Contents

General

Outbound call attempt to 'XXXXXX' failed. Reason: Unknown

I see the following error in the App Server logs: Outbound call attempt to 'XXXXXX' failed. Reason: Unknown

Resolution

- Ensure that the SIP Trunk has been configured correctly in accordance with the guidelines outlines in the Preparing Your Development Environment > Creating a SIP-Enabled Cisco Unified Communications Manager Cluster section of the CUAE Getting Started Guide. Choose a Non-Secure Security Profile for your SIP Trunk.
- If you are using Sip Virtual Devices, ensure that you have set up your devices as per the following guidelines:
- CUCM: Go to **Device >Phone**
- CUCM: Select Add New
- CUCM: Select Phone type (Say 7961G-GE)
- CUCM: Select Device Protocol
- CUCM: Enter a MAC address for your virtual device, that does not already exist in the CUCM database
- CUCM: Ensure that the Calling Search Space of all your devices/trunks are set to the same space
- CUCM: Save
- CUCM: Add a DN, save and reset your device.
- CUAE: Create a dual IP address for your CUAE machine
- CUAE: Enter this dual IP as SIPTrunkIP (**Plugins > List Plugins > Sip Provider**)
- CUAE: Enter this dual IP as Host Name / IP Address (**System >Global Parameters**)
- CUAE: Go to **Connections > Add Connection > Device Pool >SIP Device Pool** and enter details for the device pool.
- CUAE: Add the virtual device created in the CUCM to the device pool.
- CUCM: Change the Sip Trunk IP to the newly created dual IP.

Outbound call attempt to 'XXXXXX' failed. Reason: Normal

I see the following error in the App Server logs: Outbound call attempt to 'XXXXXX' failed. Reason: Normal

Resolution

This error is indicative of configuration mismatch between the calling phone and the called phone. Ensure that the Calling Search Spaces of the phones and SIP Trunk are set to the same search space (preferably an unrestricted one).

Outbound call attempt to 'XXXXXX' failed. Reason: Unreachable

I see the following error in the App Server logs: Outbound call attempt to 'XXXXXX' failed. Reason: Unreachable

Resolution

- Ensure that the number being called is reachable, i.e., it can be called from any other IP phone registered to the same CUCM.
- Ensure that the Calling Search Space of the Sip Trunk / Sip Virtual Phones are the same as that of the phone being called.
- Ensure that the SIP virtual devices were configured correctly as outlined here

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Call route group is empty or misconfigured for: MakeCall->Default

I see the following error in the App Server logs: Call route group is empty or misconfigured for: **MakeCall >Default**

Resolution

- On the CUAE, go to **Applications > List Applications**
- Select the Application you are facing the issue with
- In the Base Configuration section, ensure the Call Route Group is set to the correct protocol group you intend to use in your application.
- Further, go to **Connections > Groups > List Groups** to ensure that the members of the Call Route Group are correct.
- If you have recently changed group members, restart the App Server (Go to **Serviceability > Services** , check Application Server and click Restart) for the change to be reflected.

Changes made to member of Call Route Group not visible

I made a change to the members of a Call Route Group, but I don't see the change reflected.

Resolution

- Go to **Serviceablity > Services**
- Check Application Server and click Restart. This should resolve the issue.

Cannot call the Server though the SIP Trunk/ SIP Virtual devices are configured correctly

I cannot make a call to the appserver from my phone, even though my SIP Trunk / SIP Virtual Devices seem configured correctly.

Resolution

- Ensure that your phone is configured correctly and registered with the Call Manager you are using with your application.
- Ensure that the phone?s Calling Search Space is set to the same space as the SIP Trunk/ SIP Virtual Devices / other phones you are using in your setup.

Cannot hear audio though the media engine is configured correctly

I don't hear any audio, even though I have verified that the media engine configuration is correct.

Resolution

Ensure that the SIPTrunkIP (**Plugins > List Plugins > SIP Provider**) is set correctly.

Does BridgeCalls work with SIP?

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Resolution

BridgeCalls does work with SIP.

Which Call Control features work with SIP?

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Resolution

The following Call Control features work with SIP:

- Make Call
- Answer Call
- Accept Call w/ Early Media (SIP Trunk)
- Hold/Resume
- Reject Call
- Blind Transfer
- Redirect Call
- MOH
- DTMF - SIP KPML
- DTMF - RFC 2833
- Conference Calls
- Bridge Calls
- P2P Calls
- Call Forwarding

Caller hears Fast Busy when a call a call is redirected fro ma SIP Trunk

I'm trying to redirect a phone call that comes in from a sip trunk. Instead of reaching the redirected destination, the caller hears fastbusy. My trunk configuration seems correct.

Resolution

- Note that redirect information for SIP is supported in the CUAE only release 2.5.1 onwards.
- Ensure that the call is in Non-Answered state when it is being redirected.
- Use a SIP Profile in the SIP trunk with **Redirect by Application** enabled.
- Also, verify the redirect CSS configured in the SIP trunk and make sure the final destination is reachable with that CSS (The SIP trunk needs to be reset for any changes to take effect).

Do all SIP devices support JTAPI/CTI?

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Resolution

No, 7940s/7960s running SIP loads do not support JTAPI/CTI. The only SIP devices that support JTAPI are the TNP/Java generation phones like the 7970,7945/7965, etc.

Applicable to 2.4 release

Scenario 1

I'm trying to get GatherDigits to work (using CUAE version < 2.5.1 SR1), but when I use TermCondDigitList or TermCondMaxDigits, the last digit is always cropped off in the Digits variable in OnGatherDigits_Complete.

Resolution

This was a SIP specific issue and has been fixed and released with 2.5.1SR1.
