.1.1)

5.1.1.1 For Windows Machine

Action:	• Login as a User that has already been provisioned through a windows machine
Result:	• The User information should appear and look presentable. The layout of the GUI should allow users to change their information and/or features easily.
Test Equipment for this test case Windows machine on the network.	
Tested by Vovida.	

5.1.1.2 For Linux Machine

Action:	• Login as a User that has already been provisioned through a linux machine.
Result:	• The User information should appear and look presentable. The layout of the GUI should allow users to change their information and/or features easily.
Test Equipment for this test case Linux machine on the network.	
Tested by Vovida.	

5.1.2 OSS Server must provide alarms and warnings for abnormal network conditions (Reference to PRD-5.1.2)

Action:	• Bring up the SNMP GUI and monitor the status of all the servers and executables running on each of the servers. They should all be up and running and the status should show blue. Bring down one of the marshals or feature servers.
Result:	• The blue ball next to the marshal or feature server that was brought down will turn red. Red will indicate that this executable is no longer up. • Start the executable again and make sure that the status changes again.
Test Equipment for this test case Linux/Windows Machine on the network	
Tested by Vovida.	

5.1.3 The SNMP server must provide alarms and warnings for various system hardware and software faults (Reference to PRD-5.1.3)

Action:	• Use the SNMP GUI to view the status of all the servers and executables. • Bring down a linux host machine and an application (marshal or feature server) on another server.
Result:	• The SNMP GUI should now show that they are down. • Bring them back up and see the status change appropriately
Test Equipment for this test case Linux/Windows machine on the network	
Tested by Vovida.	

5.1.4 The system must have variable access levels with password protection. (Reference to PRD-5.1.5)

5.1.4.1 Logging in as Administrator.

Action:	• Log into the system as Administrator with the administrator password.
Result:	• Only screens that have admin access should be displayed.
Test Equipment for this test case Windows/Linux machine on the network.	
Tested by Vovida.	

5.1.4.2 Logging in as Technician

Action:	• Log into the system as Technician with the Technician password.
Result:	• Only screens that have technician access should be displayed.
Test Equipment for this test case Windows/Linux machine on the network.	
Tested by Vovida.	

5.1.4.3 Logging in as User

Action:	• Log into the system as a User with the user password.
Result:	• Only the screen relevant to the user should be displayed.
Test Equipment for this test case Windows/Linux machine on the network.	
Tested by Vovida.	

NOTE : Currently the system allows Administrator, Technician and User logins all with the same password - vovida.

5.1.5 The system OSS system must allow for remote access for support personnel to remotely log into the system for troubleshooting, diagnostics and problem resolution. (Reference to PRD - 5.1.6)

5.1.5.1 Logging in as Administrator

Action:	• Open a browser from a windows machine with internet access. Go to the appropriate web address and bring up the GUI in the Administrator level. Add a new user.
Result:	• The Administrator GUI should come up and one should be able to perform administrator functions. • User is added.
Test Equipment for this test case Windows machine with internet access.	
Tested by Vovida.	

5.1.5.2 Logging in as Technician

Action:	• Open a browser from a windows machine with internet access. Go to the appropriate web address and bring up the GUI in the Technician level. • create a server (Marshal or Feature server)
Result:	• The Technician GUI should come up and one should be able to perform technician functions. • Server is created.
Test Equipment for this test case Windows machine with internet access.	
Tested by Vovida.	

5.1.5.3 Logging in as User

Action:	• Open a browser from a windows machine with internet access. Go to the appropriate web address and log in as an existing user.
Result:	• The User GUI should come up and the user should be able to select or un-select features.

Test Equipment for this test case Windows machine with internet access.
Tested by Vovida.

NOTE : The system does allow access to the GUI from any computer on the internet. However, as stated above, the password for all levels is the same - vovida.

5.2 Test Cases for Provisioning

5.2.1 The provisioning server must provide an interface so that the administrator can input subscriber (end user) data (Reference to PRD-5.2.1).

5.2.1.1 Long Distance Call Blocking

Action:	• Login as a system administrator through the provisioning GUI. Go into an existing user. Block long-distance calls for that user. Try making a long distance call from that user's phone.
Result:	• The information is easily changed. • The long distance call does not go through.
Test Equipment for this test case Windows/Linux machine on the network, telecaster.	
Tested by Vovida.	

5.2.1.2 Enabling Long Distance Calling

Action:	• Login as a system administrator through the provisioning GUI. Go into the same user as above and enable long distance calls for that user. Make a long distance call from that user's phone.
Result:	• The information is easily changed. • The call goes through.
Test Equipment for this test case Windows/Linux machine on the network, telecaster.	
Tested by Vovida.	

5.2.2 The provisioning server must provide control information to the Policy Server(s) and Feature Server(s) for Implementation of the Features inputted above (Reference to PRD-5.2.2).

5.2.2.1 Disabling Call Forward No Answer.

Action:	• Login as a system administrator through the provisioning GUI. Go into an existing user. Un-check the forward no answer feature for that user. Make a phone call to that user.
Result:	• The information is easily changed. • The phone keeps ringing.
Test Equipment for this test case Windows/Linux machine on the network, telecaster.	
Tested by Vovida.	

5.2.2.2 Enabling Call Forward No Answer

Action:	• Login as a system administrator through the provisioning GUI. Check the Forward no Answer feature and forward it to the appropriate voicemail box. Make a call to that user and let it ring.
Result:	• The information is easily changed. • The call gets forwarded to voicemail after a certain number of rings.
Test Equipment for this test case Windows/Linux machine on the network, telecaster.	
Tested by Vovida.	

5.2.3 The provisioning server must provide a user interface so that the administrator can easily input and update the Dialing Plan information.(Reference to PRD-5.2.3)

Action:	• Login as a technician through the provisioning GUI. Go into Provisioning and bring up the dialing plan information. Modify the dial-plan so that all call to area code 415 go through the 2600. Make a call to a 415 number.
Result:	• The call goes through and is routed through the 2600.
Test Equipment for this test case Windows/Linux machine on the network.	
Tested by Vovida.	

6 Feature Requirements

The Feature Server will be a stand-alone, scriptable platform to provide a standard set of telephony features and the capability to offer new and innovative services. The Feature Server will be scriptable so that customers will be able to rapidly and easily develop new telephony features based on the existing standard feature set.

The following two matrixes are summary of all feature test cases in this Section. Each blank cell will be filled in with each corresponding test case result.

TABLE 1. Feature Matrix One

Section	AKA	Feature	IP Pone to IP Phone	IP Phone to Gateway	Gateway to IP Phone
6.1	CND	CALL NUMBER DELIV-ERY			
6.2	CNAM	CALL NAME DELIV-ERY			
6.3	CIDB	CLID BLOCK			--------
6.4	CH	CALL HOLD			
6.5	CFA	FWD ALL			
6.6	CFNA	FWD NA			
6.7	CFB	FWD BUSY			
6.8	CW	CALL WAITING		--------	
6.9	CR	CALL RETURN		--------	
6.10	ER	EARLY RTP			
6.11	CB	CALL BLOCK			
6.12	CS	CALL SCREEN		--------	

TABLE 2. **Feature Matrix Two**

Section	AKA	Feature	IP Phone to IP Phone to IP Phone	IP Phone to IP Phone to Gateway	Gateway to IP Phone to IP Phone
6.13	BT	BLIND TRANSFER			
6.14	CT	CONSULTA-TION TRANSFER			

6.1 Calling Number Delivery (CND) (Reference to PRD-1.4, Sec-6.1.1)

Calling Number Delivery provides to the line where the call is to be terminated. Reference to PRD-1.4, Sec-6.1

6.1.1 IP Phone to IP Phone

Action:	• Make a call between Telecasters (See TABLE 1. Feature Matrix One)
Result:	• The phone number should be displayed
Test Equipment for this test case VOCAL system, Two Telecasters	

6.1.2 IP Phone to Gateway

Action:	• Make a calls from Telecaster regular phone (See TABLE 1. Feature Matrix One)
Result:	• The phone number should be displayed
Test Equipment for this test case VOCAL system, 5300 system, One Telecaster, One regular phone	

6.1.3 Gateway to IP Phone

Action:	• Make a call from regular phone to Telecaster(See TABLE 1. Feature Matrix One)
Result:	• The phone number should be displayed
Test Equipment for this test case VOCAL system, 5300 system, One Telecaster, One regular phone	

6.2 Calling Name Delivery (CNAM) (Reference to PRD-1.4, Sec-6.1.2)

6.2.1 IP Phone to IP Phone

Action:	• Make a call between Telecasters (See TABLE 1. Feature Matrix One)
Result:	• The user name registered should be displayed
Test Equipment for this test case VOCAL system, Two Telecasters	

6.2.2 IP Phone to Gateway

Action:	• Make a call from Telecaster to regular phone (See TABLE 1. Feature Matrix One)
Result:	• The user name registered should be displayed
Test Equipment for this test case VOCAL system, 5300 system, One Telecaster, One regular phone	

6.2.3 Gateway to IP Phone

Action:	• Make a call from regular phone to Telecaster (See TABLE 1. Feature Matrix One)
Result:	• The user name registered should be displayed
Test Equipment for this test case VOCAL system, 5300 system, One Telecaster, One regular phone	

6.3 Caller Identity Blocking (CIDB) (Reference to PRD-1.4, Sec-6.1.3)

Caller ID Blocking allows a subscriber to control whether or not their number (CND) or name (CNAM) is delivered when they place an outgoing call. Reference to PRD-1.4, Sec-6.1.3

6.3.1 IP Phone to IP Phone

Action:	• Disable CND and CNAM • Make a call between Telecasters (See TABLE 1. Feature Matrix One)
Result:	• Both phone number and user name should be blocked
Test Equipment for this test case VOCAL system, Two Telecasters	

6.3.2 IP Phone to Gateway

Action:	• Disable CND and CNAM • Make a call from Telecaster to regular phone (See TABLE 1. Feature Matrix One)
Result:	• Both phone number and user name should be blocked
Test Equipment for this test case VOCAL system, 5300 system, One Telecaster, One regular phone	

6.4 Call Hold

6.4.1 IP Phone to IP Phone

Action:	• Make a call between Telecasters (See TABLE 1. Feature Matrix One) • Select HOLD function
Result:	• No speed path for the phone while it is in HOLD mode
Test Equipment for this test case VOCAL system, Two Telecasters	

6.4.2 IP Phone to Gateway

Action:	• Make a call from Telecaster to a regular phone (See TABLE 1. Feature Matrix One) • Select HOLD function
Result:	• No speed path for the phone while it is in HOLD mode
Test Equipment for this test case VOCAL system, 5300 system, One Telecaster, One regular phone	

6.4.3 Gateway to IP Phone

Action:	• Make a call from a regular phone to Telecaster (See TABLE 1. Feature Matrix One) • Select HOLD function
Result:	• No speed path for the phone while it is in HOLD mode
Test Equipment for this test case VOCAL system, 5300 system, One Telecasters, One regular phone	

6.5 Call Forward — All Calls (CFA) (Reference to PRD-1.4, Sec-6.2.1)

All Calls allows a customer to re-route all calls to an alternative number. When CFA is activated, a call to the listed number is re-routed to a user selected alternative number or a voice messaging system.

6.5.1 IP Phone to IP Phone

Action:	• Provision the phone which is supposed to forward all calls • Make a call from Telecaster to Telecaster (See TABLE 1. Feature Matrix One)
Result:	• The call should be routed to the phone as provisioned
Test Equipment for this test case VOCAL system, Three Telecasters	

6.5.2 IP Phone to Gateway

Action:	• Provision the phone which is supposed to forward all calls • Make a call from Telecasters to Telecaster (See TABLE 1. Feature Matrix One)
Result:	• The call should be routed to the phone as provisioned
Test Equipment for this test case VOCAL system, 5300 system, Two Telecasters, One regular phone	

6.5.3 Gateway to IP Phone

Action:	• Provision the phone which is supposed to forward all calls • Make a call from Telecaster to regular phone (See TABLE 1. Feature Matrix One)
Result:	• The call should be routed to the phone as provisioned
Test Equipment for this test case VOCAL system, 5300 system, Two Telecasters, One regular phone	

6.6 Call Forward — No Answer Mode (CFNA) (Reference to PRD-1.4, Sec-6.2.2)

No Answer Mode allows a customer to specify where an unanswered call should be routed. When CFNA is activated, a call to the listed number that does not answer in a specified number of ringing cycles will forward to a user selected alternative number.

6.6.1 IP Phone to IP Phone

Action:	• Provision the phone which is supposed to forward no answer calls • Make a call between Telecasters (See TABLE 1. Feature Matrix One)
Result:	• The call should be routed to the phone as provisioned
Test Equipment for this test case VOCAL system, Three Telecasters	

6.6.2 IP Phone to Gateway

Action:	• Provision the phone which is supposed to forward no answer calls • Make a call from Telecaster to Telecaster (See TABLE 1. Feature Matrix One)
Result:	• The call should be routed to the phone as provisioned
Test Equipment for this test case VOCAL system, 5300 system, Two Telecasters, One regular phone	

6.6.3 Gateway to IP Phone

Action:	• Provision the phone which is supposed to forward no answer calls • Make a call from Telecaster to regular phone (See TABLE 1. Feature Matrix One)
Result:	• The call should be routed to the phone as provisioned
Test Equipment for this test case VOCAL system, 5300 system, Two Telecasters, One regular phone	

6.7 Call Forward — Busy Mode (CFB) (Reference to PRD-1.4, Sec-6.2.3)

Busy Mode allows a customer to specify where a call should be routed when the listed number is in use. When CFB is activated, a call to the listed number while it is in use will forward to a user selected alternative number.

6.7.1 IP Phone to IP Phone

Action:	• Provision the phone which is supposed to forward busy calls • Make a call between Telecasters (See TABLE 1. Feature Matrix One)
Result:	• The call should be routed to the phone as provisioned
Test Equipment for this test case VOCAL system, Three Telecasters	

6.7.2 IP Phone to Gateway

Action:	• Provision the phone which is supposed to forward busy calls • Make a call from Telecasters to Telecaster (See TABLE 1. Feature Matrix One)
Result:	• The call should be routed to the phone as provisioned
Test Equipment for this test case VOCAL system, 5300 system, Two Telecasters, One regular phone	

6.7.3 Gateway to IP Phone

Action:	• Provision the phone which is supposed to forward busy calls • Make a call from Telecaster to regular phone (See TABLE 1. Feature Matrix One)
Result:	• The call should be routed to the phone as provisioned
Test Equipment for this test case VOCAL system, 5300 system, Two Telecasters, One regular phone	

6.8 Call Waiting (CW) (Reference to PRD-1.4, Sec-6.3)

6.8.1 IP Phone to IP Phone

Action:	• Make sure the phone with call waiting function • Make calls among Telecasters (See TABLE 1. Feature Matrix One)
Result:	• Call waiting call should be present
Test Equipment for this test case VOCAL system, three Telecasters	

6.8.2 Gateway to IP Phone

Action:	• Make sure the phone with call waiting function • Make a call from regular phone to Telecaster • Make the second call to the phone with call waiting function (See TABLE 1. Feature Matrix One)
Result:	• Call waiting call should be present
Test Equipment for this test case VOCAL system, 5300 system, Two Telecaster, One regular phone	

6.9 Call Return (Reference to PRD-1.4, Sec-6.5)

Call Return allows the subscriber to place a call back to the last number that called him or her.

6.9.1 IP Phone to IP Phone

Action:	• Make a call between Telecasters (See TABLE 1. Feature Matrix One) • Press *69 from the previous callee
Result:	• The call should be routed to the previous caller after pressing *69
Test Equipment for this test case VOCAL system, Two Telecasters	

6.9.2 Gateway to IP Phone

Call Return allows the subscriber to place a call back to the last number that called him or her.Early RTP (183)

Action:	• Make a call from regular phone to Telecaster (See TABLE 1. Feature Matrix One) • Press *69 from the previous callee
Result:	• The call should be routed to the previous caller after pressing *69
Test Equipment for this test case VOCAL system, 5300 system, One Telecaster, One regular phone	

6.10 Early RTP

6.10.1 IP Phone to IP Phone

Action:	• Make a call between Telecasters (See TABLE 1. Feature Matrix One)
Result:	• 183 should be in the message by IPGRAB
Test Equipment for this test case VOCAL system, Two Telecasters, IPGRAB	

6.10.2 IP Phone to Gateway

Action:	• Make a call from Telecaster to regular phone (See TABLE 1. Feature Matrix One)
Result:	• 183 should be in the message by IPGRAB
Test Equipment for this test case VOCAL system, 5300 system, One Telecaster, One regular phone, IPGRAB	

6.10.3 Gateway to IP Phone

Action:	• Make a call from regular phone to Telecaster (See TABLE 1. Feature Matrix One)
Result:	• 183 should be in the message by IPGRAB
Test Equipment for this test case VOCAL system, 5300 system, One Telecaster, One regular phone, IPGRAB	

6.11 Call Block

6.11.1 IP Phone to IP Phone

Action:	• Make sure the number wanted to be blocked • Make a call between Telecasters (See TABLE 1. Feature Matrix One)
Result:	• The call specified should not go through
Test Equipment for this test case VOCAL system, Two Telecasters	

6.11.2 IP Phone to Gateway

Action:	• Make sure the number wanted to be blocked • Make a call from Telecaster to regular phone (See TABLE 1. Feature Matrix One)
Result:	• The call specified should not go through
Test Equipment for this test case VOCAL system, 5300 system, One Telecaster, One regular phone	

6.11.3 Gateway to IP Phone

Action:	• Make sure the number wanted to be blocked • Make a call from regular phone to Telecaster (See TABLE 1. Feature Matrix One)
Result:	• The call specified should not go through
Test Equipment for this test case VOCAL system, 5300 system, One Telecaster, One regular phone	

6.12 Call Screen

6.12.1 IP Phone to Phone

Action:	• Make sure the number wanted to be screened • Make a call between Telecasters (See TABLE 1. Feature Matrix One)
Result:	• The call should be screened out from the receiving side
Test Equipment for this test case VOCAL system, Two Telecasters	

6.12.2 Gateway to Phone

Action:	• Make sure the number wanted to be screened • Make a call from regular phone to Telecaster (See TABLE 1. Feature Matrix One)

Result:	• The call should be screened out from the receiving side
Test Equipment for this test case VOCAL system, 5300 system, One Telecaster, One regular phone	

6.13 Blind Transfer (Reference to PRD-1.4, Sec-6.4)

Call Transfer allows a user on any existing two-party call to place on hold the existing call and originate another call to a third party. The user may consult privately or transfer the original call to the third party.

6.13.1 IP Phone to IP Phone to IP Phone

Action:	• Make a call from Telecaster1 to Telecaster2 • Telecaster1 does hookflash and hear dial tone, dial Telecaster3 and hangs up as soon as Telecaster3 starts ringing before it picks up (See TABLE 1. Feature Matrix Two)
Result:	• Telecaster1 should hear ringback tone • Telecaster2 and Telecaster3 should be connected after Telecaster3 picks up
Test Equipment for this test case VOCAL system, Three Telecasters	

6.13.2 IP Phone to IP Phone to Gateway

Action:	• Make a call from Telecaster1 to Telecaster2 • Telecaster1 does hookflash and hear dial tone, dial a regular phone and hangs up as soon as it starts ringing before it picks up (See TABLE 1. Feature Matrix Two)
Result:	• Telecaster1 should hear ringback tone • The regular phone and Telecaster3 should be connected after Telecaster3 picks up
Test Equipment for this test case VOCAL system, Two Telecasters, One regular phone	

6.13.3 Gateway to IP Phone to IP Phone

Action:	• Make a call from a regular phone to Telecaster1 • The regular phone does hookflash and hear dial tone, dial Telecaster2 and hangs up as soon as it starts ringing before it picks up (See TABLE 1. Feature Matrix Two)
Result:	• The regular phone should hear ringback tone • Telecaster1 and Telecaster2 should be connected after Telecaster3 picks up
Test Equipment for this test case VOCAL system, Two Telecasters, One regular phone	

6.14 Consultation Transfer (Reference to PRD-1.4, Sec-6.4)

6.14.1 IP Phone to IP Phone to IP PHone

Action:	• Make a call from Telecaster1 to Telecaster2 • Telecaster1 does hookflash and hear dial tone, dial Telecaster3 and talks with it and hangs up (See TABLE 1. Feature Matrix Two)
Result:	• Telecaster1 should hear ringback tone • Telecaster2 and Telecaster3 should be connected
Test Equipment for this test case VOCAL system, Three Telecasters	

6.14.2 IP Phone to IP Phone to Gateway

Action:	• Make a call from Telecaster1 to Telecaster2 • Telecaster1 does hookflash and hear dial tone, dial a regular phone and talks with it and hangs up (See TABLE 1. Feature Matrix Two)
Result:	• Telecaster1 should hear ringback tone • Telecaster2 and the regular phone should be connected
Test Equipment for this test case VOCAL system, Two Telecasters, One regular phone	

6.14.3 Gateway to IP Phone to IP PHone

Action:	• Make a call from a regular phone to Telecaster1 • The regular phone does hookflash and hear dial tone, dial Telecaster2 and talks with it and hangs up (See TABLE 1. Feature Matrix Two)
Result:	• The regular phone should hear ringback tone • Telecaster1 and Telecaster2 should be connected
Test Equipment for this test case VOCAL system, Two Telecasters, One regular phone	

6.15 Blind Transfer for UAs

6.15.1 UA to UA to UA

Action:	• Make a call from UA1 to UA2 • UA1 does a hookflash and should hear dial tone, dials UA3 and hangs up as soon as UA3 starts ringing and before UA3 answers
Result:	• UA2 should hear ringback tone • UA2 and UA3 should be connected after UA3 picks up
Test Equipment for this test case VOCAL system, Three UAs	

6.15.2 UA to UA to Gateway

Action:	• Make a call from UA1 to UA2 • UA1 does a hookflash and should hear dial tone, dials a regular phone and hangs up as soon as this phone starts ringing and before it answers
Result:	• UA2 should hear ringback tone • UA2 and the regular phone should be connected after it picks up
Test Equipment for this test case	
VOCAL system, 5300 system, Two UAs, One regular phone	

6.15.3 Gateway to UA to UA

Action:	• Make a call from a regular phone to UA1 • The regular phone does a hookflash and should hear dial tone, dial UA2 and hangs up as soon as UA2 starts ringing and before UA2 answers
Result:	• UA1 should hear ringback tone • UA1 and UA2 should be connected after UA2 picks up
Test Equipment for this test case	
VOCAL system, 5300 system, Two UAs, One regular phone	

6.16 Consultation Transfer for UAs

6.16.1 UA to UA to UA

Action:	• Make a call from UA1 to UA2 • UA1 does a hookflash and should hear dial tone, calls UA3 and talks to UA3 and hangs up
Result:	• UA2 and UA3 should be connected
Test Equipment for this test case	
VOCAL system, Three UAs	

6.16.2 UA to UA to Gateway

Action:	• Make a call from UA1 for UA2 • UA1 does a hookflash and should hear dialtone, calls a regular phone and talks to it and hang up
Result:	• UA2 and the regular phone should be connected
Test Equipment for this test case	
VOCAL system, 5300 system, Two UAs, One regular phone	

6.16.3 Gateway to UA to UA

Action:	• Make a call from a regular phone to UA1 • The regular phone does a hookflash and should hear dialtone, calls UA2 and talks to UA2 and hangs up
Result:	• UA1 and UA2 should be connected
Test Equipment for this test case VOCAL system, 5300 system, Two UAs, One regular phone	

6.17 Call Waiting for UAs (CW)

Action:	• Make sure the phone with call waiting function • Make calls among UAs
Result:	• Call waiting call should be present
Test Equipment for this test case VOCAL system, three UAs	

6.18 Cancel Call Waiting for UAs (CW)

Action:	• Make sure the phone without call waiting function • Make calls among UAs
Result:	• Call waiting call can not be present
Test Equipment for this test case VOCAL system, three UAs	

6.19 Calling Number Delivery for MGCP endpoints (CND)

Action:	• Make a call between two MGCP endpoints
Result:	• The phone number should be displayed
Test Equipment for this test case VOCAL system, 2600 system, two phones	

6.20 Calling Name Delivery for MGCP endpoints (CNAM)

Action:	• Make a call between two MGCP endpoints
Result:	• The user name registered should be displayed
Test Equipment for this test case VOCAL system, 2600 system, two phones	

6.21 Caller Identity Blocking for MGCP endpoints (CIDB)

Action:	• Disable CND and CNAM • Make a call between two MGCP endpoints
Result:	• Both phone number and user name should be blocked
Test Equipment for this test case VOCAL system, 2600 system, two phones	

6.22 Call Hold for MGCP endpoints (CH)

Action:	• Make a call between two MGCP endpoints
Result:	• No speed path for the phone while it is in HOLD mode
Test Equipment for this test case VOCAL system, 2600 system, two phones	

6.23 Call Forward — All Calls for MGCP endpoints (CFA)

Action:	• Provision the phone which is supposed to forward all calls • Make a call from a MGCP endpoint to another MGCP endpoint
Result:	• The call should be routed to the phone as provisioned
Test Equipment for this test case VOCAL system, 2600 system, three phones	

6.24 Call Forward — No Answer Mode for MGCP endpoints (CFNA)

Action:	• Provision the phone which is supposed to forward no answer calls • Make a call from a MGCP endpoint to another MGCP endpoint
Result:	• The call should be routed to the phone as provisioned
Test Equipment for this test case VOCAL system, 2600 system, three phones	

6.25 Call Forward — Busy Mode for MGCP endpoints (CFB)

Action:	• Provision the phone which is supposed to forward busy calls • Make a call from a MGCP endpoint to another MGCP endpoint
Result:	• The call should be routed to the phone as provisioned
Test Equipment for this test case VOCAL system, 2600 system, three phones	

6.26 Call Waiting for MGCP endpoints (CW)

Action:	• Make sure the phone with call waiting function • Make calls among MGCP endpoints
Result:	• Call waiting call should be present
Test Equipment for this test case VOCAL system, 2600 system, three phones	

6.27 Cancel Call Waiting for MGCP endpoints (CCW)

Action:	• Make sure the phone without call waiting function • Make calls among MGCP endpoints
Result:	• Call waiting call can not be present
Test Equipment for this test case VOCAL system, 2600 system, three phones	

6.28 Call Return for MGCP endpoints (CR)

Action:	• Make a call between MGCP endpoints • Press *69 from the previous callee
Result:	• The call should be routed to the previous caller after pressing *69
Test Equipment for this test case VOCAL system, 2600 system, two phones	

6.29 Early RTP for MGCP endpoints (ER)

Action:	• Make a call between MGCP endpoints
Result:	• 183 should be in the message by IPGRAB
Test Equipment for this test case VOCAL system, 2600 system, two phones, IPGRAB	

6.30 Call Block for MGCP endpoints (CB)

Action:	• Make sure the number wanted to be blocked • Make a call between MGCP endpoints
Result:	• The call specified should not go through
Test Equipment for this test case VOCAL system, 2600 system, two phones	

6.31 Call Screen for MGCP endpoints (CS)

Action:	• Make sure the number wanted to be screened • Make a call between MGCP endpoints
Result:	• The call should be screened out from the receiving side
Test Equipment for this test case VOCAL system, 2600 system, two phones	

6.32 Blind Transfer for MGCP endpoints (BT)

Action:	• Make a call from MGCP endpoint1 to MGCP endpointr2 • MGCP endpoint1 does hookflash and hear dial tone, dial MGCP endpoint3 and hangs up as soon as it starts ringing before it picks up
Result:	• MGCP endpoint1 should hear ringback tone • MGCP endpoint2 and MGCP endpoint3 should be connected after it picks up
Test Equipment for this test case VOCAL system, 2600 system, three phones	

6.33 Consultation Transfer for MGCP endpoints (CT)

Action:	• Make a call from MGCP endpoint1 to MGCP endpoint2 • MGCP endpoint1 does hookflash and hear dial tone, dial MGCP endpoint3 and talks with it and hangs up
Result:	• MGCP endpoint1 should hear ringback tone • MGCP endpoint2 and MGCP endpoint3 should be connected
Test Equipment for this test case VOCAL system, 2600 system, three phones	

7 Notes

1. Define CPS - 100,000 endpoints *.10 (10%) / 180 sec/call/endpoint (3 min avg. call) = 56 calls per second
2. Define capacity tested to failure (PRD 2.1.2)
3. Define reference H/W architecture
4. Test Bed Diagram
5. Provisioning Server redundancy configuration-Nredundant or primary and backup
6. SIP authentication and how to bypass
7. “five 9s” and “four 9s” proof document
8. Define gateway, protocol translator or marshal server failure detection
9. Partial failures or grey out?? Suppose a process consumes available memory or processing power
10. Spec all equipment and software versions
    - Telogy
    - 2600 MGCP
    - 5300 SIP
    - 2600 SIP
    - Voyant MCU
    - HP Servers
    - Reliability POGO Server
    - Telecaster