



# GENBAND™

## C20-A2 SIP Line Interoperability Test Plan

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**Vendor:** Yealink  
**Product Under Test:** T38G

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## 1 Document History

| Customer Issue | Reason   | Date        | Author             |
|----------------|--|-------------|--------------------|
| <b>1.0</b>     | Initial issue for Yealink IOT  | 4/18/2013   | Anilkumar Yadagiri |
| <b>1.1</b>     | Final report   | 5/10/2013   | Karas Shi          |
| <b>1.2</b>     | Revisions based on GENBAND Internal Review                                       | May 16 2013 | James Burnie       |
| <b>1.3</b>     | SIPIOT019 modified to more accurately reflect correct behavior with SMART phones | May 31 2013 | James Burnie       |

## 2 Scope

This test plan defines a service focused set of test cases to verify basic interoperability between SIP Clients and GENBAND's C20-A2 technology.

All calls are Intra C20-A2, meaning they are switched by the C20-A2 between the SIP Clients under test.

Media anchoring is accomplished in GENBAND's S3 Session Border Controller.

Most of the test cases are oriented towards the SIP clients and generally should not require any additional C20-A2 provisioning for execution or result assessment beyond the initial IOT lab setup implemented by GENBAND. However, in some cases, testers may need, and should request, GENBAND assistance to configure a particular test scenario.

## 3 Test Areas Covered

- Basic Communication using SIP
- Basic call line to line
- Basic call over trunk
- Announcement Test
- Basic Codec Negotiation and Fax
- Basic Feature Testing
- Custom Testing

## 4 IOT Activity Results Summary

All test case data/results should be recorded by inserting comments in appropriate areas of the test plan, including Record of Execution, Test Case Summary (Pass or Fail status) and Test Results.

GENBAND requests each IOT customer to return a copy of this document containing sufficient result information to enable preparation of a meaningful IOT Activity Report. Packet traces should be supplied for all test cases along with relevant product configuration information.

| Pass      | Fail | Not Executed | Comments  |
|-----------|------|--------------|---|
| <b>27</b> | 0    | 1            | SIPIOT016: Not Executed because T38G does not support connection to analog FAX device |

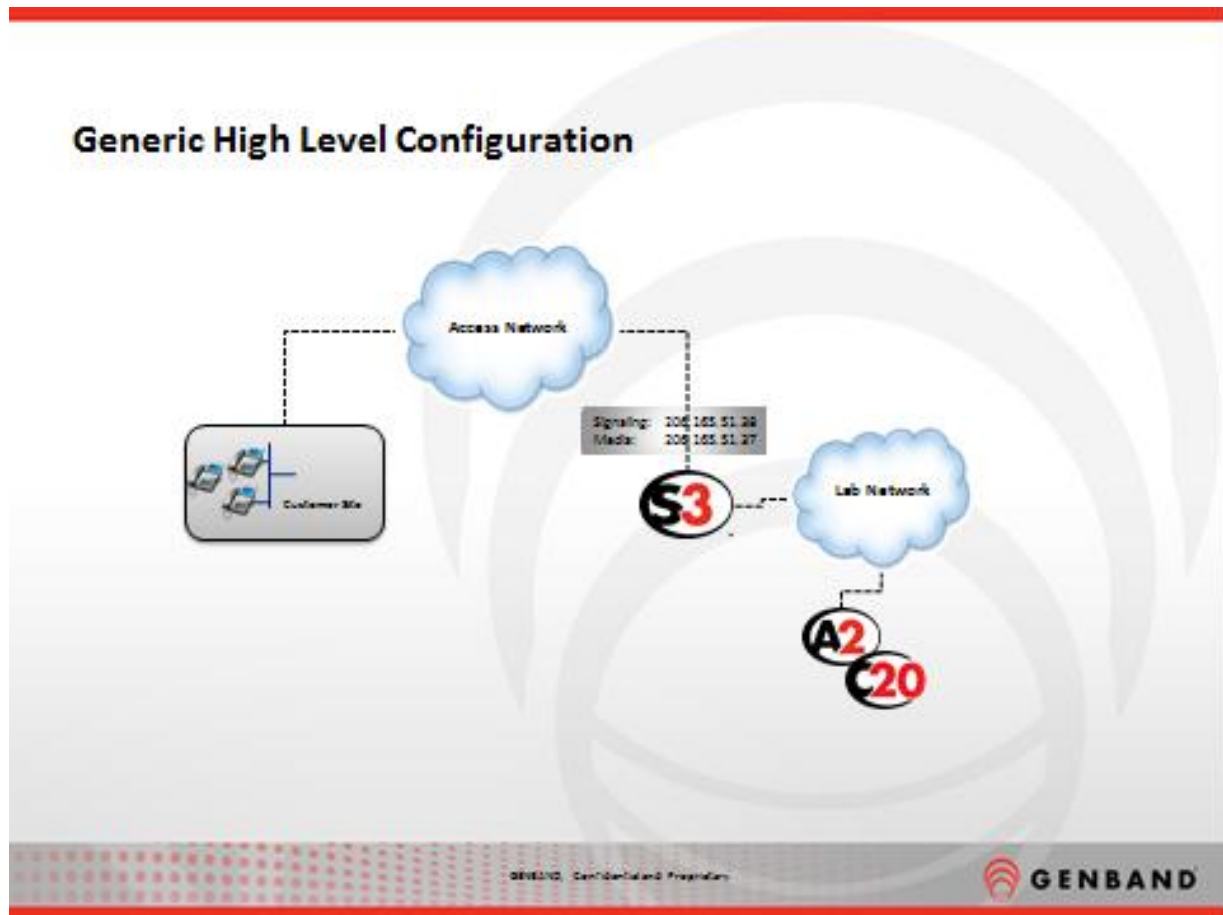


## 5 Quick Start Guide

1. Assign [SIP Proxy](#), [Domain](#), [UserID](#) and [Password](#) information to all SIP Clients under test. Note that each DN has a unique UserID.
2. Verify successful SIP Registration for all Clients under test
3. Verify successful line to line calls between all Registered SIP Clients according to the [Dial Plan](#)
4. Execute the [Test Cases](#)
  - a. record any issues or special execution criteria in the areas provided in each test case
  - b. update the [Record of Execution](#) for each test case
  - c. record/capture relevant [configuration information](#) for the SIP clients under test

## 6 IOT Activity Configuration

### 6.1 Lab Diagram



### 6.2 Access Information

| Item      | Value         |
|-----------|---------------|
| SIP Proxy | 206.165.51.38 |



| Item                            | Value  |
|---------------------------------|--|
| Protocol                        | SIP  |
| Port                            | 5060   |
| Domain                          | gb.ott7.iot  |
| Registration and Authentication | SIP Clients must register and authenticate with the <a href="#">UserID and Password</a> specified for a given DN |

### 6.3 Line Information

Lines have been provisioned for this IOT activity per the following table. The common list of features includes

- Public Name
- Calling Name and Number Display
- Calling Name and Number Blocking
- Call Waiting with Caller ID
- 3 Way Calling
- Call Forward Universal
- Call Forward No Answer (fixed to Voice Mail for some lines)
- Call Forward Busy (fixed to Voice Mail for some lines)
- Call Transfer
- Message Waiting (only on lines configured with Voice Mail)

| DN         | UserID   | Password | Public Name/Caller ID | Feature List   |
|------------|----------|----------|-----------------------|--|
| 9199928220 | u9928220 | 9232     | YEALINK 1             | CWT, 3WC, CCW, DGT, NAME PUBLIC YEALINK 1, CFW, CNDB, SCWID, CNAB, DDN, CNAMD, CFDA, CFBL, CXR, DPL, AGNTPCL                       |
| 9199928221 | u9928221 | 9232     | YEALINK 2             | CWT, 3WC, CCW, DGT, NAME PUBLIC YEALINK 2, CFW, CNDB, SCWID, CNAB, DDN, CNAMD, CFDA, CFBL, CXR, DPL, AGNTPCL                       |
| 9199928222 | u9928222 | 9232     | YEALINK 3             | CWT, 3WC, CCW, DGT, NAME PUBLIC YEALINK 3, CFW, CNDB, SCWID, CNAB, DDN, CNAMD, CFDA, CFBL, CXR, DPL, AGNTPCL                       |
| 9199928223 | u9928223 | 9232     | YEALINK 4             | CWT, 3WC, CCW, DGT, NAME PUBLIC YEALINK 4, CNDB, SCWID, CNAB, DDN, CNAMD, CFDA 9199924000, CFBL 9199924000, CXR, MWT, DPL, AGNTPCL |

### 6.4 Dial Plan

| Call Type             | Dialing Method                                |
|-----------------------|---|
| Line to Line          | 7 or 10 Digits                                |
| Line to Trunk to Line | <a href="#">Trunk Access Code</a> + 10 digits |



| Call Type                      | Dialing Method  |
|--------------------------------|---|
| Line to Blank Directory Number | 9199920000  |
| Features                       | <a href="#">Vertical Service Code</a> +<br>10 digits except for “toggle” VSC codes such as Cancel CFW |

## 6.5 Trunk Access Codes

Use Trunk Access Codes to route line originated calls over specific loop-around trunk group types

| Trunk Type                   | Code | Remarks                             |
|------------------------------|------|-------------------------------------|
| <a href="#">PVG</a> Loopback | 602  | SIP clients must support 20ms ptime |
| <a href="#">SST</a> Loopback | 610  |                                     |

## 6.6 Vertical Service Codes

Use Vertical Service Codes to activate/deactivate specific call features

| Feature | Code | Description  |
|---------|------|--|
| CCW     | *70  | <ul style="list-style-type: none"> <li>Cancel Call Waiting per call</li> <li>*70+Called Digits</li> </ul>  |
| CFWP    | *72  | <ul style="list-style-type: none"> <li>Activate Call Forward Unconditional</li> <li>*72+Digits of number to which calls will be forwarded</li> </ul>   |
| CFWC    | *73  | <ul style="list-style-type: none"> <li>Cancel Call Forward Unconditional</li> <li>*73#</li> </ul>  |
| CNDB    | *88  | <ul style="list-style-type: none"> <li>Calling Number Display Blocking per call</li> <li>Use to block delivery of the originating line’s Number to the called DN</li> <li>*88+Called Digits</li> </ul>                     |
| CNAB    | *90  | <ul style="list-style-type: none"> <li>Calling Name Display Blocking per call</li> <li>Use to block delivery of the originating line’s Name to the called DN</li> <li>*90+Called Digits</li> </ul>                         |
| CNNB    | *67  | <ul style="list-style-type: none"> <li>Calling Name and Number Blocking per call</li> <li>Use to block delivery of the originating lines Name and Number to the called DN</li> <li>*67+Called Digits</li> </ul>            |
| CFBP    | *28  | <ul style="list-style-type: none"> <li>Activate Call Forward Busy</li> <li>For lines not equipped with VM: *28+Digits of number to which calls will be forwarded</li> <li>For lines equipped with VM: *28#</li> </ul>      |
| CFBC    | *29  | <ul style="list-style-type: none"> <li>Cancel Call Forward Busy</li> <li>*29#</li> </ul>   |
| CFDP    | *30  | <ul style="list-style-type: none"> <li>Activate Call Forward No Answer</li> <li>For lines not equipped with VM: *30+Digits of number to which calls will be forwarded</li> <li>For lines equipped with VM: *30#</li> </ul> |
| CFDC    | *31  | <ul style="list-style-type: none"> <li>Cancel Call Forward No Answer</li> <li>*31#</li> </ul>  |



## 6.7 Voicemail

VM has been enabled on 9199928223 only. It therefore has Call Forwarding for Busy and No Answer conditions set to forward to the Voice Mail DN (9199924000). The forwarding destination (VM) for 9199928223 cannot be modified by the user although the user is able to activate/deactivate CFBL and CFDA via the appropriate [Vertical Service Codes](#).

Users may retrieve messages from their Voice Mail account by dialing 9199924111, password 0000#. Note that message retrieval will always be for the line that is initiating the call to 9199924111.

## 6.8 Additional Equipment

Provide detailed information on additional devices, if applicable, used during testing.

| Device | Model | Firmware version |
|--------|-------|------------------|
|        |       |                  |
|        |       |                  |
|        |       |                  |
|        |       |                  |

## 6.9 Product Configuration

### 6.9.1 T38G

1. Reset Phone to Factory Defaults  
In the phone screen, press menu->settings->advanced settings->input "admin" as default password->Reset to factory
2. Connect to Phone's WEB Interface:  
Determine IP address of phone by pressing the "OK" button once the Factory Reset/Restart is complete  
Input Phone's IP address to WEB Browser  
Login with userid=admin and password=admin (factory defaults)
3. Configure the SIP account:  
In Account->Basic, input your SIP account information then press Confirm  
Verify that the account status is Registered
4. Configure Conferencing and Message Waiting:  
Account->Advanced->Conference Type: Network  
Account->Advanced->Conference URI: Conference  
Account->Advanced->Subscription for MWI: Enabled

### 6.9.2 GENBAND SIP Profile

The SIP Profile used in conjunction with the T38G for this IOT activity is stored in the following GENBAND Sharepoint location:

<https://portal.genband.com/sites/Dev/SRL/SV/IOT%20SV/Docs/Projects/C20/Yealink/Results/T-38G%20Client>

## 6.10 Definitions

| Term | Expansion |
|------|-----------|
|------|-----------|



| Term        | Expansion                                  |
|-------------|--|
| 3WC         | 3-way calling                              |
| CFD or CFDA | Call forward no answer                     |
| CFB or CFBL | Call forward busy line                     |
| CFW or CFU  | Call forward unconditional                 |
| CXR         | Call transfer                              |
| CWT         | Call waiting                               |
| CCW         | Cancel call waiting (per call)             |
| CNAB        | Calling name display blocking (per call)   |
| CNDB        | Calling number display blocking (per call) |
| CXR         | Call Transfer                              |
| PVG         | Packet Voice Gateway (TDM)                 |
| SST         | Session Server Trunks (SIP)                |
| VM          | Voicemail                                  |
| 3PSL        | Third Party SIP Line                       |



## 7 Test Case List and Record of Execution

### 7.1 SIP Registration

| Test Case                                    | Test Prime  | Product Under Test | Product SW/FW Version | C20-A2 SW Version | Result Pass/Fail/NE | Execution Date |
|--|-------------|--------------------|-----------------------|-------------------|---------------------|----------------|
| <a href="#">SIPIOT001 – SIP Registration</a> | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 3,2013     |

### 7.2 Basic Calls Line to Line

| Test Case  | Test Prime  | Product Under Test | Product SW/FW Version | C20-A2 SW Version | Result Pass/Fail/NE | Execution Date |
|--|-------------|--------------------|-----------------------|-------------------|---------------------|----------------|
| <a href="#">SIPIOT002 - 3PSL to 3PSL, Originator disconnects after answer</a>  | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 3,2013     |
| <a href="#">SIPIOT003 - 3PSL to 3PSL, Terminator disconnects after answer</a>  | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 3,2013     |
| <a href="#">SIPIOT004 - 3PSL to 3PSL, Originator disconnects before answer</a> | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 3,2013     |
| <a href="#">SIPIOT005 - 3PSL to 3PSL, Call to BUSY Line</a>                    | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 3,2013     |
| <a href="#">SIPIOT006 - 3PSL Long Duration Call</a>                            | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 3,2013     |

### 7.3 Basic Calls over Trunk

| Test Case   | Test Prime  | Product Under Test | Product SW/FW Version | C20-A2 SW Version | Result Pass/Fail/NE | Execution Date |
|---|-------------|--------------------|-----------------------|-------------------|---------------------|----------------|
| <a href="#">SIPIOT007 - 3PSL and SIP trunk, Originator disconnects after answer</a> | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 3,2013     |
| <a href="#">SIPIOT008 - 3PSL and TDM trunk, Originator disconnects after answer</a> | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 3,2013     |



| Test Case  | Test Prime  | Product Under Test | Product SW/FW Version | C20-A2 SW Version | Result Pass/Fail/NE | Execution Date |
|--|-------------|--------------------|-----------------------|-------------------|---------------------|----------------|
| <a href="#">SIPIOT009 - 3PSL and SIP trunk, Terminator disconnects after answer</a>  | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT010 - 3PSL and TDM trunk, Terminator disconnects after answer</a>  | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT011 - 3PSL and SIP trunk Originator disconnects before answer</a>  | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT012 - 3PSL and TDM trunk, Originator disconnects before answer</a> | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |

#### 7.4 Announcements

| Test Case   | Test Prime  | Product Under Test | Product SW/FW Version | C20-A2 SW Version | Result Pass/Fail/NE | Execution Date |
|---|-------------|--------------------|-----------------------|-------------------|---------------------|----------------|
| <a href="#">SIPIOT013 - 3PSL to C20-A2 Announcement</a> | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |

#### 7.5 Basic Codec Negotiation and Fax

| Test Case  | Test Prime  | Product Under Test | Product SW/FW Version | C20-A2 SW Version | Result Pass/Fail/NE | Execution Date |
|--|-------------|--------------------|-----------------------|-------------------|---------------------|----------------|
| <a href="#">SIPIOT014 - 3PSL to 3PSL codec negotiation use G729 and G711 variants with 20ms PR</a> | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT015 - 3PSL to 3PSL (RFC 2833 DTMF)</a>   | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT016 – FAX connection using T.38</a>  | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | NE                  | May 8,2013     |



## 7.6 Basic Feature Testing

| Test Case   | Test Prime  | Product Under Test | Product SW/FW Version | C20-A2 SW Version | Result Pass/Fail/NE | Execution Date |
|---|-------------|--------------------|-----------------------|-------------------|---------------------|----------------|
| <a href="#">SIPIOT017 – HOLD/RESUME</a>                                 | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT018 - Blind Call Transfer</a>                         | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT019 – Consultative Call Transfer</a>                  | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT020 - 3 Way Call conference</a>                       | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT021 - Ad Hoc Conference</a>                           | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT022 - Call Forward no answer</a>                      | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT023 - Call Forward Busy</a>                           | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT024 - Call Forward Immediate</a>                      | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT025 - Call Waiting</a>                                | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT026 - Calling Name/Number Display</a>                 | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT027 - Calling Name/Number Blocked (at Originator)</a> | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |
| <a href="#">SIPIOT028 – Voice Mail Message waiting indicator</a>        | ZhiGang Cai | SIP-T38G           | 38.70.0.125           | CVM17             | Pass                | May 8,2013     |

## 7.7 Custom Testing (Vendor Specific Testing)

| Test Case  | Test Prime | Product Under Test | Product SW/FW Version | C20-A2 SW Version | Result Pass/Fail/NE | Execution Date |
|--|------------|--------------------|-----------------------|-------------------|---------------------|----------------|
| <a href="#">SIPIOT9XX - Example Custom Test case</a> |            |                    |                       |                   |                     |                |



## 8 Test Case Details and Results

### 8.1 SIP Registration

This section covers basic registration between the 3<sup>rd</sup> party SIP line (3PSL) and GENBAND C20-A2.

#### 8.1.1 SIPIOT001 – SIP Registration

|                       |   |
|-----------------------|---|
| Test Area             | SIP Registration  |
| Test Title            | 3PSL Registration   |
| Objective             | Verify successful SIP REGISTRATION with C20-A2  |
| Configuration & Setup | <p>A: SIP Endpoint under test</p> <ol style="list-style-type: none"> <li>1. Packet trace tool required to capture and decode the IP messaging</li> </ol>  |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. Start the trace tool</li> <li>2. Configure A with an appropriate <a href="#">USER</a> and invoke SIP REGISTRATION; wait 30 seconds</li> <li>3. Stop the trace tool and analyze the trace</li> <li>4. Verify A initiates a REGISTER request with a non-zero “expires” value</li> <li>5. Verify A receives a “200 OK” or “200 Registration Successful” response with a non-zero “expires” value</li> <li>6. Start the trace tool</li> <li>7. Invoke de-REGISTRATION at A; wait 30 seconds</li> <li>8. Stop the trace tool and analyze the trace</li> <li>9. Verify A initiates a REGISTER request with “expires” = 0</li> <li>10. Verify A receives a “200 OK” response with “expires” = 0 or a “200 Registration Successful” response</li> </ol> |
| Expected Results      | Successful outcomes for all verification steps  |
| Test Outcome          | Pass  |
| Issues                |   |
| Execution Notes       |   |



## 8.2 Basic Call Line to Line

The purpose of this section is to verify that basic line to line calls can be placed between 3PSL and 3PSL via C20-A2. These basic scenarios must function properly before more complex call control features can be tested.

### 8.2.1 SIPIOT002 - 3PSL to 3PSL, Originator disconnects after answer

|                       |  |
|-----------------------|--|
| Test Area             | Basic Call Line to Line  |
| Test Title            | 3PSL to 3PSL, Originator disconnects after answer  |
| Objective             | To verify basic line to line call when originator disconnects after the call is answered   |
| Configuration & Setup | A: SIP end point under test<br>B: SIP end point under test<br><br>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2  |
| Test Procedures       | 1. Place a call from A to B<br>2. Verify B is alerted ( ringing )<br>3. Verify A can hear audible ringing<br>4. B answers the call from A<br>5. Verify A and B can talk to each other with bi-directional speech path.<br>6. A disconnects the call as originator<br>7. Verify and A and B can make another call |
| Expected Results      | Successful outcomes for all verification steps   |
| Test Outcome          | Pass   |
| Issues                |  |
| Execution Notes       |  |

### 8.2.2 SIPIOT003 - 3PSL to 3PSL, Terminator disconnects after answer

|            |  |
|------------|--|
| Test Area  | Basic Call Line to Line  |
| Test Title | 3PSL to 3PSL, Terminator disconnects after answer                          |
| Objective  | To verify basic line to line call when terminator disconnects after answer |



|                       |   |
|-----------------------|---|
| Configuration & Setup | A: SIP end point under test<br>B: SIP end point under test<br><br>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2   |
| Test Procedures       | 1. Place a call from B to A<br>2. Verify A is alerted ( ringing )<br>3. Verify B can hear audible ringing<br>4. A answers the call from B<br>5. Verify B and A can talk to each other with bi-directional speech path<br>6. A disconnects the call as terminator<br>7. Verify A and B can make another call |
| Expected Results      | Successful outcomes for all verification steps  |
| Test Outcome          | Pass  |
| Issues                |   |
| Execution Notes       |   |

### 8.2.3 SIPIOT004 - 3PSL to 3PSL, Originator disconnects before answer

|                       |   |
|-----------------------|---|
| Test Area             | Basic Call Line to Line   |
| Test Title            | 3PSL to 3PSL, Originator disconnects before answer  |
| Objective             | To verify basic line to line call when originator disconnects before answer   |
| Configuration & Setup | A: SIP end point under test<br>B: SIP end point under test<br>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2                                       |
| Test Procedures       | 1. Place a call from A to B<br>2. Verify B is alerted ( ringing )<br>3. Verify A can hear audible ringing<br>4. B does not answer the call<br>5. A disconnects<br>6. Verify B stops ringing |
| Expected Results      | Successful outcomes for all verification steps  |
| Test Outcome          | Pass  |



|                 |  |
|-----------------|--|
| Issues          |  |
| Execution Notes |  |

#### 8.2.4 SIPIOT005 - 3PSL to 3PSL, Call to BUSY Line

|                       |   |
|-----------------------|---|
| Test Area             | Basic Call Line to Line   |
| Test Title            | 3PSL to 3PSL, Busy timeout  |
| Objective             | To verify basic line to line call busy timeout  |
| Configuration & Setup | <p>A: SIP end point under test<br/>           B: SIP end point under test<br/>           C: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A, B and C with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> <li>2. Cancel all Call Forwarding Variants at A using the appropriate <a href="#">Vertical Service Codes</a></li> </ol> |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. A calls B using the Cancel Call Waiting per call <a href="#">Vertical Service Code</a></li> <li>2. B answers the call</li> <li>3. Verify bi-directional speech path between A and B</li> <li>4. C calls A</li> <li>5. Verify C hears busy tone ( Since A is in conversation with B )</li> <li>6. Hang up A, B and C</li> </ol>                |
| Expected Results      | Successful outcomes for all verification steps  |
| Test Outcome          | Pass  |
| Issues                |   |
| Execution Notes       |   |



### 8.2.5 SIPIOT006 - 3PSL Long Duration Call

|                       |   |
|-----------------------|---|
| Test Area             | Basic Call Line to Line   |
| Test Title            | 3PSL Long Duration Call   |
| Objective             | Verify that a call between two 3PSLs under test can remain connected successfully for at least 1 hour   |
| Configuration & Setup | A: SIP end point under test<br>B: SIP end point under test<br><br>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER in C20-  |
| Test Procedures       | 1. A calls B<br>2. Verify B is alerted (ringing)<br>3. Verify A hears audible ringing<br>4. B answers the call<br>5. Verify bi-directional speech path<br>6. Leave the call up for at least 1 hour.<br>7. Verify bi-directional speech path again |
| Expected Results      | Successful outcomes for all verification steps  |
| Test Outcome          | Pass  |
| Issues                |   |
| Execution Notes       |   |



### 8.3 Basic Calls over Trunk

The purpose of this section is to verify that basic calls that can be placed between two 3PSL hosted endpoints over various C20 trunks.

#### 8.3.1 SIPIOT007 - 3PSL and SIP trunk, Originator disconnects after answer

|                       |  |
|-----------------------|--|
| Test Area             | Basic Calls over Trunk   |
| Test Title            | 3PSL and SIP trunk; Originator disconnects after answer  |
| Objective             | To verify correct outcomes for calls between two 3PSLs placed over a SIP trunk when the originator disconnects after the call is answered  |
| Configuration & Setup | A: SIP end point under test<br>B: SIP end point under test<br><br>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2  |
| Test Procedures       | 1. A calls B using the <a href="#">SST loopback trunk access code</a><br>2. Verify B is alerted (ringing),<br>3. Verify A hears audible ringing<br>4. B answers the call<br>5. Verify bi-directional speech path<br>6. A disconnects the call<br>7. Verify A and B can make another call |
| Expected Results      | Successful outcomes for all verification steps   |
| Test Outcome          | Pass   |
| Issues                |  |
| Execution Notes       |  |

#### 8.3.2 SIPIOT008 - 3PSL and TDM trunk, Originator disconnects after answer

|            |   |
|------------|---|
| Test Area  | Basic Calls over Trunk  |
| Test Title | 3PSL and TDM trunk; Originator disconnects after answer   |
| Objective  | To verify correct outcomes for calls between two 3PSLs placed over a TDM trunk when the originator disconnects after the call is answered |



|                       |   |
|-----------------------|---|
| Configuration & Setup | A: SIP end point under test<br>B: SIP end point under test<br><br><ol style="list-style-type: none"> <li>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> <li>2. A and B must support 20ms ptime</li> </ol>   |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. A calls B using the <a href="#">PVG loopback trunk access code</a></li> <li>2. Verify B is alerted (ringing)</li> <li>3. Verify A hears audible ringing</li> <li>4. B answers the call</li> <li>5. Verify bi-directional speech path</li> <li>6. A disconnects the call</li> <li>7. Verify A and B can make another call</li> </ol> |
| Expected Results      | Successful outcomes for all verification steps  |
| Test Outcome          | Pass  |
| Issues                |   |
| Execution Notes       |   |

### 8.3.3 SIPIOT009 - 3PSL and SIP trunk, Terminator disconnects after answer

|                       |  |
|-----------------------|--|
| Test Area             | Basic Calls over Trunk   |
| Test Title            | 3PSL and SIP trunk, Terminator disconnects after answer  |
| Objective             | To verify correct outcomes for calls between two 3PSLs placed over a SIP trunk when the terminator disconnects after the call is answered  |
| Configuration & Setup | A: SIP end point under test<br>B: SIP end point under test<br><br><ol style="list-style-type: none"> <li>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> </ol>  |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. A calls B using the <a href="#">SST loopback trunk access code</a></li> <li>2. Verify B is alerted (ringing),</li> <li>3. Verify A hears audible ringing</li> <li>4. B answers the call</li> <li>5. Verify bi-directional speech path</li> <li>6. B disconnects the call</li> <li>7. Verify A and B can make another call</li> </ol> |



|                  |  |
|------------------|--|
| Expected Results | Successful outcomes for all verification steps |
| Test Outcome     | Pass   |
| Issues           |  |
| Execution Notes  |  |

#### 8.3.4 SIPIOT010 - 3PSL and TDM trunk, Terminator disconnects after answer

|                       |  |
|-----------------------|--|
| Test Area             | Basic Calls over Trunk   |
| Test Title            | 3PSL and TDM trunk, Terminator disconnects after answer  |
| Objective             | To verify correct outcomes for calls between two 3PSLs placed over a TDM trunk when the terminator disconnects after the call is answered  |
| Configuration & Setup | A: SIP end point under test<br>B: SIP end point under test<br><br>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2<br>2. A and B must support 20ms ptime  |
| Test Procedures       | 1. A calls B using the <a href="#">PVG loopback trunk access code</a><br>2. Verify B is alerted (ringing).<br>3. Verify A hears audible ringing<br>4. B answers the call<br>5. Verify bi-directional speech path<br>6. B disconnects the call<br>7. Verify A and B can make another call |
| Expected Results      | Successful outcomes for all verification steps   |
| Test Outcome          | Pass   |
| Issues                |  |
| Execution Notes       |  |



### 8.3.5 SIPIOT011 - 3PSL and SIP trunk Originator disconnects before answer

|                       |  |
|-----------------------|--|
| Test Area             | Basic Calls over Trunk   |
| Test Title            | 3PSL and SIP trunk Originator disconnects before answer  |
| Objective             | To verify correct outcomes for calls between two 3PSLs placed over a SIP trunk when the originator disconnects before the call is answered   |
| Configuration & Setup | A: SIP end point under test<br>B: SIP end point under test<br><br><ol style="list-style-type: none"> <li>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> </ol>  |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. A calls B using the <a href="#">SST loopback trunk access code</a></li> <li>2. Verify B is alerted (ringing).</li> <li>3. Verify A hears audible ringing</li> <li>4. A disconnects before B answers the call</li> <li>5. Verify B stops ringing</li> </ol> |
| Expected Results      | Successful outcomes for all verification steps   |
| Test Outcome          | Pass   |
| Issues                |  |
| Execution Notes       |  |

### 8.3.6 SIPIOT012 - 3PSL and TDM trunk, Originator disconnects before answer

|                       |   |
|-----------------------|---|
| Test Area             | Basic Calls over Trunk  |
| Test Title            | 3PSL and TDM trunk, Originator disconnects before answer  |
| Objective             | To verify correct outcomes for calls between two 3PSLs placed over a TDM trunk when the originator disconnects before the call is answered  |
| Configuration & Setup | A: SIP end point under test<br>B: SIP end point under test<br><br><ol style="list-style-type: none"> <li>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> <li>2. A and B must support 20ms ptime</li> </ol> |



|                  |  |
|------------------|--|
| Test Procedures  | <ol style="list-style-type: none"> <li>1. A calls B using the <a href="#">PVG loopback trunk access code</a></li> <li>2. Verify B is alerted (ringing).</li> <li>3. Verify A hears audible ringing</li> <li>4. A disconnects before B answers the call</li> <li>5. Verify B stops ringing</li> </ol> |
| Expected Results | Successful outcomes for all verification steps   |
| Test Outcome     | Pass   |
| Issues           |  |
| Execution Notes  |  |

## 8.4 Announcements

This section verifies the ability of the 3PSL to connect to C20-A2 announcements.

### 8.4.1 SIPIOT013 - 3PSL to C20-A2 Announcement

|                       |   |
|-----------------------|---|
| Test Area             | Announcements   |
| Test Title            | 3PSL to C20-A2 Announcement   |
| Objective             | To verify 3PSL to C20-A2 announcements  |
| Configuration & Setup | <p>A: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A with an appropriate <a href="#">USER</a> and REGISTER in C20-A2</li> </ol> |



|                  |   |
|------------------|---|
| Test Procedures  | <ol style="list-style-type: none"> <li>1. Place a call from A to a <a href="#">Blank Directory Number</a></li> <li>2. Verify that A is sent to a recorded announcement</li> </ol> |
| Expected Results | Successful outcomes for all verification steps  |
| Test Outcome     | Pass  |
| Issues           |   |
| Execution Notes  |   |

## 8.5 Basic Codec Negotiation and Fax

The purpose of this section is to validate proper codec negotiation between the 3PSL and other C20-A2 hosted endpoints (including announcements and trunks). RTP Packet Analysis will be performed for each test case.

### 8.5.1 SIPIOT014 - 3PSL to 3PSL codec negotiation use G729 and G711 variants with 20ms PR

|            |  |
|------------|--|
| Test Area  | Basic Codec Negotiation and FAX  |
| Test Title | 3PSL to 3PSL codec negotiation use G729 and G711 variants with 20ms PR         |
| Objective  | Verify line to line call using G729, G711A, G711u with 20ms packetization rate |



|                       |  |
|-----------------------|--|
| Configuration & Setup | <p>A: SIP end point under test with G729 and 20ms packetization enforced<br/>B: SIP end point under test supporting G711A, G711u and G729</p> <ol style="list-style-type: none"> <li>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> <li>2. For each test scenario, set up B to use codec G729, G711A, G711u with 20ms packetization rate</li> <li>3. Packet trace tool required to capture and decode the IP messaging</li> </ol>  |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. Start trace tool</li> <li>2. Place a call from A to B</li> <li>3. Verify B is alerted (ringing).</li> <li>4. Verify A hears audible ring-back</li> <li>5. B answers the call</li> <li>6. Verify bi-directional speech path</li> <li>7. A &amp; B disconnect</li> <li>8. Stop trace tool</li> <li>9. Analyze traces collected</li> <li>10. Verify G729 and 20ms packetization rate are used.</li> <li>11. Repeat step 1-&gt;9 with A enforced with G711u</li> <li>12. Verify G711u and 20ms packetization rate are used</li> <li>13. Repeat step 1-&gt;9 with A enforced with G711A</li> <li>14. Verify G711A and 20ms packetization rate are used</li> </ol> |
| Expected Results      | Successful outcomes for all verification steps   |
| Test Outcome          | Pass   |
| Issues                |  |
| Execution Notes       |  |

### 8.5.2 SIPIOT015 - 3PSL to 3PSL (RFC 2833 DTMF)

|                       |   |
|-----------------------|---|
| Test Area             | Basic Codec Negotiation and FAX   |
| Test Title            | 3PSL to 3PSL (RFC 2833 DTMF)  |
| Objective             | Verify line to line RFC2833 DTMF via voicemail  |
| Configuration & Setup | <p>A: SIP end point under test<br/>B: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2 in C20-A2; B's USER must support <a href="#">Voice Mail</a></li> <li>2. Activate Call Forward No Answer at B using the appropriate <a href="#">Vertical Service Code</a></li> </ol> |



|                  |   |
|------------------|---|
|                  | <ol style="list-style-type: none"> <li>3. Configure A to use RFC 2833 DTMF</li> <li>4. Packet trace tool required to capture and decode the IP messaging</li> </ol>   |
| Test Procedures  | <ol style="list-style-type: none"> <li>1. Start the trace tool</li> <li>2. A calls B</li> <li>3. Allow B to ring until it forwards to Voice Mail</li> <li>4. Verify A is connected to B's Voice Mail</li> <li>5. Press # at A during the Voice Mail Greeting</li> <li>6. Verify the VM Greeting is interrupted</li> <li>7. Leave a message and press #</li> <li>8. Verify VM presents menu of options</li> <li>9. Press # at A</li> <li>10. Verify session terminates</li> <li>11. Stop trace tool</li> <li>12. Analyze contents of trace file</li> <li>13. Verify that the 3 #'s are present as RFC 2833 events in the RTP stream</li> </ol> |
| Expected Results | Successful outcomes for all verification steps  |
| Test Outcome     | Pass  |
| Issues           |   |
| Execution Notes  |   |

### 8.5.3 SIPIOT016 – FAX connection using T.38

|                       |  |
|-----------------------|--|
| Test Area             | Basic Codec Negotiation and FAX  |
| Test Title            | FAX connection using T.38  |
| Objective             | Verify successful T.38 FAX call  |
| Configuration & Setup | <p>NOTE: this test case only applies to SIP Clients that can be physically connected to a FAX machine - usually limited to SIP Analog Telephone Adapter (ATA) devices</p> <p>A: SIP end point under test<br/>B: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> <li>2. Configure T.38 and G.729 on both A and B</li> <li>3. Connect FAX machines to A and B.</li> <li>4. Packet trace tool required to capture and decode the IP messaging</li> </ol> |



|                  |  |
|------------------|--|
| Test Procedures  | <ol style="list-style-type: none"> <li>1. Start the trace tool</li> <li>2. A calls B</li> <li>3. Verify B is alerted (ringing).</li> <li>4. B answers</li> <li>5. A and B negotiate fax connection</li> <li>6. Verify B successfully receives all pages of the fax from A</li> <li>7. A &amp; B disconnect after fax-sending is complete</li> <li>8. Stop the trace tool</li> <li>9. Analyze the trace captured</li> <li>10. Verify the call is initially set up using G.729 and then successful negotiation of T.38 for the fax transmission</li> </ol> |
| Expected Results | Successful outcomes for all verification steps   |
| Test Outcome     | Not Executed   |
| Issues           | Not applicable   |
| Execution Notes  | T28P does not support connection to Analog FAX device  |

## 8.6 Basic Feature Testing

The purpose of this section is to validate C20-A2 line based features using 3PSL. The feature under test will be enabled on the agents involved in the call.

### 8.6.1 SIPIOT017 – HOLD/RESUME

|                       |  |
|-----------------------|--|
| Test Area             | Basic Feature Testing  |
| Test Title            | HOLD/RESUME  |
| Objective             | To verify basic line to line HOLD/RESUME feature   |
| Configuration & Setup | <p>A: SIP end point under test<br/>B: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> <li>2. Packet trace tool required to capture and decode the IP messaging</li> </ol> |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. Start the packet trace tool</li> <li>2. A calls B</li> <li>3. Verify B is alerted (ringing)</li> <li>4. Verify A hears audible ringing</li> </ol>  |



|                  |   |
|------------------|---|
|                  | <ol style="list-style-type: none"> <li>5. B answers</li> <li>6. Verify bi-directional speech path</li> <li>7. A puts B on HOLD</li> <li>8. Verify B cannot hear A</li> <li>9. A takes B off HOLD</li> <li>10. Verify bi-directional speech path</li> <li>11. A and B disconnect</li> <li>12. Stop the packet trace</li> <li>13. Analyze the packet trace and note in the test case results whether the client implemented HOLD according to RFC 3264 (current), RFC 2543 (deprecated) or whether it just implements MEDIA HOLD ( A opens speech path to B but no evidence of either RFC 3264 or RFC 2543 in the SIP messaging)</li> </ol> |
| Expected Results | Successful outcomes for all verification steps  |
| Test Outcome     | Pass  |
| Issues           |   |
| Execution Notes  | <ul style="list-style-type: none"> <li>• Client initiating HOLD Re-INVITES with SENDONLY media attribute in the SDP</li> <li>• Trace information for this test case was captured by the Yealink tester and has been stored in GENBAND Sharepoint location:<br/> <a href="https://portal.genband.com/sites/Dev/SRL/SV/IOT%20SV/Docs/Projects/C20/Yealink/Results/T-38G%20Client/Traces">https://portal.genband.com/sites/Dev/SRL/SV/IOT%20SV/Docs/Projects/C20/Yealink/Results/T-38G%20Client/Traces</a> </li> </ul>   |

### 8.6.2 SIPIOT018 - Blind Call Transfer

|                       |   |
|-----------------------|---|
| Test Area             | Basic Feature Testing   |
| Test Title            | Blind Call Transfer   |
| Objective             | Verify 3PSL Interoperability with C20-A2 Blind Call Transfer  |
| Configuration & Setup | <p>A: SIP end point under test<br/> B: SIP end point under test<br/> C: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A ,B and C with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> </ol> |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. Place a call from A to B</li> <li>2. Verify B is alerted (ringing).</li> <li>3. Verify A hears audible ringing</li> <li>4. B answers</li> </ol>   |



|                  |   |
|------------------|---|
|                  | <ol style="list-style-type: none"> <li>5. Verify bi-directional speech path</li> <li>6. B invokes call transfer to C</li> <li>7. Verify C is alerted</li> <li>8. B disconnects</li> <li>9. C answers</li> <li>10. Verify bi-directional speech path between A &amp; C</li> <li>11. A &amp; C disconnect to complete the test process</li> </ol> |
| Expected Results | Successful outcomes for all verification steps  |
| Test Outcome     | Pass  |
| Issues           |   |
| Execution Notes  |   |

### 8.6.3 SIPIOT019 – Consultative Call Transfer

|                       |  |
|-----------------------|--|
| Test Area             | Basic Feature Testing  |
| Test Title            | Consultative Call Transfer   |
| Objective             | Verify 3PSL Interoperability with C20-A2 Consultative Call Transfer  |
| Configuration & Setup | <p>A: SIP end point under test<br/> B: SIP end point under test<br/> C: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A, B and C with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> </ol>  |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. A calls B</li> <li>2. Verify B is alerted (ringing)</li> <li>3. A hears audible ring-back</li> <li>4. B answers</li> <li>5. Verify bi-directional speech path</li> <li>6. B invokes call transfer to call C</li> <li>7. Verify A is put on hold</li> <li>8. Verify C is alerted</li> <li>9. C answers</li> <li>10. Verify 2 way speech path between B and C</li> <li>11. B completes the transfer</li> <li>12. Verify B disconnects from the call</li> <li>13. Verify A and C are connected with 2 way speech path</li> <li>14. A and C disconnect to complete the test process</li> </ol> |



|                  |  |
|------------------|--|
| Expected Results | Successful outcomes for all verification steps |
| Test Outcome     | Pass   |
| Issues           |  |
| Execution Notes  |  |

#### 8.6.4 SIPIOT020 - 3 Way Call conference

|                       |  |
|-----------------------|--|
| Test Area             | Basic Feature Testing  |
| Test Title            | 3 Way Call Conference  |
| Objective             | Verify 3PSL Interoperability with C20-A2 3 Way Conference  |
| Configuration & Setup | <p>A: SIP end point under test<br/>           B: SIP end point under test<br/>           C: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A, B and C with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> </ol>  |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. A calls B</li> <li>2. Verify B is alerted (ringing)</li> <li>3. Verify A hears audible ring-back</li> <li>4. B answers</li> <li>5. Verify 2-way speech path</li> <li>6. A invokes conference to C</li> <li>7. B is put on hold</li> <li>8. Verify C is alerted</li> <li>9. Verify A hears audible ringing</li> <li>10. C answers</li> <li>11. Verify bi-directional speech path between A and C</li> <li>12. A joins A, B and C</li> <li>13. Verify bi-directional speech path among A, B and C</li> <li>14. A, B and C disconnect to complete the test process</li> </ol> |
| Expected Results      | Successful outcomes for all verification steps   |
| Test Outcome          | Pass   |
| Issues                |  |



|                 |  |
|-----------------|--|
| Execution Notes |  |
|-----------------|--|

#### 8.6.5 SIPIOT021 - Ad Hoc Conference

|                       |  |
|-----------------------|--|
| Test Area             | Basic Feature Testing  |
| Test Title            | Ad Hoc Conference  |
| Objective             | Verify 3PSL Interoperability with C20-A2 Ad Hoc Conference   |
| Configuration & Setup | <p>NOTE: This test case requires assistance from GENBAND to verify the Media Application Server is controlling the conference</p> <p>A: SIP end point under test<br/>B: SIP end point under test<br/>C: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A, B and C with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> <li>2. Configure the Conference URI of A to be conference. Note that the Request URI in the INVITE sent by A to establish the conference must be as follows: Request-URI: sip:conference@gb.ott7.iot</li> </ol>  |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. A calls B</li> <li>2. Verify B is alerted (ringing)</li> <li>3. Verify A hears audible ring-back</li> <li>4. B answers</li> <li>5. Verify 2-way speech path</li> <li>6. A invokes a second call to C and B is put on hold</li> <li>7. Verify C is alerted</li> <li>8. Verify A hears audible ringing</li> <li>9. C answers</li> <li>10. Verify bi-directional speech path between A and C</li> <li>11. A "joins" all parties together in a conference that is hosted by A2 Media Application Server</li> <li>12. Verify bi-directional speech path among A, B and C</li> <li>13. Verify MAS is controlling the conference</li> <li>14. A, B and C disconnect to complete the test process</li> </ol> |
| Expected Results      | Successful outcomes for all verification steps   |
| Test Outcome          | Pass   |
| Issues                |  |



|                 |  |
|-----------------|--|
| Execution Notes | Trace information for this test case was captured by the Yealink tester and has been stored in GENBAND Sharepoint location:<br><a href="https://portal.genband.com/sites/Dev/SRL/SV/IOT%20SV/Docs/Projects/C20/Yealink/Results/T-38G%20Client/Traces">https://portal.genband.com/sites/Dev/SRL/SV/IOT%20SV/Docs/Projects/C20/Yealink/Results/T-38G%20Client/Traces</a> |
|-----------------|--|

#### 8.6.6 SIPIOT022 - Call Forward no answer

|                       |  |
|-----------------------|--|
| Test Area             | Basic Feature Testing  |
| Test Title            | Call Forward No Answer   |
| Objective             | Verify 3PSL Interoperability with C20-A2 Call Forward No Answer  |
| Configuration & Setup | <p>A: SIP end point under test<br/>B: SIP end point under test<br/>C: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A, B and C with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> <li>2. Activate <a href="#">Call Forward No Answer</a> at B forwarding to C</li> </ol> |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. A calls B</li> <li>2. Verify B is alerted (ringing)</li> <li>3. Verify that the call forwards to C after a few rings and that C is alerted</li> <li>4. C answers the call</li> <li>5. Verify bi-directional speech path between A and C</li> <li>6. A and C disconnect</li> </ol>      |
| Expected Results      | Successful outcomes for all verification steps   |
| Test Outcome          | Pass   |
| Issues                |  |
| Execution Notes       |  |

#### 8.6.7 SIPIOT023 - Call Forward Busy

|            |                       |
|------------|-----------------------|
| Test Area  | Basic Feature Testing |
| Test Title | Call Forward Busy     |



|                       |  |
|-----------------------|--|
| Objective             | Verify 3PSL Interoperability with C20-A2 Call Forward Busy   |
| Configuration & Setup | <p>A: SIP end point under test<br/> B: SIP end point under test<br/> C: SIP end point under test<br/> D: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A, B, C and D with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> <li>2. Activate <a href="#">Call Forward Busy</a> at A forwarding to C</li> </ol>  |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. Place a call from A to D that is dialed using the <a href="#">Cancel Call Waiting Vertical Service Code</a></li> <li>2. D answers and both parties remain off hook in conversation</li> <li>3. Place a call from B to A</li> <li>4. Verify that the call from B forwards to C immediately</li> <li>5. Verify C is alerted</li> <li>6. Answer the call at C</li> <li>7. Verify bi-directional speech path between B and C</li> <li>8. Disconnect A, B, C and D</li> </ol> |
| Expected Results      | Successful outcomes for all verification steps   |
| Test Outcome          | Pass   |
| Issues                |  |
| Execution Notes       |  |

#### 8.6.8 SIPIOT024 - Call Forward Immediate

|                       |  |
|-----------------------|--|
| Test Area             | Basic Feature Testing  |
| Test Title            | Call Forward Unconditional   |
| Objective             | Verify 3PSL Interoperability with C20-A2 Call Forward Unconditional  |
| Configuration & Setup | <p>A: SIP end point under test<br/> B: SIP end point under test<br/> C: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A, B and C with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> <li>2. Activate <a href="#">Call Forward Unconditional</a> at B forwarding to C</li> </ol> |



|                  |   |
|------------------|---|
| Test Procedures  | <ol style="list-style-type: none"> <li>1. Place a call from A to B</li> <li>2. Verify that C is alerted</li> <li>3. C answers the call</li> <li>4. Verify bi-directional speech path between A and C</li> <li>5. A and C disconnect to complete the test process</li> </ol> |
| Expected Results | Successful outcomes for all verification steps  |
| Test Outcome     | Pass  |
| Issues           |   |
| Execution Notes  |   |

#### 8.6.9 SIPIOT025 - Call Waiting

|                       |  |
|-----------------------|--|
| Test Area             | Basic Feature Testing  |
| Test Title            | Call Waiting   |
| Objective             | Verify 3PSL Interoperability with C20-A2 Call Waiting  |
| Configuration & Setup | <p>A: SIP end point under test<br/> B: SIP end point under test provisioned with call waiting<br/> C: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A, B and C with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> </ol>  |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. Place a call from A to B</li> <li>2. Verify B is alerted (ringing).</li> <li>3. B answers</li> <li>4. Verify bi-directional speech path</li> <li>5. C calls B</li> <li>6. B observes Call Waiting Tone (or shown on a display phone )</li> <li>7. B toggles to answer C and place A on hold</li> <li>8. B and C talk</li> <li>9. Verify bi-directional speech path</li> <li>10. B toggles to A and place C on hold</li> <li>11. B and A talk</li> <li>12. Verify bi-directional speech path</li> <li>13. C disconnects</li> <li>14. A &amp; C disconnect to complete the test process</li> </ol> |
| Expected Results      | Successful outcomes for all verification steps   |
| Test Outcome          | Pass   |



|                 |  |
|-----------------|--|
| Issues          |  |
| Execution Notes |  |

#### 8.6.10 SIPIOT026 - Calling Name/Number Display

|                       |   |
|-----------------------|---|
| Test Area             | Basic Feature Testing   |
| Test Title            | Name/Number Display   |
| Objective             | Verify 3PSL Interoperability with C20-A2 Name and Number Display  |
| Configuration & Setup | A: SIP end point under test<br>B: SIP end point under test<br><br>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2   |
| Test Procedures       | 1. Place a call from B to A<br>2. Verify A is alerted (ringing)<br>3. B hears audible ringing<br>4. Verify Caller info of B name and number is observed on A<br>5. A answers<br>6. Verify bi-directional speech path<br>7. A disconnects to complete the test process |
| Expected Results      | Successful outcomes for all verification steps  |
| Test Outcome          | Pass  |
| Issues                |   |
| Execution Notes       |   |

#### 8.6.11 SIPIOT027 - Calling Name/Number Blocked (at Originator)



|                       |   |
|-----------------------|---|
| Test Area             | Basic Feature Testing   |
| Test Title            | Calling Name/Number Blocking  |
| Objective             | Verify 3PSL Interoperability with C20-A2 Calling Name and Number Blocking   |
| Configuration & Setup | <p>A: SIP end point under test<br/>B: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2</li> </ol>  |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. B invokes <a href="#">Calling Name/Number Delivery Blocking</a></li> <li>2. Place a call from B to A</li> <li>3. Verify A is alerted (ringing)</li> <li>4. Verify A hears audible ringing</li> <li>5. Verify that B's name and number are not displayed at A</li> <li>6. A answers</li> <li>7. Verify bi-directional speech path</li> <li>8. A and B disconnect to complete the test process</li> </ol> |
| Expected Results      | Successful outcomes for all verification steps  |
| Test Outcome          | Pass  |
| Issues                |   |
| Execution Notes       |   |

#### 8.6.12 SIPIOT028 – Voice Mail Message waiting indicator

|                       |   |
|-----------------------|---|
| Test Area             | Basic Feature Testing   |
| Test Title            | Voice mail - Message Waiting Indication   |
| Objective             | Verify that SIP DN's with valid Voice Mail accounts, receive Message Waiting Indication from the Voice Mail service   |
| Configuration & Setup | <p>A: SIP end point under test<br/>B: SIP end point under test</p> <ol style="list-style-type: none"> <li>1. Configure A and B with appropriate <a href="#">USERS</a> and REGISTER them in C20-A2; select a <a href="#">Voice Mail</a> capable USER for B</li> <li>2. Activate <a href="#">Call Forward No Answer</a> at B</li> </ol> |
| Test Procedures       | <ol style="list-style-type: none"> <li>1. Place a call from A to B</li> <li>2. B does not answer</li> </ol>   |



|                  |  |
|------------------|--|
|                  | <ol style="list-style-type: none"><li>3. Verify the call forwards to Voice Mail</li><li>4. A leaves a message for B and disconnects</li><li>5. Verify B receives a Message Waiting Indication from the Voice Mail system</li><li>6. B <a href="#">retrieves the message from the VM sytem</a> and disconnects</li><li>7. Verify the Message Waiting Indicator cancels at B</li></ol> |
| Expected Results | Successful outcomes for all verification steps   |
| Test Outcome     | Pass   |
| Issues           |  |
| Execution Notes  |  |



## 8.7 Custom Test cases

The purpose of this section is to test any customer specific features and their interoperability with GENBAND C20-A2. All requested test cases or tests required for this specific interoperability event.

### 8.7.1 SIPIOT9XX - Example Custom Test case

|                       |   |
|-----------------------|---|
| Test Area             | Custom Testing  |
| Test Title            | Example Custom Test case  |
| Objective             | Verify test objective is met  |
| Configuration & Setup | <ul style="list-style-type: none"> <li>• Identify all set-up and configuration</li> <li>• Include any special requirements or provisioning</li> <li>• </li> </ul> |
| Test Procedures       | <ul style="list-style-type: none"> <li>• Include all steps to execute test</li> <li>• Include all verify steps</li> </ul>   |
| Expected Results      | Successful outcomes for all verification steps  |
| Test Outcome          |   |
| Issues                |   |
| Execution Notes       |   |