Verizon Technology Organization

Technical Memorandum

SIT.GNIE.TM.VOIP.2006.003

Technology Test Suite for CarrierIP Termination Interoperability

Version 1.2

September 28th, 2006

Revision History

Version	Contact	Action	
Version 1.0 05/30/2006	Kyle Fahrenthold	New	Draft document
Version 1.1 09/26/2006	Wayne Garrett	Change	Modified TC7; Edited for Public web access
Version 1.2 09/28/06	Wayne Garrett	Change	Edited Footer
Actions Taken are: **New** = new document, **Add/Delete/Change** = a section or topic has been added, or deleted, or changed.			

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1 Executive Overview

Verizon would like to rapidly accelerate the turn up of Wholesale Carrier IP customers into its network. As a result, Verizon is providing a test facility to enable potential customers a means to validate interoperability with their VoIP systems.

This Technical Memorandum provides a set of test cases that will be used by the customer community to ensure that their products will conform to Verizon's requirements for SIP Carrier IP Termination interoperability. These test cases are based on Verizon's SIP deployment strategy and Internet Engineering Task Force's (IETF) Request for Comments (RFC); mainly Session Initiation Protocol (RFC 3261), Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity with Trusted Networks (RFC 3325), A Privacy Mechanism for the Session Initiation Protocol (SIP) (RFC 3323) and RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals (RFC 2833).

2 Technical Scope

Based on Verizon's VoIP deployment strategy, the Voice Systems Testing (VST) team has developed a Technology Test Suite consisting of SIP interoperability test cases to ensure customer's SIP equipment compatibility. The requirements are for customers to test their Wholesale SIP products against a standard test suite so that their SIP products can be correctly integrated into the Verizon network with limited interoperability issues. This Technology Test Suite focuses on VoIP-SIP interfaces for test cases that require interaction with the PSTN. Test cases do not show complete exchange of messages. Only messages relevant to the test case are shown.

3 Technical Description

This section presents the typical SIP test configurations along with the associated basic call flow. The term MGC will be used in a general manner to represent both the media gateway controller (s) and the media gateway (s) which are also referred to as "soft switches" or "call agents." Figure 1 illustrates a high level network architecture drawing of a basic Carrier IP configuration.

Click here to view a comprehensive list of customer Proxy/IPSec hardware combinations that have successfully completed Verizon Interoperability testing.

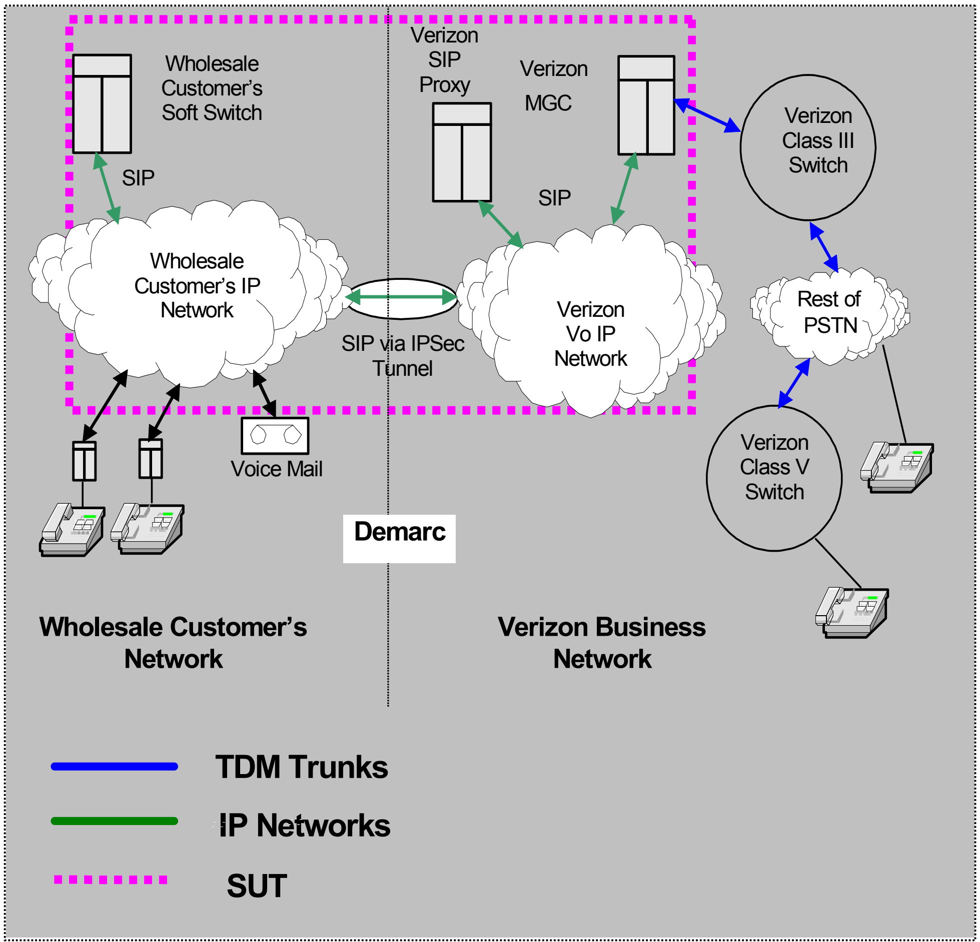

Figure 1: Verizon Wholesale **Carrier IP Network Architecture**

3.1 IP Origination - PSTN Termination

A basic Carrier IP call scenario for IP origination to PSTN destination message flow is shown below.

```
SIP UA        Customer       Verizon       Verizon        Class 3
              Proxy/MGC      Proxy           MGC          Switch
    |---INVITE--->|             |             |             |
    |<-100 Trying-|             |             |             |
    |             |---INVITE--->|             |             |
    |             |<-100 TRYING-|             |             |
    |             |             |             |             |
    |             |             |---INVITE--->|             |
    |             |             |             |---- IAM---->|
    |             |             |<-100 TRYING-|             |
    |             |             |             |<----ACM-----|
    |             |             |<----18X-----|             |
    |             |<-----18x----|             |             |
    |<----18x-----|             |             |             |
    |             |             |             |<----ANM-----|
    |             |             |<---200 OK---|             |
    |             |<---200 OK---|             |             |
    |<---200 OK---|             |             |             |
    |             |             |             |             |
    |-----ACK---->|             |             |             |
    |             |-----ACK---->|             |             |
    |             |             |-----ACK---->|             |
    |<=================Conversation========================>|
    |-----BYE---->|             |             |             |
    |             |-----BYE---->|             |             |
    |             |             |-----BYE---->|             |
    |             |             |             |---REL 16--->|
    |             |             |             |<----RLC-----|
    |             |             |<--200 OK----|             |
    |             |<--200 OK----|             |             |
    |<--200 OK----|             |             |             |
```

Test Configuration Requirements:

1. A minimum of one Verizon MGC and one Verizon SIP proxy connected via an IP network.
2. A minimum of one customer SIP Signaling endpoint. For example, this device could be a SIP Proxy, SBC, or Softswitch to name a few.
3. A minimum of one IPSec peering point to establish a secure VPN tunnel.
4. A minimum of one POTS phone and one SIP phone set both with Call ID.
5. A SIP protocol analyzer (e.g. Ethereal).

4 Test Instructions

4.1 Test Objectives

Verizon Wholesale IP Communications Services interoperability testing verifies that a potential wholesale customer's Voice over IP (VoIP) platform properly interoperates with the Verizon VoIP Network from the perspective of Session Initiation Protocol (SIP) and Real-Time Transport Protocol (RTP) and is approved for use with the production Verizon Network.

4.2 Interop Lab Entrance Criteria

- The Verizon Customer Rep will have already worked with the Customer to obtain completed legal documentation covering a non-disclosure agreement and bailment/beta test agreement.
- The Verizon Customer Rep will have worked with the Customer to technically prequalify their capabilities and confirm that the Customer is ready for testing.
- The Verizon Customer Rep will initiate and attend a Kickoff meeting between the Customer and Verizon Interop Engineering.
    - The Verizon Customer Rep will provide Verizon Interop Engineering with the Customer's service requirements.
    - The Verizon Customer Rep will provide Verizon Interop Engineering with a completed Customer Connectivity Questionnaire, which is contained in Appendix A.
    - Verizon Interop Engineering will provide the Customer with connectivity information to Verizon's Interop Lab as indicated in Appendix B.
    - Verizon Interop Engineering will provide the Customer with this Interop Test Plan.
    - Verizon Interop Engineering will indicate which category(s) of test cases are applicable for this specific Customer's service.
- Following the kickoff meeting, Verizon Interop Engineering will work with the Customer to establish IP connectivity between the Customer's equipment and Interop lab equipment.
- Verizon Interop Engineering will provide the Verizon Customer Rep and Customer with notification of entrance criteria completion, connectivity, and readiness to begin Interop testing.

4.3 Testing Requirements

- The Customer will need the ability to provision their equipment and environment.
- Communication with the Interop lab will primarily be through the exchange of e-mail.
- The Verizon Customer Rep should be copied on all e-mail exchanged.

- A “hot line” phone to the Verizon Interop Engineering will be used when more interactive dialog is needed.
- The Customer will need the ability to capture SIP traces for each test case executed and send them to Verizon Interop Engineering.
- The Customer should also be able log RTP traffic and provide traces if needed for troubleshooting

4.4 Test Suite Execution

- The customer will run each applicable test case as defined the test sections of this document.
  - Most test cases can be run by the customer independently.
- As test cases are executed, test results along with test numbers and associated SIP trace should be e-mailed to the Interop lab where the results can be reviewed by Verizon Interop Engineering for final pass/fail determination.
  - It is recommended that these not be sent in bulk, but in small groups as they are completed.
- The Verizon Interop Engineering group will respond to this e-mail within one business day.
  - If many test cases are included then a complete review and response by Verizon Interop Engineering may require additional time.
- A reply will be sent to the customer by Verizon Interop Engineering to indicate whether the test case meets the requirements (pass/fail).
  - If failed, the reasons for the failure will be indicated.

4.5 Support during Test Case Execution

- For any issue that occurs during test case execution, pertinent information should be sent in an e-mail along with SIP logs and the following additional information:
  - Test case number and issue with the test case
  - Call-ID for the call
  - “From” phone number
  - “To” phone number
  - Date and Time of test case call
- Verizon Interop Engineering will respond to this e-mail within one business day and provide assistance in resolving the issue.
- It is mandatory that test case issues and SIP logs be sent to Verizon Interop Engineering for review as they occur.

4.6 Test Suite Completion

- If there are no issues, then test cases should take less than one week to complete, depending on number of applicable test cases, including review.

- When all test cases have been executed and passed from the Customer perspective, an e-mail will be sent indicating completion.
- Any issues still believed to be open by Verizon Interop Engineering will be addressed with the Customer and Customer Rep in an attempt to resolve or clarify those issues.
- Testing and/or troubleshooting will not extend beyond a two week period if issues persist.
- When all, if any, open issues have been resolved then the Customer will have completed the interoperability testing and the Customer Rep will schedule a Test Completion Review.
- Handoff documents and test results will be provided by Verizon Interop Engineering to the Customer and the Verizon Customer Rep at the Test Completion Review.
- The Customer Rep will work with the Customer and Operations to complete Operations IP connectivity and handoff.

4.7 Restrictions and Limitations

- The Interop test plan does not mandate bulk call traffic during Interop testing.
- Supported SIP methods: INVITE, ACK, BYE CANCEL, OPTIONS
- Unsupported SIP methods: REFER, SUBSCRIBE, NOTIFY, UPDATE, INFO
- Carrier IP Wholesalers must send a valid ANI in the P-Asserted-Identity (RFC 3325) and optionally a Privacy Header (RFC 3323) to Verizon otherwise the call will be rejected.
- Delayed SDP, where the SDP is sent in the ACK, is not supported.
- SIP-MIME, SIPS/TLS, and other application level authentication and encryption techniques are not supported.
- Verizon obeys RFC3261 and ignores any headers it does not understand. Verizon may send proprietary headers to the wholesaler, which must be ignored.
- Verizon may send RTCP as part of the audio session, however, RTCP is not fully deployed and Verizon does not provide any control based on the RTCP data.
- Group 3 FAX sent over G.711 is "best effort". T.38 FAX is not supported.
- RFC3389 RTP comfort noise is not supported.
- Mid-session codec changes, without a re-INVITE, are not supported.
- The SIP INFO, KPML, SUBSCRIBE and NOTIFY methods for DTMF transport are not supported.
- SRTP and other methods of media encryption are not supported.
- Verizon authenticates and only supports SIP signaling to and from the wholesaler via IPSEC tunnels.
- Centrex functions are not supported.
- Compact SIP Headers must be supported.
- The P-Asserted-Identity header from Wholesale Carrier IP customers should be in E.164 format.

- Only valid e.164 numbers should be prefixed with a "+". All other number formats should be sent in raw digit form.
- Carrier IP services does NOT support inbound calling for the carrier or the carrier's end-users.
- Carrier IP services does NOT support Emergency Services, 101xxxx casual dialing, Operator Services, or toll-free originations.

5 Security and Authentication Testcases

TC1: Customer IPSec Peerpoint to Verizon IPSec Peerpoint

Test Steps

1. Configure Phase 1 with pre-g2-3des-md5 and pre-shared key.
2. Configure Phase 2 with g2-esp-3des-md5.
3. Configure access list policies accordingly.

Expected Results

1. Verify IPSec handshake for phase 1 proposal
2. Verify IPSec handshake for phase 2 proposal.
3. Verify interesting traffic can be exchanged through the VPN tunnel.
4. Verify the following handshake exchange.

```
Phase 1 Proposal - Main Mode

Verizon                  Customer
Peer Point              Peer Point
 |                           |
 |-----------HDR, SA-------->|        SA: Security Association
 |                           |        KE: Diffie-Hellman exchanged public value
 |<----------HDR, SA---------|        Ni, Nr: The Nonce
 |                           |        ID_I, ID_R: the Initiator, Responder
 |<--------HDR, KE Ni--------|        CERT: Certificate
 |                           |        SIG_I, SIG_R: the signature of the
 |---------HDR, KE Ni------->|                      Initiator, Responder
 |                           |        [x]: x is optional
 |<-HDR*, ID_I,[CERT},SIG_I--|        *: encryption must begin after the header
 |                           |        HDR: ISAKMP header
 |--HDR*, ID_R,[CERT],SIG_R->|
 |                           |

Phase 2 Proposals - Quick Mode

Verizon                  Customer
Peer Point              Peer Point
 |                           |
```

```
|  HDR*, HASH1, SA, Ni SA    |
|-----[KE],[ID_I,[ID_R]---->|
|                           |
|  HDR*, HASH2, SA Nr, [KE] |
|<------[ID_I], [ID_R]------|
|                           |
|<--------HDR*, HASH3------>|
|                           |
```

TC2: Calling Party Number Authentication

Test Steps

1. Originate a voice call via customer MGC to Verizon MGC.
2. Answer the call. Calling party disconnects the call.

Expected Results

1. Verify that the calling party hears audible ring back.
2. Verify that the called party hears power ringing (alerted).
3. Verify that a two way speech path is established.
4. Verify Caller ID on the terminating line.
5. Verify that the SIP INVITE contains P-Asserted-Identity with correct calling party information as shown in the example INVITE below.

INVITE sip:+19725553265@verizon.com;user=phone SIP/2.0

Via:SIP/2.0/UDP 10.10.10.10;branch=71V5060-0-912782047

From:"Customer"<sip:+19727282400@10.10.10.10;user=phone>;tag=321064913

To:<sip:9725553265@verizon.com;user=phone>

Call-ID:133032125250506744915018

CSeq:912782047 INVITE

Contact:<sip:10.10.10.10:5060>

P-Asserted-Identity:"Carrier IP Custname"<sip:+19727282400@10.10.10.10;user=phone>

Privacy:none

Allow:ACK,BYE,CANCEL,INFO,INVITE

Supported:

Accept:application/sdp,application/dtmf

Max-Forwards:10

Content-Type:application/sdp

Content-Length:292

6 IP to PSTN Testcases

6.1 Call Setup Test Cases

TC3: MGC to MGC basic North American Long Distance call

Test Steps

1. Originate a long distance voice call via customer MGC to Verizon MGC.
2. Answer the call. Calling party disconnects the call.

Expected Results

1. Verify that the calling party hears audible ring back.
2. Verify that the called party hears power ringing (alerted).
3. Verify that a two way speech path is established.
4. Verify Caller ID on the terminating line.
5. Verify that the SIP INVITE contains P-Asserted-Identity with correct calling party information.
6. Verify the Userinfo portion of the Request URI is in the following format:+1NPANXXXXXX
7. Verify the following call flow.

```
Customer    Verizon       Verizon        Class 3
 MGC         Proxy          MGC          Switch
  |             |             |             |
  |---INVITE--->|             |             |
  |             |---INVITE--->|             |
  |             |             |---- IAM---->|
  |             |<-100 TRYING-|             |
  |<-100 TRYING-|             |             |
  |             |             |             |
  |             |             |<----ACM-----|
  |             |<----18X-----|             |
  |<-----18x----|             |             |
  |             |             |             |
  |             |             |<----ANM-----|
  |             |<---200 OK---|             |
  |<---200 OK---|             |             |
  |             |             |             |
  |             |             |             |
  |             |             |             |
  |-----ACK---->|             |             |
  |             |-----ACK---->|             |
  |<==========Conversation=================>|
  |             |             |             |
  |-----BYE---->|             |             |
  |             |-----BYE---->|             |
  |             |             |---REL 16--->|
  |             |             |<----RLC-----|
```

```
 |             |<--200 OK----|             |
 |<--200 OK----|             |             |
 |             |             |             |
```

TC4: MGC to MGC basic North American to International Destination call

Test Steps

1. Originate an International voice call via customer MGC to Verizon MGC.
2. Answer the call. Calling party disconnects the call.

Expected Results

1. Verify that the calling party hears audible ring back.
2. Verify that the called party hears power ringing (alerted).
3. Verify that a two way speech path is established.
4. Verify that the SIP INVITE contains calling party information in the P-Asserted-Identity SIP Header.
5. Verify the Userinfo of the Request URI is in the following format: +CCXXXXXXXX
6. Verify the following call flow.

```
Customer    Verizon       Verizon        Class 3
 MGC         Proxy          MGC          Switch
 |             |             |             |
 |---INVITE--->|             |             |
 |             |---INVITE--->|             |
 |             |             |---- IAM---->|
 |             |<-100 TRYING-|             |
 |<-100 TRYING-|             |             |
 |             |             |             |
 |             |             |<----ACM-----|
 |             |<----18X-----|             |
 |<-----18x----|             |             |
 |             |             |             |
 |             |             |<----ANM-----|
 |             |<---200 OK---|             |
 |<---200 OK---|             |             |
 |             |             |             |
 |             |             |             |
 |             |             |             |
 |-----ACK---->|             |             |
 |             |-----ACK---->|             |
 |<==========Conversation=================>|
 |             |             |             |
 |-----BYE---->|             |             |
 |             |-----BYE---->|             |
 |             |             |---REL 16--->|
 |             |             |<----RLC-----|
 |             |<--200 OK----|             |
 |<--200 OK----|             |             |
 |             |             |             |
```

TC5: MGC to MGC NPA-555-1212 Directory Assistance

Test Steps

1. Originate a long distance voice call via customer MGC to Verizon MGC dialing NPA-555-1212.
2. Directory Assistance should be reached. Calling party disconnects the call.

Expected Results

1. Verify that the calling party hears audible ring back.
2. Verify that a two way speech path is established.
3. Verify the Directory Assistance Operator is reached.
4. Verify that the SIP INVITE contains P-Asserted-Identity with correct calling party information.
5. Verify the Userinfo portion of the Request URI is in the following format:+1NPA5551212
6. Verify the following call flow.

```
Customer    Verizon       Verizon        Class 3
 MGC         Proxy          MGC          Switch
  |             |             |             |
  |---INVITE--->|             |             |
  |             |---INVITE--->|             |
  |             |             |---- IAM---->|
  |             |<-100 TRYING-|             |
  |<-100 TRYING-|             |             |
  |             |             |             |
  |             |             |<----ACM-----|
  |             |<----18X-----|             |
  |<-----18x----|             |             |
  |             |             |             |
  |             |             |<----ANM-----|
  |             |<---200 OK---|             |
  |<---200 OK---|             |             |
  |             |             |             |
  |             |             |             |
  |             |             |             |
  |-----ACK---->|             |             |
  |             |-----ACK---->|             |
  |<==========Conversation================>|
  |             |             |             |
  |-----BYE---->|             |             |
  |             |-----BYE---->|             |
  |             |             |---REL 16--->|
  |             |             |<----RLC-----|
  |             |<--200 OK----|             |
  |<--200 OK----|             |             |
  |             |             |             |
```

TC6: MGC to MGC with Fast Answer

Test Steps

1. Originate a long distance voice call via Customer MGC to Verizon MGC.
2. Answer the call. Called party disconnects the call.

Expected Results

1. Verify that the calling party does not receive audible ring back.
2. Verify the call is immediately answered.
3. Verify that a two way speech path is established.
4. Verify Caller ID on the terminating line.
5. Verify that the SIP INVITE contains P-Asserted-Identity with correct calling party information.
6. Verify the Userinfo in the Request URI is in the following format: +1NPANXXXXXX
7. Verify the following call flow.

```
Customer    Verizon       Verizon        Class 3
 MGC         Proxy          MGC          Switch
  |              |             |              |
  |---INVITE--->|             |              |
  |              |---INVITE--->|              |
  |              |             |---- IAM---->|
  |              |<-100 TRYING-|              |
  |<-100 TRYING-|             |              |
  |              |             |              |
  |              |             |<----ANM-----|
  |              |<---200 OK---|              |
  |<---200 OK---|             |              |
  |              |             |              |
  |-----ACK---->|             |              |
  |              |-----ACK---->|              |
  |<==========Conversation=================>|
  |              |             |              |
  |              |             |<--REL 16----|
  |              |<---BYE----- |              |
  |<----BYE-----|             |              |
  |              |             |----RLC----->|
  |---200 OK--->|             |              |
  |              |---200 OK--->|              |
  |              |             |              |
```

- Optional

TC7: Call Origination with 180 Ringing –Alternate Gateway Test

NOTE: Please contact the Verizon Interop lab for assistance with this test.

Test Steps

1. Contact the Interop lab for a terminating test number.
2. Originate a voice call via Customer MGC to Verizon MGC.
3. Verify originator receives near-end ring back.

~~3.~~4. Verify Call is answered. When prompted, leave message and then retrieve message to verify voice path. Calling party disconnects the call.

Expected Results

1. Verify that the calling party hears audible ring back.

~~3.~~2. Verify the 180 Ringing is received without SDP.

3. Verify that a two way speech path is established.
4. Verify the following call flow.

```
  Customer     Verizon       Verizon        Class 3
 MGC          Proxy          MGC          Switch
 |             |             |             |
 |---INVITE--->|             |             |
 |             |---INVITE--->|             |
 |             |             |---- IAM---->|
 |             |<-100 TRYING-|             |
 |<-100 TRYING-|             |             |
 |             |             |             |
 |             |             |<----ACM-----|
 |             |<-180 Ringing|             |
 |<-180 Ringing|             |             |
 |             |             |             |
 |             |             |<----ANM-----|
 |             |<---200 OK---|             |
 |<---200 OK---|             |             |
 |             |             |             |
 |             |             |             |
 |             |             |             |
 |-----ACK---->|             |             |
 |             |-----ACK---->|             |
 |<==========Conversation=================>|
 |             |             |             |
 |-----BYE---->|             |             |
 |             |-----BYE---->|             |
 |             |             |---REL 16--->|
 |             |             |<----RLC-----|
 |             |<--200 OK----|             |
 |<--200 OK----|             |             |
 |             |             |             |
```

TC8: Call Origination with 183 Session Progress

Test Steps

1. Determine a destination that requires an early media connection, for example, airline reservation.
2. Originate a voice call via Customer MGC to Verizon MGC.
3. Answer the call. Calling party disconnects the call.

Expected Results

1. Verify that the calling party hears audible ring back.
2. Verify that the called party hears power ringing (alerted). Depending on the destination type, ring tone may not be experienced.
3. Verify that a two way speech path is established.
4. Verify Caller ID on the terminating line.
5. Verify the 183 Session Progress is received with correct SDP configuration.
6. Verify the following call flow.

```
Customer              Verizon                 Verizon       Class 3
 MGC                   Proxy                    MGC         Switch
 |                       |                       |             |
 |--------INVITE-------->|                       |             |
 |                       |--------INVITE-------->|             |
 |                       |                       |---- IAM---->|
 |                       |<------100 TRYING------|             |
 |<------100 TRYING------|                       |             |
 |                       |                       |             |
 |                       |                       |<----ACM-----|
 |                       |<-183 Session Progress-|             |
 |<-183 Session Progress-|                       |             |
 |                       |                       |             |
 |                       |                       |<----ANM-----|
 |                       |<--------200 OK--------|             |
 |<-------200 OK---------|                       |             |
 |                       |                       |             |
 |                       |                       |             |
 |                       |                       |             |
 |----------ACK--------->|                       |             |
 |                       |----------ACK--------->|             |
 |<=====================Conversation==========================>|
 |                       |                       |             |
 |----------BYE--------->|                       |             |
 |                       |----------BYE--------->|             |
 |                       |                       |---REL 16--->|
 |                       |                       |<----RLC-----|
 |                       |<-------200 OK---------|             |
 |<------200 OK----------|                       |             |
 |                       |                       |             |
```

TC9: Call setup with "early media"

Test Steps

1. Originate a voice call via Customer MGC to Verizon MGC that the Switch is unable to complete and is directed to a treatment announcement, for example, mis-dialed number.
2. Calling party disconnects the call.

Expected Results

1. Verify that the calling party hears the error tone/announcement.
2. Verify that the ACM with cause is mapped into a SIP 183 progressing message by Verizon MGC.
3. Verify that the SIP 183 progressing message contain an SDP.
4. Verify the following call flow.

```
Customer              Verizon                 Verizon       Class 3
 MGC                   Proxy                    MGC         Switch
 |                       |                       |             |
 |--------INVITE-------->|                       |             |
 |                       |--------INVITE-------->|             |
 |                       |                       |---- IAM---->|
 |                       |<------100 TRYING------|             |
 |<------100 TRYING------|                       |             |
 |                       |                       |             |
 |                       |                       |<----ACM-----|
 |                       |<-183 Session Progress-|             |
 |<-183 Session Progress-|                       |             |
 |                       |                       |             |
 |<<===========<===============early media=========<============<<|
                                    .
                                    .
                                    .
* Optional
```

6.2 Disconnect Procedure Test Cases

TC10: Calling party disconnects after the call is answered

Test Steps

1. Originate a voice call via customer MGC to Verizon MGC.
2. Answer the call. Calling party disconnects the call.

Expected Results

1. Verify that the calling party hears audible ring back.
2. Verify that the called party hears power ringing (alerted).
3. Verify that a two way speech path is established.
4. Verify Caller ID on the terminating line.
5. Verify the following call flow.

```
Customer    Verizon       Verizon        Class 3
 MGC         Proxy          MGC          Switch
 |             |             |             |
 |---INVITE--->|             |             |
```

```
   |              |---INVITE--->|              |
   |              |             |---- IAM---->|
   |              |<-100 TRYING-|              |
   |<-100 TRYING-|              |              |
   |              |              |              |
   |              |              |<----ACM-----|
   |              |<----18X-----|              |
   |<-----18x----|              |              |
   |              |              |              |
   |              |              |<----ANM-----|
   |              |<---200 OK---|              |
   |<---200 OK---|              |              |
   |              |              |              |
   |              |              |              |
   |              |              |              |
   |-----ACK---->|              |              |
   |              |-----ACK---->|              |
   |<==========Conversation=================>|
   |              |              |              |
   |-----BYE---->|              |              |
   |              |-----BYE---->|              |
   |              |              |---REL 16--->|
   |              |              |<----RLC-----|
   |              |<--200 OK----|              |
   |<--200 OK----|              |              |
   |              |              |              |
```

TC11: Called party disconnects after the call is answered

Test Steps

1. Originate a voice call via Customer MGC to Verizon MGC.
2. Answer the call. Called party disconnects the call.

Expected Results

1. Verify that the calling party does not receive audible ring back.
2. Verify the call is immediately answered.
3. Verify that a two way speech path is established.
4. Verify Caller ID on the terminating line.
5. Verify the following call flow.

```
Customer    Verizon       Verizon        Class 3
 MGC         Proxy          MGC          Switch
  |              |              |              |
  |---INVITE--->|              |              |
  |              |---INVITE--->|              |
  |              |              |---- IAM---->|
  |              |<-100 TRYING-|              |
  |<-100 TRYING-|              |              |
  |              |              |              |
  |              |              |<----ANM-----|
```

```
|             |<---200 OK---|             |
|<---200 OK---|             |             |
|             |             |             |
|-----ACK---->|             |             |
|             |-----ACK---->|             |
|<==========Conversation================>|
|             |             |             |
|             |             |<--REL 16----|
|             |<---BYE----- |             |
|<----BYE-----|             |             |
|             |             |----RLC----->|
|---200 OK--->|             |             |
|             |---200 OK--->|             |
|             |             |             |
```

TC12: Called party line busy – 486 Busy Here

Test Steps

1. Make the terminating line user busy.
2. Originate a voice call via Customer MGC to Verizon MGC.
3. Calling party goes back on hook.

Expected Results

1. Verify that the calling party hears a busy tone.
2. Verify a SIP 486 Busy Here message is generated at the Verizon MGC.
3. Verify the customer SIP UA interprets the 486 Busy Here to a Busy Signal.
4. Verify the following call release sequences.

```
Customer       Verizon          Verizon        Class 3
 MGC            Proxy            MGC           Switch
 |                |                |              |
 |-----INVITE---->|                |              |
 |                |-----INVITE---->|              |
 |                |                |---- IAM---->|
 |                |<--100 TRYING---|              |
 |<--100 TRYING---|                |              |
 |                |                |              |
 |                |                |              |
 |                |                |<--REL 17----|
 |                |<-486 Busy Here-|              |
 |<-486 Busy Here |                |              |
 |                |                |----RLC----->|
 |------ACK------>|                |              |
 |                |-------ACK----->|              |
 |                |                |              |
```

=== Busy tone played by the Customer SIP UA or Media Server===

TC13: Called party does not answer (hang up after approximately ten secconds)

Test Steps

1. Originate a voice call via Customer MGC to Verizon MGC.
2. Calling party goes on hook during ring phase.

Expected Results

1. Verify that the calling party hears audible ring back.
2. Verify that the called party hears power ringing (alerted).
3. Verify that the originator initiates disconnect procedures
4. Verify that the call is torn down and the circuits are released.
5. Verify that the SIP CANCEL message is generated by the Customer MGC.
6. Verify the following call release sequence.

```
Customer    Verizon       Verizon        Class 3
 MGC         Proxy          MGC          Switch
  |             |             |             |
  |---INVITE--->|             |             |
  |             |---INVITE--->|             |
  |             |             |---- IAM---->|
  |             |<-100 TRYING-|             |
  |<-100 TRYING-|             |             |
  |             |             |             |
  |             |             |<----ACM-----|
  |             |<----18X-----|             |
  |<-----18x----|             |             |
  |             |             |             |
  |             |             |             |
  |---CANCEL--->|             |             |
  |             |---CANCEL--->|             |
  |             |             |---REL 16--->|
  |             |<--200 OK----|             |
  |             |             |<----RLC-----|
  |<--200 OK----|             |             |
  |             |<----487-----|             |
  |<----487-----|             |             |
  |             |             |             |
  |-----ACK---->|             |             |
  |             |-----ACK---->|             |
```

6.3 Privacy Test Cases

TC14: Presentation restricted

Test Steps

1. Configure the originator to restrict presentation.
2. Originate a voice call via Customer MGC to Verizon MGC.

3. Answer the call. Calling party releases the call.

Expected Results

1. Verify that the calling party hears audible ring back.
2. Verify that the called party hears power ringing (alerted).
3. Verify that a two way speech path is established.
4. Verify the SIP INVITE contains the P-Asserted-Identity header with calling party information.
5. Verify the SIP PRIVACY header contains value = id

```
INVITE sip:+19725553265@verizon.com;user=phone SIP/2.0
Via:SIP/2.0/UDP 10.10.10.10;branch=71V5060-0-912782047
From:"Customer"<sip:+19727282400@10.10.10.10;user=phone>;tag=321064913
To:<sip:9725553265@verizon.com;user=phone>
Call-ID:13303212525050506744915018
CSeq:912782047 INVITE
Contact:<sip:10.10.10.10:5060>
P-Asserted-Identity:"Carrier IP Custname"<sip:+19727282400@10.10.10.10;user=phone>
Privacy:id
Allow:ACK,BYE,CANCEL,INFO,INVITE
Supported:
Accept:application/sdp,application/dtmf
Max-Forwards:10
Content-Type:application/sdp
Content-Length:292
```

6. Verify that Caller ID validates that presentation is restricted.

TC15: Calling party number is not present

Test Steps

1. Configure the originator so that it does not include the calling party information in the P-Asserted-Identity header in the INVITE message.
2. Originate a voice call via Customer MGC to Verizon MGC.

Expected Results

1. Verify the call is rejected by the Verizon Proxy with "403 Subscriber Not Found".

```
Customer                          Verizon                          Verizon
```

```
    MGC                        Proxy                      MGC
    |                            |                          |
    |-----------INVITE---------->|                          |
    |                            |----------INVITE--------->|
    |                            |                          |
    |                            |<-------100 TRYING--------|
    |<---------100 TRYING- -------|                          |
    |                            |                          |
    |                            |                          |
    |                            |<-403 Subscriber Not Found--|
    |<-403 Subscriber Not Found---|                          |
    |                            |                          |
    |------------ACK------------>|                          |
    |                            |------------ACK----------->|
    |                            |                          |
```

*Optional

6.4 CODEC Negotiation Test Cases

TC16: CODEC Offer / Answer with preferred codec accepted

Test Steps

1. Configure Customer MGC to support the CODEC precedence order, for example, PCMU, PCMA and G729.
2. The Verizon MGC codec precedence order is G729, PCMU and PCMA.
3. Originate a voice call via Customer MGC to Verizon MGC.
4. Answer the call. Calling party disconnects the call.

Expected Results

1. Verify that the calling party hears audible ring back.
2. Verify that the called party hears power ringing (alerted).
3. Verify that a two way speech path is established.
4. Verify that the customer's preferred CODEC is negotiated for media.
5. Verify that the correct CODECs are presented as shown in the "m=" lines below.

Offer: This example contains G.711 as the preferred CODEC.

v=0

o=24232 1 IN IP4 192.168.78.70

s=-

c=IN IP4 192.168.78.70

t=0 0

m=audio 50542 RTP/AVP 0 8 18 101

a=rtpmap:0 PCMU/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:18 G729/8000

Answer

v=0

o=PVG 0 0 IN IP4 192.168.78.253

s=-

p=+1 6135555555

c=IN IP4 192.168.78.253

t=0 0

m=audio 50700 RTP/AVP 0

a=ptime:20

a=rtpmap:0 PCMU/8000

TC17: Codec Offer / Answer with RFC 2833 supported on both MGCs

NOTE: If G.729 is the customer's preferred CODEC, RFC 2833 must be supported.

Test Steps

1. Configure both MGCs to support RFC 2833.
2. Originate a voice call via customer MGC to Verizon MGC to a line with a DTMF menu device.
3. Follow the menu prompts by addressing each one with the appropriate DTMF signal.

Expected Results

1. Verify that the calling party hears audible ring back.
2. Verify that the called party hears power ringing (alerted).
3. Verify that the menu prompts are heard.
4. Verify that the menu system responds correctly to the keypad entries.
5. Verify that "telephone-event" is present in the offer and answer.

Offer: This example uses payload type of 101. The supported range is 97 – 127, however 101 is most common.

v=0

o= 24232 1 IN IP4 192.168.78.70

s=-

c=IN IP4 192.168.78.70

t=0 0

m=audio 50542 RTP/AVP 0 8 18 **101**

a=rtpmap:0 PCMU/8000

a=rtpmap:8 PCMA/8000

a=rtpmap:18 G729/8000

a=rtpmap:101 telephone-event/8000

Answer

v=0

o=PVG 0 0 IN IP4 192.168.78.253

s=-

p=+1 6135555555

c=IN IP4 192.168.78.253

t=0 0

m=audio 50700 RTP/AVP 18 **101**

a=rtpmap:18 G729/8000

a=rtpmap:101 telephone-event/8000

TC18: T.30 fax call

Test Steps

1. Configure both MGCs to support T.30 fax.
2. Originate a fax call via customer MGC to Verizon MGC.
3. Transmit a three page fax.

Expected Results

1. Verify that the calling party hears audible ring back.
2. Verify that the called party hears power ringing (alerted).
3. Verify that the SIP messaging initially sets up a voice call.
4. Verify that when the Customer MGC detects fax preamble and that G.711 is re-negotiated if necessary.
5. Verify that the fax is transmitted successfully.

6.5 Implementation Test Cases

TC19: Alternate SIP interface using SRV DNS Record

Test Steps

1. Configure the Customer MGC to route calls to the Verizon Proxy using an SRV record with multiple end-point resolution.
2. Configure the primary SIP Proxy with a lower cost than the secondary SIP Proxy.
3. Make all of the resources on the primary SIP Proxy to the Verizon MGC unavailable.
4. Originate a voice call via the Customer MGC that advances to the alternate (secondary) SIP Proxy.
5. Answer the call. Calling party releases the call.

Expected Results

1. Verify that the calling party hears audible ring back.
2. Verify that the called party hears power ringing (alerted).
3. Verify that a two way speech path is established.
4. Verify that the alternate (secondary) SIP Proxy is used to complete the call.

TC20: Load Balance SIP interface using SRV DNS Record

Test Steps

1. Configure the Customer MGC to route calls to the Verizon Proxies using an SRV record with multiple end-point resolution.
2. Configure the primary and secondary SIP Proxy with an equal cost values in the SRV record.
3. Generate multiple calls from the customer MGC to the Verizon Proxy.
4. Validate the calls equally balance between the Verizon Proxy endpoints.

Expected Results

1. Verify that the calling party hears audible ring back.
2. Verify that the called party hears power ringing (alerted).
3. Verify that a two way speech path is established.
4. Verify the calls will load balance between the Verizon SIP Proxies.

TC21: Route to Verizon SIP Proxy using DNS A-Record

Test Steps

1. Configure the Customer MGC to route calls to the Verizon Proxy using an A record DNS name.
2. Originate a voice call via the Customer MGC that routes to the Verizon SIP Proxy using the A-record resolution.
3. Answer the call. Calling party releases the call.

Expected Results

1. Verify that the calling party hears audible ring back.

2. Verify that the called party hears power ringing (alerted).
3. Verify that a two way speech path is established.
4. Verify the call routes to the Verizon SIP Proxy using the A-Record DNS resolution.

7 Acronyms

ACK	Acknowledgment
ACM	Address Complete Message
ANM	Answer Message
ANSI	American National Standards Institute
CAS	Channel Associated Signaling
CIC	Carrier Identification Code
CPG	Call Progress Message
DTMF	Dial Tone Multi Frequency
FQDN	Fully Qualified Domain Name.
G.711	ITU standard for encoding telephone audio signals on a 64 Kbps channel – PCM scheme using an 8 bit sample at an 8 KHZ sample rate
GW	Gateway
IAM	Initial Address Message
IETF	Internet Engineering Task Force
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
ITU-T	International Telecommunication Union
LAN	Local Area Network
LNP	Local Number Portability
MG	Media Gateway
MGC	Media Gateway Controller
MTP	Message Transfer Part
PBX	Private Branch Exchange
POTS	Plain Old telephone service
PRI	Primary Rate Interface
PSTN	Public Service Telephone Network
REL	Release Message
RES	Resume Message
RFC	Request for Comments
RLC	Release Complete Message
RTP	Real Time Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SS7	Signaling System No. 7
TCP	Transmission Control Protocol
TDM	Time Division Multiplexing
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
VoIP	Voice over Internet Protocol
WAN	Wide Area Network

8 References

[1] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M. and E. Schooler, "SIP: Session Initiation Protocol", RFC 3261, June 2002.

[2] Peterson, J., "A Privacy Mechanism for the Session Initiation Protocol (SIP)", RFC 3323, November 2002.

[3] Schulzrinne, H., Petrack, S., "RTP Payload for DTMF Digits, Telephone Tones and Telephony Signals (SIP)", RFC 2833, May 2000.

[4] Jennings, C., Peterson, J., Watson, M., "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks", RFC 3325, November 2003.

Appendix A: Customer Connectivity Questionnaire

Customer Name: ________________________________

Contact: ________________________________

Phone Number(s): ________________________________

Email: ________________________________

DNS(es) (if required) for Soft-Switch/SBC:

- __________________
- __________________
- __________________

IPSEC Requirements

- **SIP Endpoints**
  - **Softswitch (s) IP address(es):**
    - ___________________
    - ___________________
    - ___________________
  - **Session Border Controller (if any) IP addresses:**
    - ___________________
    - ___________________

  - ____________________

- **Media Endpoints:**
  - **RTP endpoint IP address(es) or range:**
    - ____________________
    - ____________________
    - ____________________

- **Router(s) / Firewall(s):**
  - **Type of Firewall/Router (Cisco/Netscreen):**
    - ____________________
    - ____________________
    - ____________________
  - **Access Control List(s):**
    - ____________________
    - ____________________
    - ____________________
  - **IPSEC Peer IP addresses:**
    - ____________________
    - ____________________
    - ____________________

Appendix B: Verizon Interoperability Lab Connectivity Information Sheet:

Interop Lab Contact: ______________________________

Hot Line: ______________________________

Email: voipinterop@verizonbusiness.com

DNS (if required by customer) for Verizon Interop SIP proxy or Session Border Controller:

- **Verizon NS DNS : _________________**
- **Verizon SBC DNS: _________________**

IPSEC Requirements

- **Router(s) / Firewall(s):**
  - **IPSEC Peer Addresses:**
    - **Verizon Netscreen Peer: __________________**
    - **Customer Firewall/Router Peer: __________________**
    - **Customer Firewall/Router Peer: __________________**
  - **IP ISAKMP Crypto Key:**
    - **PHASE 1 information: __________________**
    - **PHASE 2 information: __________________**
    - **Verizon PRESHARED key: __________________**
- **SIP Endpoints**
  - **Proxy IP address(es):**
    - **Verizon NS: __________________**
    - **Verizon NS: __________________**
    - **Customer SIP Proxy or Softswitch: __________________**
    - **Customer SIP Proxy or Sotfswitch: __________________**
    - **Customer SIP Proxy or Softswitch: __________________**
  - **Session Border Controller (if any) IP addresses:**

- **Verizon SBC:** ___________________
- **Customer SBC:** ___________________
- **Customer SBC:** ___________________

- **Media Endpoints:**
  * **RTP endpoint address(es) or range:**
    - ___________________
    - ___________________
    - ___________________

DID numbers assigned for customer use during Interop testing only:

- ___________________
- ___________________
- ___________________
- ___________________
- ___________________