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Standalone A2 Basic IOT Test Plan for SIP Clients

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DRIVING THE NETWORK EVOLUTION

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Document Handling Notice

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Document History

| Date | Base Version | Issue | GENBAND Prime | Remarks |
|-----------|--------------|-------|---------------|---------------|
| 1/24/2013 | 2.9 | 1.0 | Anilkumar | Initial Issue |
| 4/24/2013 | 2.9 | 1.1 | Anilkumar | Final report |
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Document Navigation and Recording Results

- This document makes extensive use of links and link buttons for navigation and is best viewed using electronic media
- The Contents page provides links to all key subject matter in this document
- The Test Case Summary pages provide additional links to individual test case descriptions and status/results
- Link buttons appear at the RH bottom corner of most pages and have the following functions
- 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 and Test Results, for each test case using a PDF editor. GENBAND expects each IOT customer to return a copy of this document containing sufficient result information to enable preparation of a meaningful IOT Activity Report
- GENBAND will complete the IOT Activity Results Summary section once results have been returned



Return to most relevant higher level of links in context



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Scope

- This test plan defines a service focused set of test cases to verify basic interoperability between SIP Clients and GENBAND's Standalone A2 technology. It is intended to be used in conjunction with a customer-specific IOT information package
- All calls are "Intra A2", meaning they are switched by A2 between SIP Clients registered within the same A2 server
- GENBAND Media Application Server (MAS) technology is used to support Conferencing, Announcements and Voice Mail functions
- The majority of test cases are oriented towards the SIP Client vendor/user and generally should not require an A2 provisioning client or System Manager for execution or result assessment beyond the initial IOT lab setup implemented by GENBAND. However, in some instances, testers may need, and should request, GENBAND assistance to configure the particular test scenario
- Client level service activation and control is achieved via SIP Client functionality and/or A2 Vertical Service Codes
- Client DTMF capability is examined in the context of successful interaction with Voice Mail, MeetMe Conference and Last Call Return test cases
- The Client's ability to return a BUSY condition is examined in the context of the Call Forward Busy variant tests



IOT Activity Results Summary

- Total count of tested cases: 23
- Passed: 22
- Failed: 0
- Not executed: 1
- Issues with case A2IOT014 - Receive Identity of Last Client that Called and Return Call:
 - Yealink comments: We can hear the response from server, but when we input the digit, nothing happened. We found that when the phone sends INVITE *91 to server, A2 response with SIP 183 and in that package it says it's "sendonly" which means it don't receive RTP packets. So the phone won't send DTMF back to the server.



Record of Execution

| Test Case | Test Prime | Product Under Test (Model and Version) | Product SW/FW Release | A2 Release | Execution Complete Date |
|-----------|------------|--|-----------------------|------------|-------------------------|
| A2IOT001 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT002 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT003 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT004 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT005 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT006 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT007 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT008 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT009 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |



Record of Execution

| Test Case | Test Prime | Product Under Test (Model and Version) | Product SW/FW Release | A2 Release | Execution Complete Date |
|-----------|------------|--|-----------------------|------------|-------------------------|
| A2IOT010 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT011 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT012 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT013 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT014 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT015 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT016 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT017 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT018 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |



Record of Execution

| Test Case | Test Prime | Product Under Test (Model and Version) | Product SW/FW Release | A2 Release | Execution Complete Date |
|-----------|------------|--|-----------------------|------------|-------------------------|
| A2IOT019 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT020 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT021 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT022 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT023 | Karas Shi | SIP-VP530 | 23.70.0.229 | 10.1 | Apr 19,2013 |
| A2IOT024 | | | | | |
| A2IOT025 | | | | | |



Standalone A2 Lab Configuration for Basic SIP Client Interoperability tests

Characteristics

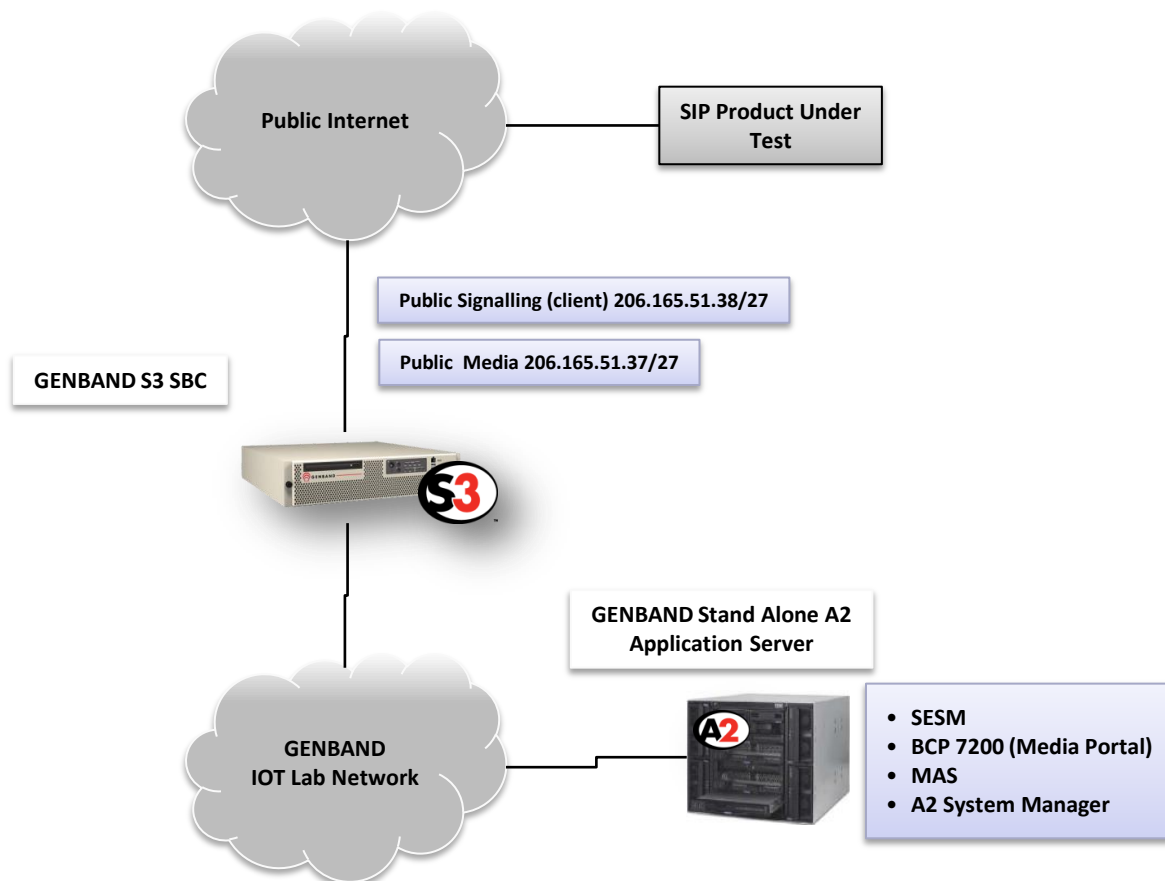
- All switched connections are managed by the Standalone A2 Application Server
- SIP Clients under test are registered in same A2 domain

Available Services

- Ad Hoc Conferencing
- MeetMe Conferencing
- Anonymous Call Rejection
- Call Forwarding
- Network Call Waiting
- Calling Line Identification Restriction
- Do Not Disturb
- Last Call Return
- Voice Mail

Configuration Details

- [Configuration Information for Product Under Test](#)
- [A2 SIP Profile](#)
- [A2 Authorized Methods – SIP](#)
- [A2 Registrar](#)



Configuring the VP530 Phone

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: Enable

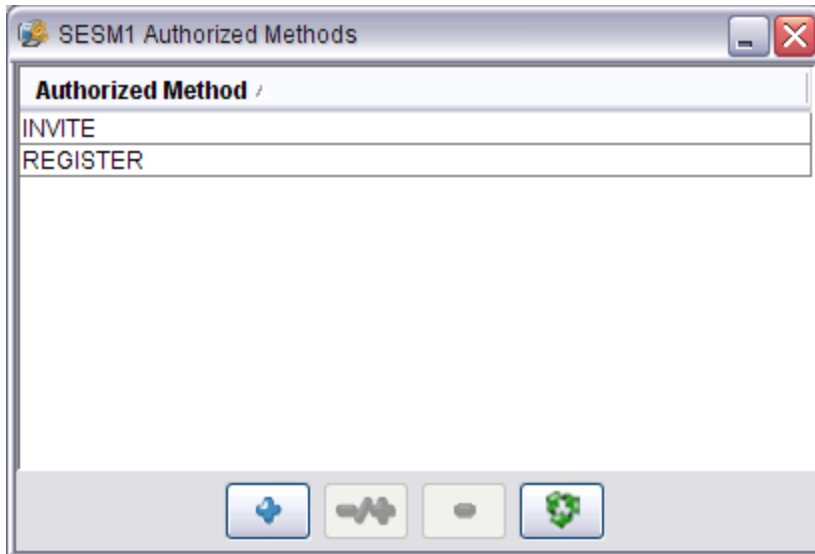


A2 SIP Profile

- SIP Profiles are used to facilitate interoperability between various SIP clients and the GENBAND A2
- The SIP Profile name is derived from the User Agent information contained in the client's SIP invite
- GENBAND will create a unique SIP Profile for each vendor's client as part of the initial provisioning for a given IOT activity based on information supplied by the IOT customer
- The final SIP Profile used for this IOT activity is stored in GENBAND's IOT document repository at:
<https://portal.genband.com/sites/Dev/SRL/SV/IOT%20SV/Docs/Projects/A2-SA/Yealink/Results/VP530%20Client>



A2 Authorized Methods - SIP



- The A2 IOT lab is configured to require authentication for SIP REGISTER and INVITE transactions



SIP Registration

- SIP Clients will be required to supply authentication credentials (User-id and password) provided by GENBAND to the A2 REGISTRAR following an initial SIP invite for registration purposes
- The GENBAND S3 Session Border Controller will proxy the initial registration request and subsequent authentication information to the A2 REGISTRAR and then manage following re-registration requests, based on the registration expiry interval dictated by the A2 REGISTRAR, on behalf of each SIP client.
- Independent of this process, the S3 SBC will request each SIP client to periodically re-register following successful initial registration at a rate determined by the SBC (approximately 240 seconds per current IOT lab setup)
- Authentication requests will be conveyed to the SIP clients in “407 authentication required” messages
- Registration success indications will be returned to the SIP clients from the SBC as “200 OK” messages



Vertical Service Code Summary

| Functionality | Vertical Service Code |
|---|-----------------------|
| Ad_Hoc_Conferencing_Disable | *61 |
| Ad_Hoc_Conferencing_Enable | *60 |
| Anonymous_Call_Rejection_Disable | *83 |
| Anonymous_Call_Rejection_Enable | *82 |
| Call_Forward_All_Disable | *71 |
| Call_Forward_Conditional_Disable | *75 |
| Call_Forward_Conditional_Enable | *74 |
| Call_Forward_Immediate_Disable | *73 |
| Call_Forward_Immediate_Enable | *72 |
| Call_Forward_To_Voicemail_Disable | *77 |
| Call_Forward_To_Voicemail_Enable | *76 |
| Call_Forward_Variants_Busy_Disable | *94 |
| Call_Forward_Variants_Busy_Enable | *93 |
| Call_Forward_Variants_No_Answer_Disable | *96 |
| Call_Forward_Variants_No_Answer_Enable | *95 |
| Call_Return_CLI_Erase | *90 |
| Call_Return_CLI_Notification | *91 |
| Call_Return_Disable | *89 |
| Call_Return_Enable | *88 |
| Call_Return_Immediate | *92 |
| Calling_Line_ID_Restriction_Disable | *81 |



Vertical Service Code Summary

| Functionality | Vertical Service Code |
|---------------------------------------|-----------------------|
| Calling_Line_ID_Restriction_Enable | *80 |
| Calling_Line_ID_Restriction_per_Call | *79 |
| Do_Not_Disturb_Disable | *87 |
| Do_Not_Disturb_Enable | *86 |
| Network_Call_Waiting_Disable | *65 |
| Network_Call_Waiting_Disable_per_Call | *66 |
| Network_Call_Waiting_Enable | *64 |
| Number_Of_Rings | *97 |
| Voicemail_Retrieve | *78 |



Telephony Treatments

- For simplicity, all available telephony treatments in the standalone A2 lab have been divided into two groups:
 - Treatments for Starcode (Vertical Service Code) outcomes
 - Treatments for other telephony outcomes
- All “non-routed” Vertical Service Code outcomes will receive a confirmation tone (three short beeps); routed VSC outcomes should reach the intended destination
- All other treated telephony outcomes, including misdialed Vertical Service Codes, will receive an announcement stating that the number cannot be reached as dialed followed by dial tone



Test Case Summary



Test Case Summary

| TCID | Subject | Test Name | Status |
|----------|--------------------------|--|----------------------|
| A2IOT001 | Registration | SIP Client Registration to A2 System in Specific Domain | Pass |
| A2IOT002 | Simple Calls | Basic Call between two A2 Registered SIP Clients in Same Domain using Alias Routing | Pass |
| A2IOT003 | Announcements | SIP Client Routes to Announcement for Defined Treatment Reason | Pass |
| A2IOT004 | Call Forward Conditional | Call Forwarding for Busy and No Answer Conditions | Pass |
| A2IOT005 | Call Forward Immediate | Call Forward Immediate – All Incoming Calls to A2 Registered SIP Client Forward to Specified Destination | Pass |
| A2IOT006 | Call Mute | Call Mute | Pass |
| A2IOT007 | Call Waiting | Call Waiting for A2 Registered SIP Clients | Pass |
| A2IOT008 | Call Waiting | A2 Registered SIP Client Disables Call Waiting Service on Per Call Basis | Pass |
| A2IOT009 | Calling Line ID Blocking | A2 Registered SIP Client uses Calling Line ID Restriction Service to Block Display Information to other A2 Clients | Pass |



Test Case Summary

| TCID | Subject | Test Name | Status |
|----------|--------------------------|--|--------------------------------|
| A2IOT010 | Anonymous Call Rejection | A2 Registered SIP Client Uses Anonymous Call Rejection to Block Incoming Calls Having No Calling Line Identification | Pass |
| A2IOT011 | Conference | AdHoc Conference - Client Initiated Conference for Multiple A2 Registered SIP Clients | Pass |
| A2IOT012 | Conference | MeetMe Conference - Multiple A2 Registered SIP Clients Join A2 MeetMe Conference | Pass |
| A2IOT013 | Do Not Disturb | Calls to A2 Registered SIP Client with Do Not Disturb Service Active | Pass |
| A2IOT014 | Last Call Return | Receive Identity of Last Client that Called and Return Call | Not Executable |
| A2IOT015 | Last Call Return | Return Call to Last Client that Called without First Learning Client Identity | Pass |
| A2IOT016 | Music on Hold | A2 Music on Hold Service Provides Audio to SIP Client on Hold | Pass |
| A2IOT017 | Voice Mail | SIP Client Terminates on A2 Voice Mail and Leaves Message | Pass |
| A2IOT018 | Voice Mail | Message Waiting Indication Provided to SIP Client as part of A2 Voice Mail Service | Pass |



Test Case Summary

| TCID | Subject | Test Name | Status |
|----------|------------------|---|-------------------------|
| A2IOT019 | Voice Mail | SIP Client Retrieves Message from A2 Voice Mail Service | Pass |
| A2IOT020 | Client Stability | IDLE Client SIP Re-registration | Pass |
| A2IOT021 | Client Stability | Long Duration Call with Re-registration | Pass |
| A2IOT022 | CODEC Selection | CODEC Support – Default Client Configuration | Pass |
| A2IOT023 | CODEC Selection | CODEC Support – All Mutually Supported CODECS | Pass |
| A2IOT024 | CODEC Selection | CODEC Support – Mid Call CODEC Change | Not Run |
| A2IOT025 | T.38 Facsimile | T.38 FAX Call Between 2 FAX Capable SIP Clients | Not Run |



Test Case Details



A2IOT001 - SIP Client Registration to A2 System in Specific Domain

Objective

- Verify that target SIP clients are able to register within the Standalone A2 SIP Registrar

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail

Procedure

1. Select a valid user for A within the target A2 server and domain
2. Select a valid user for B within the same A2 server and domain as A
3. Configure A and B appropriately such that they are able to register in the target A2 server, domain and location with the chosen SIP users
4. Verify both A and B register successfully according to the context of their particular User Interface

Expected Results

1. Successful outcomes for all verification steps



A2IOT002 - Basic Call between two A2 Registered SIP Clients in Same Domain using Alias Routing

Objective

- Verify successful calls between target SIP clients registered in the same A2 server and domain

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail

Procedure

1. Register A and B in the same A2 server, domain and location
2. Place call from A to B
3. Verify B is alerted of call from A
4. Answer Call at B
5. Verify two way media between A and B
6. A and B terminate the call
7. Verify A and B disconnect successfully according to their respective user interfaces
8. Place a call from B to A
9. Verify A is alerted of call from B
10. Answer call at A
11. Verify two way media between B and A
12. A and B terminate the call
13. Verify B and A disconnect according to their respective user interfaces

Expected Results

1. Successful outcomes for all verification steps



A2IOT003 - SIP Client Routes to Announcement for Defined Treatment Reason

Objective

- Verify that the target SIP Client is able to connect to MAS announcements for a specified treatment condition (s) as defined within the Client's domain

Configuration and Setup

A: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. All SIP Endpoints are registered in the same domain and location within the same A2 server

Procedure

1. A calls an invalid alias or userid
2. Verify that A reaches an announcement according to the treatment mappings established for A's domain

Expected Results

1. Successful outcomes for all verification steps



A2IOT004 - Call Forwarding for Busy and No Answer Conditions

Objective

- Verify that calls to an A2 registered SIP Client will forward according to the SIP Client's user's Call Forwarding Conditional Service

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

C: SIP Endpoint under test

D: SIP Endpoint under test

1. Unless specified otherwise, the term "SIP Endpoint under test" shall refer to the Product under Test as defined in the [Record of Execution](#).
2. SIP Endpoints under test having native functionality equivalent or similar to the A2 service/functionality being tested must have that native functionality DISABLED in order to avoid conflicts while testing the A2 service. If the vendor opts to execute the recommended optional additional test coverage, then the native functionality must be re-enabled for that portion of the testing.
3. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
4. All SIP Endpoints are registered in the same domain and location within the same A2 server
5. Call Forward Conditional service is assigned to Client A's user
6. Vertical Service Codes are assigned within the user's domain to activate and deactivate Call Forward Conditional

Procedure

1. Activate Call Forward Conditional at A towards C using the appropriate Vertical Service Code
2. Place a call from B to A; A does not answer
3. Verify that the call from B forwards to C after a number of ring cycles
4. Answer the Call from Client B at Client C
5. Verify 2 way media between B and C
6. Hang up the call at both B and C
7. Create a condition at A such that calls to it return a "486 BUSY HERE" response. Note that this step usually requires A to be off hook with Call Waiting disabled and will require use of a packet trace tool, such as Wireshark, to confirm the "486 BUSY HERE" condition. If necessary, contact GENBAND for further guidance on this step.
8. Place a call from B to A
9. Verify that the call from B forwards to C immediately
10. Answer the call at C
11. Verify 2 way media between B and C

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A2IOT004 - Call Forwarding for Busy and No Answer Conditions

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Expected Results

1. Successful outcomes for all verification steps

Recommended Optional Additional Test Coverage

If the product under test has native functionality equivalent to the A2 service/feature targeted in this test case, the tester should re-execute applicable portions of this test case using that functionality instead of the A2 service/feature. Note that in some cases, it may be necessary to disable the A2 service/feature via an appropriate Vertical Service Code.. GENBAND will assist with any instances where a change in A2 provisioning is necessary to accomplish this.



A2IOT005 - Call Forward Immediate – All Incoming Calls to A2 Registered SIP Client Forward to Specified Destination

Objective

- Verify that calls to target SIP Clients registered in A2 will forward according to the SIP Client's user's Call Forward Immediate service

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

C: SIP Endpoint under test

1. Unless specified otherwise, the term "SIP Endpoint under test" shall refer to the Product under Test as defined in the [Record of Execution](#).
2. SIP Endpoints under test having native functionality equivalent or similar to the A2 service/functionality being tested must have that native functionality DISABLED in order to avoid conflicts while testing the A2 service. If the vendor opts to execute the recommended optional additional test coverage, then the native functionality must be re-enabled for that portion of the testing.
3. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
4. All SIP Endpoints are registered in the same domain and location within the same A2 server
5. Call Forward Immediate is assigned to Client A's user
6. Vertical Service Codes are assigned within the user's domain to activate and deactivate Call Forward Immediate

Procedure

1. Disable all Call Forward features for A using the "Call Forward All Disable" Vertical Service code
2. Activate Call Forward Immediate at A using the "Call Forward Immediate Vertical Service Code"; specify C as the "forwarded to" destination
3. Place a call from B to A
4. Verify that the call from B forwards immediately to C
5. Answer the Call from B at C
6. Verify 2 way media between B and C
7. Hang up the call at both B and C
8. Verify all clients disconnect

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A2IOT005 - Call Forward Immediate – All Incoming Calls to A2 Registered SIP Client Forward to Specified Destination

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Expected [Results](#)

1. Successful outcomes for all verification steps

Recommended Optional Test Coverage

If the product under test has native functionality equivalent to the A2 service/feature targeted in this test case, the tester should re-execute applicable portions of this test case using that functionality instead of the A2 service/feature. Note that in some cases, it may be necessary to disable the A2 service/feature via the appropriate Vertical Service Code prior to doing so. GENBAND will assist with any instances where a change in A2 provisioning is necessary.



A2IOT006 - Call Mute

Objective

- Verify that target SIP Clients registered in A2 having a MUTE feature are able to MUTE calls

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. All SIP Endpoints are registered in the same domain and location within the same A2 server

Procedure

1. A calls B; B answers
2. Verify 2 way media between A and B
3. B activates MUTE feature
4. Verify that A receives no media from B
5. Verify that B receives media from A
6. B deactivates MUTE
7. Verify 2 way media between A and B
8. A and B both activate their MUTE feature
9. Verify that A receives no media from B
10. Verify that B receives no media from A
11. A and B deactivate their MUTE feature
12. Verify 2 way media between A and B

Expected Results

1. Successful outcomes for all verification steps



A2IOT007 - Call Waiting for A2 Registered SIP Clients

Objective

- Verify A2 Call Waiting functions for a target SIP Clients registered in A2

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

C: SIP Endpoint under test

1. Unless specified otherwise, the term "SIP Endpoint under test" shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. All SIP Endpoints are registered in the same domain and location within the same A2 server

Procedure

1. Disable all Call Forward features for A using the "Call Forward All Disable" Vertical Service code
2. Enable Network Call Waiting for A at the A2 using the "Network Call Waiting Enable" Vertical Service Code
3. Enable Call Waiting (if applicable) locally within A
4. Establish and maintain a call between A and B
5. C calls A
6. Verify A receives notification of the incoming call, including caller identity if A is equipped with a display
7. A answers the call from C
8. Verify B goes on HOLD while A and C are connected
9. A and C end their call and B remains off hook
10. Verify that A receives ringing (notification that B is still on HOLD)
11. A answers the incoming call
12. Verify A and B resume their call with 2 way media
13. A and B disconnect

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A2IOT007 - Call Waiting for A2 Registered SIP Clients

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Expected Results

1. Successful outcomes for all verification steps



A2IOT008 - A2 Registered SIP Client Disables Call Waiting Service on Per Call Basis

Objective

- Verify that A2 Call Waiting can be disabled for target SIP Clients on a per call basis

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

C: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. All SIP Endpoints are registered in the same domain and location within the same A2 server
4. A is configured to route BUSY calls to voicemail

Procedure

1. Enable Network Call Waiting for A at the A2 using the “Network Call Waiting Enable” Vertical Service Code
2. A calls B using this format: *+ Vertical Service Code to Disable Network Call Waiting per call + SIP URI or alias for B
Example, if the alias for B is 5972002 and the Network Call Waiting Disable Per Call Vertical Service Code is 66, then A would dial *665972002 as one continuous string
3. A hears confirmation tone and is then routed to B
4. B answers the call; A and B engage in conversation
5. C calls A
6. Verify A is NOT notified of the call from C
7. Verify C reaches A's Voice Mail
8. A and B disconnect
9. A calls B without dialing the Network Call Waiting Per Call Disable Vertical Service Code (e.g.. 5972002)
10. B answers the call; A and B engage in conversation
11. C calls A

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A2IOT008 - A2 Registered SIP Client Disables Call Waiting Service on Per Call Basis

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13. Verify A receives notification of the incoming call, including caller identity if A is equipped with a display
14. A answers the call from C
15. Verify B goes on HOLD while A and C are connected
16. A and C end their call and B remains off hook
17. Verify B remains on HOLD to A
18. A RESUMEs call with B
19. Verify 2 way media between A and B
20. A and B disconnect

Expected Results

1. Successful outcomes for all verification steps



A2IOT009 - A2 Registered SIP Client uses Calling Line ID Restriction Service to Block Display Information to other A2 Clients

Objective

- Verify that Calling Line ID information for the target SIP Client is NOT presented when Calling Line ID Blocking is active

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. SIP Endpoints under test having native functionality equivalent or similar to the A2 service/functionality being tested must have that native functionality DISABLED in order to avoid conflicts while testing the A2 service. If the vendor opts to execute the recommended optional additional test coverage, then the native functionality must be re-enabled for that portion of the testing.
3. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
4. All SIP Endpoints are registered in the same domain and location within the same A2 server
5. All SIP Endpoints have the ability to display Calling Line Identity Information
6. Vertical Service Codes to ENABLE and DISABLE Calling Line ID Restriction are assigned and available to A

Procedure

1. A calls B
2. Verify B receives Calling Line Identity information for A
3. A terminates the call
4. A activates Calling Line Id Restriction via the “Calling Line ID Restriction Enable” Vertical Service Code
5. A calls B
6. Verify B does NOT receive any Calling Line Identity information for A
7. A terminates the call

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A2IOT009 - A2 Registered SIP Client uses Calling Line ID Restriction Service to Block Display Information to other A2 Clients

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Expected Results

- Successful outcomes for all verification steps

Recommended Optional Test Coverage

If the product under test has native functionality equivalent to the A2 service/feature targeted in this test case, the tester should re-execute applicable portions of this test case using that functionality instead of the A2 service/feature. Note that in some cases, it may be necessary to disable the A2 service/feature via the appropriate Vertical Service Code prior to doing so. GENBAND will assist with any instances where a change in A2 provisioning is necessary.



A2IOT010 - A2 Registered SIP Client Uses Anonymous Call Rejection to Block Incoming Calls Having No Calling Line Identification

Objective

- Verify that anonymous calls to target SIP Clients having Anonymous Call Rejection ENABLED do NOT complete to those clients

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. SIP Endpoints under test having native functionality equivalent or similar to the A2 service/functionality being tested must have that native functionality DISABLED in order to avoid conflicts while testing the A2 service. If the vendor opts to execute the recommended optional additional test coverage, then the native functionality must be re-enabled for that portion of the testing.
3. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
4. All SIP Endpoints are registered in the same domain and location within the same A2 server
5. All SIP Endpoints have the ability to display Calling Line Identity Information

Procedure

1. Disable Anonymous Call Rejection at A using the “Anonymous Call Rejection Disable” Vertical Service Code
2. Enable Calling Line ID Restriction at B using the “Calling Line ID Restriction Enable” Vertical Service Code
3. B calls A
4. Verify that A is alerted of an incoming “anonymous” call
5. A answers; B and A engage in conversation
6. A and B disconnect
7. A turns Anonymous Call Rejection ON using the “Anonymous Call Rejection Enable” Vertical Service Code
8. B calls A
9. Verify the call does NOT reach A

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A2IOT010 - A2 Registered SIP Client Uses Anonymous Call Rejection to Block Incoming Calls Having No Calling Line Identification

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Expected Results

1. Successful outcomes for all verification steps

Recommended Optional Test Coverage

If the product under test has native functionality equivalent to the A2 service/feature targeted in this test case, the tester should re-execute applicable portions of this test case using that functionality instead of the A2 service/feature. Note that in some cases, it may be necessary to disable the A2 service/feature via the appropriate Vertical Service Code prior to doing so. GENBAND will assist with any instances where a change in A2 provisioning is necessary.



A2IOT011 - AdHoc Conference - Client Initiated MAS based Conference for Multiple A2 Registered SIP Clients

Objective

- Verify the ability of target SIP Clients to successfully establish a 3 way conference call using the A2 AdHoc Conference feature in conjunction with the A2 MAS

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

C: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. SIP Endpoints under test having native functionality equivalent or similar to the A2 service/functionality being tested must have that native functionality DISABLED in order to avoid conflicts while testing the A2 service. If the vendor opts to execute the recommended optional additional test coverage, then the native functionality must be re-enabled for that portion of the testing.
3. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
4. All SIP Endpoints are registered in the same domain and location within the same A2 server

Procedure

1. Enable “server based” conferencing within A
2. A calls B
3. B answers; A and B remain off hook
4. Verify 2 way media between A and B
5. A initiates a second call to C
6. Verify C is alerted and B is put on HOLD
7. Answer call at C; A and C remain off hook
8. Verify 2 way media between A and C
9. A “joins” all parties together
10. Verify 2 way media between all 3 parties
11. Verify that the A2 MAS is hosting the conference – contact the GENBAND IOT Prime for assistance with this step
12. A, B and C terminate the call
13. Verify A, B and C disconnect

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A2IOT011 - AdHoc Conference - Client Initiated MAS based Conference for Multiple A2 Registered SIP Clients

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Expected Results

1. Successful outcomes for all verification steps

Recommended Optional Test Coverage

If the product under test has native functionality equivalent to the A2 service/feature targeted in this test case, the tester should re-execute applicable portions of this test case using that functionality instead of the A2 service/feature. Note that in some cases, it may be necessary to disable the A2 service/feature via the appropriate Vertical Service Code prior to doing so. GENBAND will assist with any instances where a change in A2 provisioning is necessary.



A2IOT012 - MeetMe Conference - Multiple A2 Registered SIP Clients Join MAS Based Meet Me Conference

Objective

- Verify that multiple target SIP Clients can join an A2/MAS based Meet Me Conference session hosted by one of the participating clients

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

C: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. All SIP Endpoints are registered in the same domain and location within the same A2 server
4. A has Meet Me Conferencing configured with an access code, conference pass code and chairperson access code; refer to the IOT information package for specific details

Procedure

1. A calls the Meet Me Conference server DN
2. Verify A receives the Meet Me Conference welcome message
3. A enters the conference pass code
4. A then presses * and enters the chairperson access code
5. Verify A successfully joins the Meet Me Conference session and receives notification “you are the first person to join the conference”
6. B calls the Meet Me Conference access code
7. Verify B receives the Meet Me Conference welcome message
8. B enters the conference pass code
9. Verify B successfully joins the Meet Me Conference session
10. C calls the Meet Me Conference access code
11. Verify C receives the Meet Me Conference welcome message
12. C enters the conference pass code
13. Verify C successfully joins the Meet Me Conference session
14. Verify A, B and C are all able to communicate with 2 way speech path

Expected Results

1. Successful outcomes for all verification steps



A2IOT013 - Calls to A2 Registered SIP Client with Do Not Disturb Service Active

Objective

- Verify Do Not Disturb functions correctly for target SIP Clients registered in A2

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. SIP Endpoints under test having native functionality equivalent or similar to the A2 service/functionality being tested must have that native functionality DISABLED in order to avoid conflicts while testing the A2 service. If the vendor opts to execute the recommended optional additional test coverage, then the native functionality must be re-enabled for that portion of the testing.
3. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
4. All SIP Endpoints are registered in the same domain and location within the same A2 server

Procedure

1. Disable Do Not Disturb at A using the “Don Not Disturb Disable” Vertical Service Code
2. B calls A
3. Verify A receives alerting from B
4. A answers call from B
5. A and B engage in 2 way conversation and disconnect
6. A activates the Do Not Disturb service via the “Do Not Disturb Enable” Vertical Service Code
7. B calls A
8. Verify A is NOT alerted of the call from B
9. Verify B receives either recorded announcement or Voice Mail
10. Deactivate Do Not Disturb at A using the “Do Not Disturb Disable” Vertical Service Code

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A2IOT013 - Calls to A2 Registered SIP Client with Do Not Disturb Service Active

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Expected Results

1. Successful outcomes for all verification steps

Recommended Optional Test Coverage

If the product under test has native functionality equivalent to the A2 service/feature targeted in this test case, the tester should re-execute applicable portions of this test case using that functionality instead of the A2 service/feature. Note that in some cases, it may be necessary to disable the A2 service/feature via an appropriate Vertical Service Code prior to doing so. GENBAND will assist with any instances where a change in A2 provisioning is necessary.



A2IOT014 - Receive Identity of Last Client that Called and Return Call

Objective

- Verify Last Call Return for target SIP Clients registered in A2

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. All SIP Endpoints are registered in the same domain and location within the same A2 server
4. Vertical Service Codes (VSC) controlling Last Call Return are assigned and active for A. Note that activation and deactivation of Call Return by Vertical Service Code also requires a PIN that is dialed as part of the VSC string (e.g. *891234 where *89 is the VSC to deactivate Call Return and 1234 is the VSC PIN). Also note that if Call Return is deactivated, it is still possible to invoke Call Return to the last number that dialed the client prior to service deactivation; use the VSC for “Call Return CLI Erase” to clear the last caller’s identity from memory.

Procedure

1. Activate Last Call Return at A by dialing the “Call Return Enable” Vertical Service Code + PIN (1234) – see Configuration and Setup notes above
2. Deactivate Calling Line Identity Restriction at B by dialing the “Calling Line ID Restriction Disable” Vertical Service Code
3. B calls A
4. A receives alerting from B but does not answer
5. A dials the Vertical Service Code for Call Return with CLI Notification
6. Verify A receives B’s Calling Line Identification information and also an offer to return the call by pressing a specific digit
7. Press the digit for Call Return provided in the Call Return announcement
8. Verify that B is alerted
9. B answers the call
10. Verify 2 way communication between A and B
11. A and B hang up
12. Verify A and B disconnect

Expected Results

1. Successful outcomes for all verification steps



A2IOT015 - Return Call to Last Client that Called without First Learning Client Identity

Objective

- Verify Last Call Return for target SIP Clients registered in A2

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. All SIP Endpoints are registered in the same domain and location within the same A2 server
4. Vertical Service Codes (VSC) controlling Last Call Return are assigned and active for A. Note that activation and deactivation of Call Return by Vertical Service Code also requires a PIN that is dialed as part of the VSC string (e.g. *891234 where *89 is the VSC to deactivate Call Return and 1234 is the VSC PIN). Also note that if Call Return is deactivated, it is still possible to invoke Call Return to the last number that dialed the client prior to service deactivation; use the VSC for “Call Return CLI Erase” to clear the last caller’s identity from memory.

Procedure

1. Activate Last Call Return at A by dialing the “Call Return Enable” Vertical Service Code + PIN (1234) – see Configuration and Setup notes above
2. Deactivate Calling Line Identity Restriction at B by dialing the “Calling Line ID Restriction Disable” Vertical Service Code
3. B calls A
4. A does not answer
5. A dials the Vertical Service Code for Call Return Immediate
6. Verify A does NOT audibly receive B’s Calling Line Identification information
7. Verify that B is alerted
8. B answers the call
9. Verify 2 way communication between A and B
10. A and B hang up
11. Verify A and B disconnect

Expected Results

1. Successful outcomes for all verification steps



A2IOT016 - A2 Music on Hold Service Provides Audio to SIP Client on Hold

Objective

- Verify that the A2 Music On Hold service functions for the target SIP Clients

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term "SIP Endpoint under test" shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. All SIP Endpoints are registered in the same domain within the same A2 server
4. The Music On Hold service is fully configured within the A2 server and MAS
5. The Music On Hold service is assigned to A and B

Procedure

1. A calls B
2. B answers
3. A puts B on HOLD
4. Verify that B goes on HOLD and that B hears the media being presented by the A2 Music On Hold service
5. A takes B off HOLD
6. Verify that B is reconnected to A and no longer hears the media from the A2 Music On Hold service
7. A and B hang up
8. Verify that A and B disconnect

Expected Results

1. Successful outcomes for all verification steps



A2IOT017 - SIP Client Terminates on A2 Voice Mail and Leaves Message

Objective

- Verify that target SIP Clients can leave Voice Mail messages for other target SIP clients having A2 Voice Mail accounts

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. All SIP Endpoints are registered in the same domain within the same A2 server
4. A has a valid A2 Voice Mail Account

Procedure

1. B calls A
2. A does not answer
3. Verify the call forwards to Voice Mail
4. Verify that B is able to leave a message for A
5. B disconnects after leaving the message

Expected Results

1. Successful outcomes for all verification steps



A2IOT018 - Message Waiting Indication Provided to SIP Client as Part of A2 Voice Mail Service

Objective

- Verify that target SIP Clients, registered in A2 and having A2 Voice Mail accounts, receive Message Waiting Indication from the A2 Voice Mail service

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. All SIP Endpoints are registered in the same domain within the same A2 server

Procedure

1. B calls A
2. A does not answer
3. Verify the call forwards to Voice Mail according to A’s call forwarding criteria
4. B leaves a message for A and disconnects
5. Verify A receives a Message Waiting Indication from the A2 Voice mail system

Expected Results

1. Successful outcomes for all verification steps



A2IOT019 - SIP Client Retrieves Message from A2 Voice Mail Service

Objective

- Verify that target SIP Clients, registered in A2 and having A2 Voice Mail accounts, can retrieve messages from the A2 Voice Mail System. Also verify cancellation of Message Waiting Indication

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term "SIP Endpoint under test" shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. All SIP Endpoints are registered in the same domain within the same A2 server

Procedure

1. B calls A
2. A does not answer
3. Verify the call forwards to Voice Mail according to A's call forwarding criteria
4. B leaves a message for A and disconnects
5. Verify A receives a Message Waiting Indication from the A2 Voice mail system
6. A dials either the message retrieval access code for its Voice Mail account or the Vertical Service Code for Voice Mail retrieval
7. A enters the appropriate account information
8. Verify A and is presented with the voice mail from B
9. A deletes the Voice Mail and hangs up
10. Verify that the Message Waiting Indicator for A turns off

Expected Results

1. Successful outcomes for all verification steps



A2IOT020 – IDLE Client SIP Re-registration

Objective

- Verify that target SIP clients are able to re-register with the SA A2 SIP [Registrar](#), while IDLE and logged in, within the boundaries of the [Registrar's registration expiry parameters](#)

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term "SIP Endpoint under test" shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. Connect Wireshark, or similar packet trace tool, appropriately to the test environment for the purpose of capturing messaging to and from A and B
4. All SIP Endpoints are registered in the same domain within the same A2 server

Procedure

1. Start the packet trace tool
2. Select a valid user for A and B within the target A2 server and domain
3. Configure A and B such that they are able to register in the target A2 server, domain and location with the chosen SIP user
4. Verify successful registration for both clients according to their user interface
5. Verify successful registration according to the packet trace information (look for the "200 OK" messages associated with each client's registration request)
6. Note the expiry time presented to the clients in the "200 OK" messages as part of the "Contact" statement within each such message (e.g. Contact: <sip: <userid@ip address>; **expires=231**)
7. Allow A and B to remain IDLE for about 15 minutes following initial registration; ensure packet trace is running
8. Analyze the packet trace information at the end of the 15 minute period
9. Verify that both clients continue to send "REGISTER" messages to the Outbound Proxy (SBC) and receive "200 OK" messages from the SBC such that registration expiry does not occur. In this case, expect the "REGISTER" and "200 OK" messages for each client to occur within time intervals less than the value for "expires=" in the "200 Register Successful" message.

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A2IOT020 – IDLE Client SIP Re-registration

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Procedure

10. Place a call from A to B
11. Verify that B is alerted
12. B answers the call
13. Verify 2 way media exchange between A and B
14. A hangs up
15. Verify that the call terminates successfully
16. Verify that both A and B are still registered according to their respective user interfaces

Expected Results

1. Successful outcomes for all verification steps



A2IOT021 – Long Duration Call with Re-registration

Objective

- Verify that target SIP clients are able to remain connected with 2 way media and multiple re-registrations for 1 hour

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. Connect Wireshark, or similar packet trace tool, appropriately to the test environment for the purpose of capturing messaging to and from A and B
4. All SIP Endpoints are registered in the same domain within the same A2 server

Procedure

1. Start the packet trace tool
2. Select a valid user for A and B within the target A2 server and domain
3. Configure A and B appropriately such that they are able to register in the target A2 server, domain and location with the chosen SIP users
4. Place a call from A to B
5. Verify that B is alerted
6. B answers the call
7. Verify 2 way media exchange between A and B
8. Allow the call to remain connected for 1 hour
9. Verify 2 way media exchange between A and B
10. Hang up both clients
11. Verify both clients disconnect successfully
12. Stop the packet trace and analyze the messaging looking for “REGISTER” and “200 OK” messages associated with A and B
Consider use of a wireshark filter such as [sip contains REGISTER and (ip.addr == <ip address of A> or ip.addr == <ip address of B>)]
13. Verify that registrations were successfully achieved by each client within 240 seconds ongoing throughout the entire call

Expected Results

1. Successful outcomes for all verification steps



A2IOT022 – CODEC Support – Default Client Configuration

Objective

- Verify that target SIP clients are able to use their default CODEC settings to place calls successfully within A2 (i.e. verify “out of the box” CODEC configuration)

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. Connect Wireshark, or similar packet trace tool, appropriately to the test environment for the purpose of capturing messaging to and from A and B
4. All SIP Endpoints are registered in the same domain within the same A2 server

Procedure

1. Determine the default CODEC configuration for the SIP Endpoint under test
2. Start packet trace tool
3. Place a call from A to B
4. Verify alerting at B
5. B answers
6. Verify 2 way speech between A and B
7. Hang up at both A and B
8. Stop the packet trace tool
9. Analyze the packet trace data collected for this call
10. Verify that the default CODECs for A and B were used to establish the RTP connection between A and B

Expected Results

1. Successful outcomes for all verification steps



A2IOT023 – CODEC Support – All Mutually Supported CODECS

Objective

- Verify that target SIP clients are able to use all CODECS mutually supported by the SIP Endpoint under test and the A2

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. Connect Wireshark, or similar packet trace tool, appropriately to the test environment for the purpose of capturing messaging to and from A and B
4. All SIP Endpoints are registered in the same domain within the same A2 server

Procedure

1. Determine the list of CODECs supported by the SIP Endpoint under test
2. Determine which of the CODECs supported by the SIP Endpoint under test are also supported by A2
3. Execute all remaining steps, iteratively, for each mutually supported CODEC
4. Configure A and B to use the chosen CODEC mutually supported by A2 and the Sip Endpoint under test
5. Start packet trace tool
6. Place a call from A to B
7. Verify alerting at B
8. B answers
9. Verify 2 way speech between A and B
10. Hang up at both A and B
11. Stop the packet trace tool
12. Analyze the packet trace data collected for this call
13. Verify that the chosen CODEC was used to establish the RTP connection between A and B

Expected Results

1. Successful outcomes for all verification steps



A2IOT024 – CODEC Support – Mid Call CODEC Change

Objective

- Verify that target SIP clients are able to successfully change CODECs while call is in progress
- **Note that this Test Case only applies to SIP Endpoints that are capable of altering CODEC based voice quality settings while a call is in progress**

Configuration and Setup

A: SIP Endpoint under test

B: SIP Endpoint under test

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. Connect Wireshark, or similar packet trace tool, appropriately to the test environment for the purpose of capturing messaging to and from A and B
4. All SIP Endpoints are registered in the same domain within the same A2 server

Procedure

1. Choose a CODEC that is mutually supported by the SIP Endpoint under test and A2
2. Configure A and B to use the chosen mutually supported CODEC
3. Start packet trace tool
4. Place a call from A to B
5. Verify alerting at B
6. B answers
7. Verify 2 way speech between A and B
8. Change the CODEC based voice quality settings on A to a different value
9. Verify that 2 way speech path is maintained during and after the voice quality setting change
10. Hang up at both A and B

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A2IOT024 – CODEC Support – Mid Call CODEC Change

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11. Stop the packet trace tool
12. Analyze the packet trace data collected for this call
13. Verify that the chosen CODEC was used to initially establish RTP between A and B
14. Verify that the new CODEC value is used for RTP between A and B in response to the mid-call CODEC change

Expected Results

1. Successful outcomes for all verification steps



A2IOT025 - T.38 FAX Call Between 2 FAX Capable SIP Clients

Objective

- Verify that target FAX capable SIP Clients can successfully transmit and receive a FAX using T.38 protocol

Configuration and Setup

A: SIP Endpoint under test registered in A2

B: SIP Endpoint under test registered in A2

1. Unless specified otherwise, the term “SIP Endpoint under test” shall refer to the Product under Test as defined in the [Record of Execution](#).
2. Refer to the Customer IOT Information Package for details on Vertical Service Code usage, available SIP Client userids and passwords plus lab information including the A2 SIP Server address and domain, Meet Me Conference access and Voice Mail
3. All SIP Endpoints are registered in the same domain within the same A2 server
4. SIP Endpoints A and B are capable of transmitting and receiving a FAX using T.38 protocol
5. SIP Endpoints A and B are configured in a suitable IP Network space to enable packet tracing

Procedure

1. Connect A and B to FAX devices that are set to auto answer incoming calls
2. Start a packet trace using a suitable tool such as wireshark to capture the IP packets that will be transmitted/received by A and B during execution of this test scenario
3. Initiate a FAX call from A's FAX device to B's FAX device for a multiple page document
4. Verify that B's FAX device receives the call from A
5. Verify that B's FAX device answers the call from A
6. Verify that A's FAX device and B's FAX device achieve data communication
7. Verify that the 2 devices disconnect upon conclusion of the FAX transmission
8. Verify that all pages of the original document are received at B
9. Stop the packet trace
10. Analyze the packet trace
11. Verify that the packet trace indicates initial call setup using a standard voice CODEC and then invites T.38 for the FAX portion of the call.

Expected Results

1. Successful outcomes for all verification steps



Test Results



A2IOT001 Results

| | |
|-------------------------------|--|
| Test Name | SIP Client Registration to A2 System in Specific Domain |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT002 Results

| | |
|----------------------------|--|
| Test Name | Basic Call between two A2 Registered SIP Clients in Same Domain using Alias Routing |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT003 Results

| | |
|----------------------------|--|
| Test Name | SIP Client Routes to Announcement for Defined Treatment Reason |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT004 Results

| | |
|-------------------------------|--|
| Test Name | Call Forwarding for Busy and No Answer Conditions |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT005 Results

| | |
|----------------------------|--|
| Test Name | Call Forward Immediate – All Incoming Calls to A2 Registered SIP Client Forward to Specified Destination |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT006 Results

| | |
|-------------------------------|--|
| Test Name | Call Mute |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT007 Results

| | |
|-------------------------------|--|
| Test Name | Call Waiting for A2 Registered SIP Clients |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT008 Results

| | |
|----------------------------|--|
| Test Name | A2 Registered SIP Client Disables Call Waiting Service on Per Call Basis |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT009 Results

| | |
|----------------------------|--|
| Test Name | A2 Registered SIP Client uses Calling Line ID Restriction Service to Block Display Information to other A2 Clients |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT010 Results

| | |
|----------------------------|--|
| Test Name | A2 Registered SIP Client Uses Anonymous Call Rejection to Block Incoming Calls Having No Calling Line Identification |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT011 Results

| | |
|----------------------------|--|
| Test Name | AdHoc Conference - Client Initiated Conference for Multiple A2 Registered SIP Clients |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT012 Results

| | |
|----------------------------|--|
| Test Name | MeetMe Conference - Multiple A2 Registered SIP Clients Join A2 MeetMe Conference |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT013 Results

| | |
|-------------------------------|--|
| Test Name | Calls to A2 Registered SIP Client with Do Not Disturb Service Active |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Netwok Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT014 Results

| | |
|----------------------------|--|
| Test Name | Receive Identity of Last Client that Called and Return Call |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Not Executable |
| Issues | When the phone sends INVITE *91 to server, server response with SIP 183 with 'sendonly' tag. So the phone won't send DTMF back to the server. |
| Execution Notes | |



A2IOT015 Results

| | |
|----------------------------|--|
| Test Name | Return Call to Last Client that Called without First Learning Client Identity |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT016 Results

| | |
|----------------------------|--|
| Test Name | A2 Music on Hold Service Provides Audio to SIP Client on Hold |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT017 Results

| | |
|----------------------------|--|
| Test Name | SIP Client Terminates on A2 Voice Mail and Leaves Message |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT018 Results

| | |
|-------------------------------|--|
| Test Name | Message Waiting Indication Provided to SIP Client as part of A2 Voice Mail Service |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | Server would send notification to the phones, only if the subscription for MWI is enabled in the account->advanced settings. |



A2IOT019 Results

| | |
|----------------------------|--|
| Test Name | SIP Client Retrieves Message from A2 Voice Mail Service |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | Server would send notification to the phones, only if the subscription for MWI is enabled in the account->advanced settings. |



A2IOT020 Results

| | |
|-------------------------------|--|
| Test Name | IDLE Client SIP Re-registration |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT021 Results

| | |
|-------------------------------|--|
| Test Name | Long Duration Call with Re-registration |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT022 Results

| | |
|-------------------------------|--|
| Test Name | CODEC Support – Default Client Configuration |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | |



A2IOT023 Results

| | |
|----------------------------|--|
| Test Name | CODEC Support – All Mutually Supported CODECS |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | Yealink Network Technology |
| Tester Name | Karas Shi |
| Outcome (Pass or Fail) | Pass |
| Issues | |
| Execution Notes | Successful outcomes for all terminal codec: PCMU/PCMA/G729/G723/G722/G726 |



A2IOT024 Results

| | |
|----------------------------|--|
| Test Name | CODEC Support – Mid Call CODEC Change |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | |
| Tester Name | |
| Outcome (Pass or Fail) | NT |
| Issues | |
| Execution Notes | Codec change is not supported in the mid call |



A2IOT025 Results

| | |
|-------------------------------|--|
| Test Name | T.38 FAX Call Between 2 FAX Capable SIP Clients |
| Criteria | Pass: Expected Results achieved during test execution Fail: Expected Results NOT achieved during test execution |
| Test Executed By (Company) | |
| Tester Name | |
| Outcome (Pass or Fail) | NT |
| Issues | |
| Execution Notes | Not supported |

