



GENBAND™

C20-A2 SIP Line Interoperability Test Plan

GENBAND Contact: Anilkumar
Email: anil.yadagiri@genband.com
Phone: +1.972.521.5998
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1 Document History

Customer Issue	Reason	Date	Author
1.0	Initial issue for Yealink IOT	4/18/2013	Anilkumar Yadagiri
1.1	Final report	5/1/2013	Karas Shi
1.2	Revisions based on Internal GENBAND Review	May 16 2013	James Burnie
1.3	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
27	0	1	SIPIOT016: Not Executed because T28P does not support connection to analog FAX device

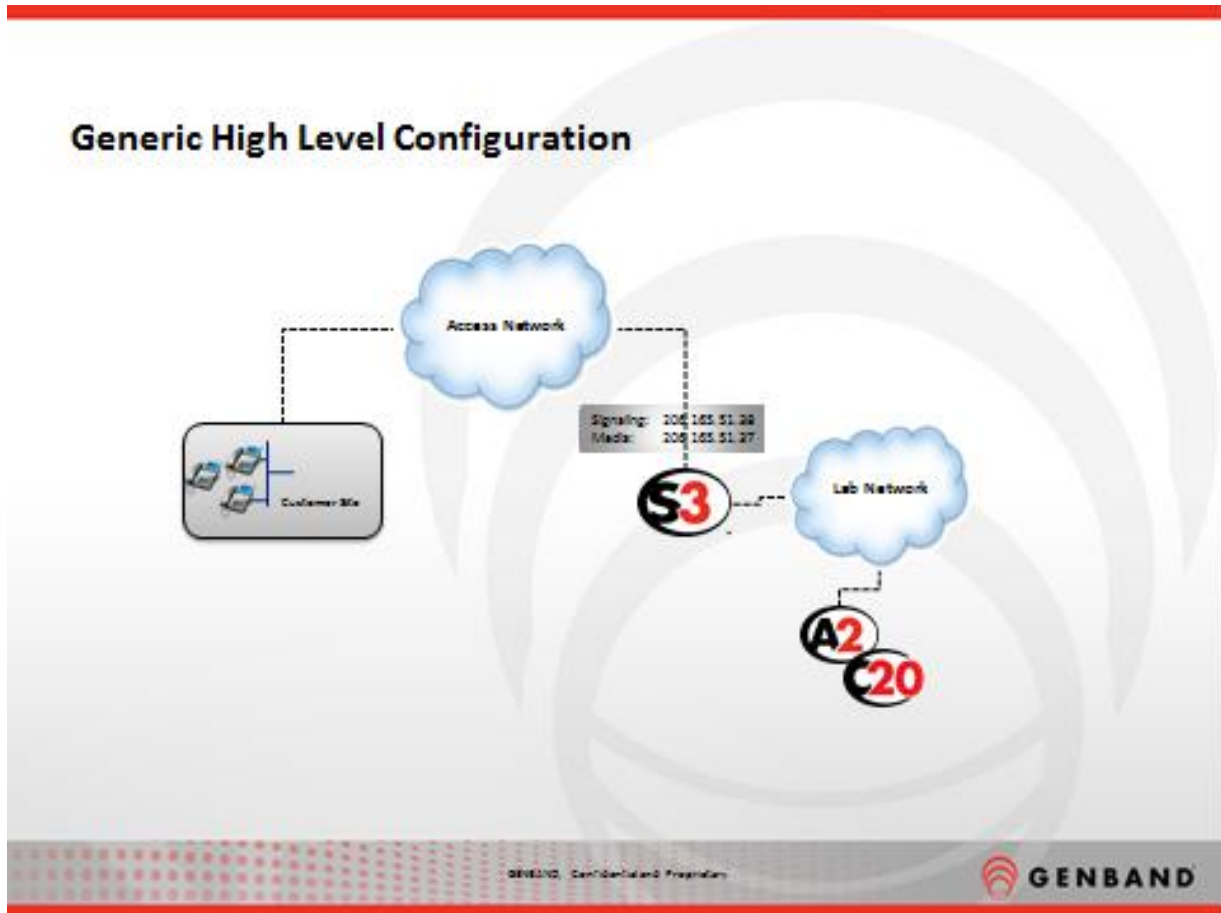


5 Quick Start Guide

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
------	-------



Item	Value
SIP Proxy	206.165.51.38
Protocol	SIP
Port	5060
Domain	gb.ott7.iot
Registration and Authentication	SIP Clients must register and authenticate with the UserID and Password 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



Call Type	Dialing Method
Line to Trunk to Line	Trunk Access Code + 10 digits
Line to Blank Directory Number	9199920000
Features	Vertical Service Code + 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
PVG Loopback	602	SIP clients must support 20ms ptime
SST 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"> Cancel Call Waiting per call *70+Called Digits
CFWP	*72	<ul style="list-style-type: none"> Activate Call Forward Unconditional *72+Digits of number to which calls will be forwarded
CFWC	*73	<ul style="list-style-type: none"> Cancel Call Forward Unconditional *73#
CNDB	*88	<ul style="list-style-type: none"> Calling Number Display Blocking per call Use to block delivery of the originating line’s Number to the called DN *88+Called Digits
CNAB	*90	<ul style="list-style-type: none"> Calling Name Display Blocking per call Use to block delivery of the originating line’s Name to the called DN *90+Called Digits
CNNB	*67	<ul style="list-style-type: none"> Calling Name and Number Blocking per call Use to block delivery of the originating lines Name and Number to the called DN *67+Called Digits
CFBP	*28	<ul style="list-style-type: none"> Activate Call Forward Busy For lines not equipped with VM: *28+Digits of number to which calls will be forwarded For lines equipped with VM: *28#
CFBC	*29	<ul style="list-style-type: none"> Cancel Call Forward Busy *29#
CFDP	*30	<ul style="list-style-type: none"> Activate Call Forward No Answer For lines not equipped with VM: *30+Digits of number to which calls will be forwarded For lines equipped with VM: *30#



Feature	Code	Description
CFDC	*31	<ul style="list-style-type: none"> Cancel Call Forward No Answer *31#

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 T28P

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 T28P 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-28P%20Client>



6.10 Definitions

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
SIPIOT001 – SIP Registration	ZhiGang Cai	SIP-T28P	2.70.0.130	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
SIPIOT002 - 3PSL to 3PSL, Originator disconnects after answer	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT003 - 3PSL to 3PSL, Terminator disconnects after answer	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT004 - 3PSL to 3PSL, Originator disconnects before answer	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT005 - 3PSL to 3PSL, Call to BUSY Line	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT006 - 3PSL Long Duration Call	ZhiGang Cai	SIP-T28P	2.70.0.130	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
SIPIOT007 - 3PSL and SIP trunk, Originator disconnects after answer	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT008 - 3PSL and TDM trunk, Originator disconnects after answer	ZhiGang Cai	SIP-T28P	2.70.0.130	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
SIPIOT009 - 3PSL and SIP trunk, Terminator disconnects after answer	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT010 - 3PSL and TDM trunk, Terminator disconnects after answer	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT011 - 3PSL and SIP trunk Originator disconnects before answer	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT012 - 3PSL and TDM trunk, Originator disconnects before answer	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013

7.4 Announcements

Test Case	Test Prime	Product Under Test	Product SW/FW Version	C20-A2 SW Version	Result Pass/Fail/NE	Execution Date
SIPIOT013 - 3PSL to C20-A2 Announcement	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013

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
SIPIOT014 - 3PSL to 3PSL codec negotiation use G729 and G711 variants with 20ms PR	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT015 - 3PSL to 3PSL (RFC 2833 DTMF)	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT016 – FAX connection using T.38	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	NE	May 3,2013



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
SIPIOT017 – HOLD/RESUME	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT018 - Blind Call Transfer	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT019 – Consultative Call Transfer	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT020 - 3 Way Call conference	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT021 - Ad Hoc Conference	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT022 - Call Forward no answer	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT023 - Call Forward Busy	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT024 - Call Forward Immediate	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT025 - Call Waiting	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT026 - Calling Name/Number Display	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT027 - Calling Name/Number Blocked (at Originator)	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013
SIPIOT028 – Voice Mail Message waiting indicator	ZhiGang Cai	SIP-T28P	2.70.0.130	CVM17	Pass	May 3,2013

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
SIPIOT9XX - Example Custom Test case						



8 Test Case Details and Results

8.1 SIP Registration

This section covers basic registration between the 3rd 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"> 1. Packet trace tool required to capture and decode the IP messaging
Test Procedures	<ol style="list-style-type: none"> 1. Start the trace tool 2. Configure A with an appropriate USER and invoke SIP REGISTRATION; wait 30 seconds 3. Stop the trace tool and analyze the trace 4. Verify A initiates a REGISTER request with a non-zero "expires" value 5. Verify A receives a "200 OK" or "200 Registration Successful" response with a non-zero "expires" value 6. Start the trace tool 7. Invoke de-REGISTRATION at A; wait 30 seconds 8. Stop the trace tool and analyze the trace 9. Verify A initiates a REGISTER request with "expires" = 0 10. Verify A receives a "200 OK" response with "expires" = 0 or a "200 Registration Successful" response
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 B: SIP end point under test 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2
Test Procedures	1. Place a call from A to B 2. Verify B is alerted (ringing) 3. Verify A can hear audible ringing 4. B answers the call from A 5. Verify A and B can talk to each other with bi-directional speech path. 6. A disconnects the call as originator 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 B: SIP end point under test 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2
Test Procedures	1. Place a call from B to A 2. Verify A is alerted (ringing) 3. Verify B can hear audible ringing 4. A answers the call from B 5. Verify B and A can talk to each other with bi-directional speech path 6. A disconnects the call as terminator 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 B: SIP end point under test 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2
Test Procedures	1. Place a call from A to B 2. Verify B is alerted (ringing) 3. Verify A can hear audible ringing 4. B does not answer the call 5. A disconnects 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 B: SIP end point under test C: SIP end point under test</p> <ol style="list-style-type: none"> 1. Configure A, B and C with appropriate USERS and REGISTER them in C20-A2 2. Cancel all Call Forwarding Variants at A using the appropriate Vertical Service Codes
Test Procedures	<ol style="list-style-type: none"> 1. A calls B using the Cancel Call Waiting per call Vertical Service Code 2. B answers the call 3. Verify bi-directional speech path between A and B 4. C calls A 5. Verify C hears busy tone (Since A is in conversation with B) 6. Hang up A, B and C
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 B: SIP end point under test 1. Configure A and B with appropriate USERS and REGISTER in C20-
Test Procedures	1. A calls B 2. Verify B is alerted (ringing) 3. Verify A hears audible ringing 4. B answers the call 5. Verify bi-directional speech path 6. Leave the call up for at least 1 hour. 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 B: SIP end point under test 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2
Test Procedures	1. A calls B using the SST loopback trunk access code 2. Verify B is alerted (ringing), 3. Verify A hears audible ringing 4. B answers the call 5. Verify bi-directional speech path 6. A disconnects the call 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	<p>A: SIP end point under test B: SIP end point under test</p> <ol style="list-style-type: none"> 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2 2. A and B must support 20ms ptime
Test Procedures	<ol style="list-style-type: none"> 1. A calls B using the PVG loopback trunk access code 2. Verify B is alerted (ringing) 3. Verify A hears audible ringing 4. B answers the call 5. Verify bi-directional speech path 6. A disconnects the call 7. Verify A and B can make another call
Expected Results	Successful outcomes for all verification steps
Test Outcome	Pass
Issues	
Execution Notes	

8.3.3 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	<p>A: SIP end point under test B: SIP end point under test</p> <ol style="list-style-type: none"> 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2
Test Procedures	<ol style="list-style-type: none"> 1. A calls B using the SST loopback trunk access code 2. Verify B is alerted (ringing), 3. Verify A hears audible ringing 4. B answers the call 5. Verify bi-directional speech path 6. B disconnects the call 7. Verify A and B can make another call



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

8.3.4 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 B: SIP end point under test 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2 2. A and B must support 20ms ptime
Test Procedures	1. A calls B using the PVG loopback trunk access code 2. Verify B is alerted (ringing). 3. Verify A hears audible ringing 4. B answers the call 5. Verify bi-directional speech path 6. B disconnects the call 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 B: SIP end point under test <ol style="list-style-type: none"> 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2
Test Procedures	<ol style="list-style-type: none"> 1. A calls B using the SST loopback trunk access code 2. Verify B is alerted (ringing). 3. Verify A hears audible ringing 4. A disconnects before B answers the call 5. Verify B stops ringing
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 B: SIP end point under test <ol style="list-style-type: none"> 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2 2. A and B must support 20ms ptime



Test Procedures	<ol style="list-style-type: none"> 1. A calls B using the PVG loopback trunk access code 2. Verify B is alerted (ringing). 3. Verify A hears audible ringing 4. A disconnects before B answers the call 5. Verify B stops ringing
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"> 1. Configure A with an appropriate USER and REGISTER in C20-A2



Test Procedures	<ol style="list-style-type: none"> 1. Place a call from A to a Blank Directory Number 2. Verify that A is sent to a recorded announcement
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 B: SIP end point under test supporting G711A, G711u and G729</p> <ol style="list-style-type: none"> 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2 2. For each test scenario, set up B to use codec G729, G711A, G711u with 20ms packetization rate 3. Packet trace tool required to capture and decode the IP messaging
Test Procedures	<ol style="list-style-type: none"> 1. Start trace tool 2. Place a call from A to B 3. Verify B is alerted (ringing). 4. Verify A hears audible ring-back 5. B answers the call 6. Verify bi-directional speech path 7. A & B disconnect 8. Stop trace tool 9. Analyze traces collected 10. Verify G729 and 20ms packetization rate are used. 11. Repeat step 1->9 with A enforced with G711u 12. Verify G711u and 20ms packetization rate are used 13. Repeat step 1->9 with A enforced with G711A 14. Verify G711A and 20ms packetization rate are used
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 B: SIP end point under test</p> <ol style="list-style-type: none"> 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2 in C20-A2; B's USER must support Voice Mail 2. Activate Call Forward No Answer at B using the appropriate Vertical Service Code



	<ol style="list-style-type: none"> 3. Configure A to use RFC 2833 DTMF 4. Packet trace tool required to capture and decode the IP messaging
Test Procedures	<ol style="list-style-type: none"> 1. Start the trace tool 2. A calls B 3. Allow B to ring until it forwards to Voice Mail 4. Verify A is connected to B's Voice Mail 5. Press # at A during the Voice Mail Greeting 6. Verify the VM Greeting is interrupted 7. Leave a message and press # 8. Verify VM presents menu of options 9. Press # at A 10. Verify session terminates 11. Stop trace tool 12. Analyze contents of trace file 13. Verify that the 3 #'s are present as RFC 2833 events in the RTP stream
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 B: SIP end point under test</p> <ol style="list-style-type: none"> 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2 2. Configure T.38 and G.729 on both A and B 3. Connect FAX machines to A and B. 4. Packet trace tool required to capture and decode the IP messaging



Test Procedures	<ol style="list-style-type: none"> 1. Start the trace tool 2. A calls B 3. Verify B is alerted (ringing). 4. B answers 5. A and B negotiate fax connection 6. Verify B successfully receives all pages of the fax from A 7. A & B disconnect after fax-sending is complete 8. Stop the trace tool 9. Analyze the trace captured 10. Verify the call is initially set up using G.729 and then successful negotiation of T.38 for the fax transmission
Expected Results	Successful outcomes for all verification steps
Test Outcome	Not Executed
Issues	
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 B: SIP end point under test</p> <ol style="list-style-type: none"> 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2 2. Packet trace tool required to capture and decode the IP messaging
Test Procedures	<ol style="list-style-type: none"> 1. Start the packet trace tool 2. A calls B 3. Verify B is alerted (ringing) 4. Verify A hears audible ringing 5. B answers



	6. Verify bi-directional speech path 7. A puts B on HOLD 8. Verify B cannot hear A 9. A takes B off HOLD 10. Verify bi-directional speech path 11. A and B disconnect 12. Stop the packet trace 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)
Expected Results	Successful outcomes for all verification steps
Test Outcome	Pass
Issues	
Execution Notes	<ul style="list-style-type: none"> Client initiating HOLD Re-INVITES with SENDONLY media attribute and c=0.0.0.0 in the SDP Trace information for this test case was captured by the Yealink tester and has been stored in GENBAND Sharepoint location: https://portal.genband.com/sites/Dev/SRL/SV/IOT%20SV/Docs/Projects/C20/Yealink/Results/T-28P%20Client/Traces

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	A: SIP end point under test B: SIP end point under test C: SIP end point under test 1. Configure A ,B and C with appropriate USERS and REGISTER them in C20-A2
Test Procedures	1. Place a call from A to B 2. Verify B is alerted (ringing). 3. Verify A hears audible ringing 4. B answers 5. Verify bi-directional speech path



	6. B invokes call transfer to C 7. Verify C is alerted 8. B disconnects 9. C answers 10. Verify bi-directional speech path between A & C 11. A & C disconnect to complete the test process
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	A: SIP end point under test B: SIP end point under test C: SIP end point under test 1. Configure A, B and C with appropriate USERS and REGISTER them in C20-A2
Test Procedures	1. A calls B 2. Verify B is alerted (ringing) 3. A hears audible ring-back 4. B answers 5. Verify bi-directional speech path 6. B invokes call transfer to call C 7. Verify A is put on hold 8. Verify C is alerted 9. C answers 10. Verify 2 way speech path between B and C 11. B completes the transfer 12. Verify B disconnects from the call 13. Verify A and C are connected with 2 way speech path 14. A and C disconnect to complete the test process



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



Execution Notes	
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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 B: SIP end point under test C: SIP end point under test</p> <ol style="list-style-type: none"> 1. Configure A, B and C with appropriate USERS and REGISTER them in C20-A2 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
Test Procedures	<ol style="list-style-type: none"> 1. A calls B 2. Verify B is alerted (ringing) 3. Verify A hears audible ring-back 4. B answers 5. Verify 2-way speech path 6. A invokes a second call to C and B is put on hold 7. Verify C is alerted 8. Verify A hears audible ringing 9. C answers 10. Verify bi-directional speech path between A and C 11. A "joins" all parties together in a conference that is hosted by A2 Media Application Server 12. Verify bi-directional speech path among A, B and C 13. Verify MAS is controlling the conference 14. A, B and C disconnect to complete the test process
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: https://portal.genband.com/sites/Dev/SRL/SV/IOT%20SV/Docs/Projects/C20/Yealink/Results/T-28P%20Client/Traces
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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 B: SIP end point under test C: SIP end point under test</p> <ol style="list-style-type: none"> 1. Configure A, B and C with appropriate USERS and REGISTER them in C20-A2 2. Activate Call Forward No Answer at B forwarding to C
Test Procedures	<ol style="list-style-type: none"> 1. A calls B 2. Verify B is alerted (ringing) 3. Verify that the call forwards to C after a few rings and that C is alerted 4. C answers the call 5. Verify bi-directional speech path between A and C 6. A and C disconnect
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 B: SIP end point under test C: SIP end point under test D: SIP end point under test</p> <ol style="list-style-type: none"> 1. Configure A, B, C and D with appropriate USERS and REGISTER them in C20-A2 2. Activate Call Forward Busy at A forwarding to C
Test Procedures	<ol style="list-style-type: none"> 1. Place a call from A to D that is dialed using the Cancel Call Waiting Vertical Service Code 2. D answers and both parties remain off hook in conversation 3. Place a call from B to A 4. Verify that the call from B forwards to C immediately 5. Verify C is alerted 6. Answer the call at C 7. Verify bi-directional speech path between B and C 8. Disconnect A, B, C and D
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 B: SIP end point under test C: SIP end point under test</p> <ol style="list-style-type: none"> 1. Configure A, B and C with appropriate USERS and REGISTER them in C20-A2 2. Activate Call Forward Unconditional at B forwarding to C



Test Procedures	<ol style="list-style-type: none"> 1. Place a call from A to B 2. Verify that C is alerted 3. C answers the call 4. Verify bi-directional speech path between A and C 5. A and C disconnect to complete the test process
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 B: SIP end point under test provisioned with call waiting C: SIP end point under test</p> <ol style="list-style-type: none"> 1. Configure A, B and C with appropriate USERS and REGISTER them in C20-A2
Test Procedures	<ol style="list-style-type: none"> 1. Place a call from A to B 2. Verify B is alerted (ringing). 3. B answers 4. Verify bi-directional speech path 5. C calls B 6. B observes Call Waiting Tone (or shown on a display phone) 7. B toggles to answer C and place A on hold 8. B and C talk 9. Verify bi-directional speech path 10. B toggles to A and place C on hold 11. B and A talk 12. Verify bi-directional speech path 13. C disconnects 14. A & C disconnect to complete the test process
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 B: SIP end point under test 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2
Test Procedures	1. Place a call from B to A 2. Verify A is alerted (ringing) 3. B hears audible ringing 4. Verify Caller info of B name and number is observed on A 5. A answers 6. Verify bi-directional speech path 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	A: SIP end point under test B: SIP end point under test 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2
Test Procedures	1. B invokes Calling Name/Number Delivery Blocking 2. Place a call from B to A 3. Verify A is alerted (ringing) 4. Verify A hears audible ringing 5. Verify that B's name and number are not displayed at A 6. A answers 7. Verify bi-directional speech path 8. A and B disconnect to complete the test process
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	A: SIP end point under test B: SIP end point under test 1. Configure A and B with appropriate USERS and REGISTER them in C20-A2; select a Voice Mail capable USER for B 2. Activate Call Forward No Answer at B
Test Procedures	1. Place a call from A to B 2. B does not answer



	<ol style="list-style-type: none">3. Verify the call forwards to Voice Mail4. A leaves a message for B and disconnects5. Verify B receives a Message Waiting Indication from the Voice Mail system6. B retrieves the message from the VM sytem and disconnects7. Verify the Message Waiting Indicator cancels at B
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"> • Identify all set-up and configuration • Include any special requirements or provisioning •
Test Procedures	<ul style="list-style-type: none"> • Include all steps to execute test • Include all verify steps
Expected Results	Successful outcomes for all verification steps
Test Outcome	
Issues	
Execution Notes	