Unity Connection Call Transfer Failure to External Numbers

Contents

Introduction

This document describes how to set up a transfer to external numbers and how to troubleshoot common problems. It discusses the methods used to enable any caller to transfer calls to external numbers.

Prerequisites

Requirements

Cisco recommends that you have knowledge of these topics:

- Cisco Unity Connection (CUC)
- Cisco Unified Communications Manager (CUCM)

Components Used

The information in this document is based on these software versions:

- Cisco Unity Connection Release 8.X or later
- Cisco Unified Communications Manager Release 8.X or later

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

Call Transfer

In CUC the calls can be transferred to CUCM by three different methods:

1. Configure the call action **Transfer to Alternate Contact Number** under **Caller Input** in order to send a call out of CUC. In the call, press the configured digit for the transfer to take place. Ensure that these points are taken into consideration to use this option:Only administrators can enable this option and define the extension number via CUC Admin page. Users cannot enable this option. However, when an administrator enables this option, the users can change the extension number via TUI.The **restriction table** is checked, when a user changes the extension number via TUI conversation.The **restriction table** check box will not be checked when an administrator changes the extension number via CUC admin page.
2. Dial any number if the **Allow Transfers to Numbers Not Associated with Users or Call Handlers** check box is checked on the Greeting page. CUC performs the transfer only when the **Default System Transfer**restriction table permits it.
3. Choose the **Conversation** option after the greeting.There are two types of Conversation which can be used for this purpose:**Caller System Transfer**: This conversation prompts callers to enter the number that they want to transfer to. CUC performs the transfer only when the **Default System Transfer** restriction table permits it.**User System Transfer**: This conversation prompts callers to log on to CUC. After the caller enters their user ID and pin, Cisco Unity Connection prompts them to enter the number that they want to transfer to. CUC performs the transfer only when permitted by the transfer restriction table that is associated with the user.

Integration Requirements for Transfers from CUC to work

- If the integration between CUCM and CUC is Skinny Call Control Protocol (SCCP) - The Voicemail Port's Calling Search Space (CSS) must have the partition of the Route Pattern (RP) to the Public Switched Telephone Network (PSTN) number.
- If the integration between CUCM and CUC is Session Initiation Protocol (SIP) - The SIP trunk's Rerouting CSS must have the partition of the RP to the PSTN number.
- If the call is transferred via a CTI RP/Translation pattern - The voicemail port/SIP trunk must have access to it and the CSS of CTI RP/Translation pattern must have the partition of the RP to the PSTN number.

Troubleshoot

This section provides information in order to troubleshoot external number call transfer failure, some common problems, and the possible solutions to it.

Problem 1. "Sorry this number does not answer" Message

The greeting plays "**Sorry this number does not answer**".

Log Analysis

```
|RouteListControl::idle_CcSetupReq - RouteList(PSTNRL), numberSetup=1
numberMember=0 vmEnabled=0
|RoutePlanServer::getRouteList() - ERROR: a Routelist (xxxxxxxx-xxxx-xxxx-xxxx-xxxxxxxxxxxx)
contains no Routes
|RouteListCdrc::null0_CcSetupReq - Terminating a call after the RouteListCdrc cannot find any
more device.
|RouteListCdrc::terminateCall - No more Routes in RouteListName = PSTNRL. Rejecting the call
|RouteListCdrc::terminateCall - Sending CcRejInd, with the cause code (17), to RouteListControl
because all devices are busy/stopped.
|RouteListCdrc::terminateCall - precedenceBlocked == 0, cause = 27|
|RouteListCdrc::terminateCall - Sending CcRejInd, with cause code (27), to Cc because it has not
sent CcRegisterPartyB to Cc.|
|RouteListCdrc::sendDStopInd|
|RouteListCdrc::routeListExhausted_shutting_down_DStopConf|
```

Solution

Trace analysis shows that the Route List does not contain any routes. This is because, the Route List points to a Standard Local Route Group and the calling party does not have a Local Route Group in its device pool. In order to fix this problem, assign a local Route Group to the VM Port's/SIP Trunk's Device Pool or assign a valid Route Group/Gateway to the Route List.

Problem 2. Call is Connected and then Disconnected

The call is connected and then it is disconnected. The call transfer is successful when an internal extension initiates the call whereas the call fails, if the calling side is from PSTN.

Call Flow

PSTN>H323 GW>CUCM>Unity Connection Call Handler (Any Caller Input - Transfer to Alternate Contact Number)>External Number or CTI RP with CFA to External Number.

Core Issue/Call Flow Analysis

Here is an analysis of the call flow and the common problem for a failed call transfer:

- The First call leg is a H323 Fast Start, from the Gateway to the CUCM. CUC receives the call and transfers the call back to CUCM. CUCM in turn places the first call on hold and initiates a new call to the PSTN.
- The second call leg is a H323 Slow start. Finally, when the call is answered, the CUCM or the Gateway does not send H245 capabilities. This causes a timeout and the call is disconnected.

Solution

By default, **Wait for Far End H.245 Terminal Capability Set (TCS)** check box is checked. As a result, CUCM expects the far-end H.245 TCS before it sends its H.245 TCS. If this checkbox is unchecked, CUCM must initiate capabilities exchange.

In order to resolve this problem:

- Uncheck the **Wait for Far End H.245 Terminal Capability Set (TCS)** check box.

Or

- Make a change on the gateway so that the gateway initiates capabilities exchange.

Enter these commands in order to configure a change required on the gateway.

```
conf t
    voice service voip
   h323
          h225 start-h245 on-connect
        exit
```

Trace Analysis

```
// Gateway signaling events on CCM sdi traces
############ For the second Call Leg - CUCM to PSTN - outbound ############
// CUCM receives Invalid number format message from the Gateway
03:35:41.256 H.225 0x8002 PROGRESS RX
PROGRESS pd = 8 callref = 0x8002
Cause i = 0x809C - Invalid number format or Special Intercept
Facility i =
Progress Ind i = 0x8088 - In-band info or appropriate now available
03:35:46.398 H.225 0x8002 RELEASE_COMP RX
RELEASE_COMP pd = 8 callref = 0x8002
```

```
Cause i = 0x80A9 - Temporary failure
############ For the first Call Leg - PSTN to CUCM - Inbound #############
// CUCM sends Resources unavailable
03:35:55.473 H.225 0x84F7 RELEASE_COMP TX -->
RELEASE_COMP pd = 8 callref = 0x84F7
Cause i = 0x80AF - Resources unavailable, unspecified
03:35:55.559 H.225 0x04F7 RELEASE_COMP RX
RELEASE_COMP pd = 8 callref = 0x04F7
Cause i = 0x80AF - Resources unavailable, unspecified
```

The trace analysis shows that, the gateway originates a release complete message with temporary failure being the cause code while attempting to extend the external call to PSTN. Then, the first call leg disconnects with 'Resources unavailable, unspecified cause' message.

```
Cause code 41 (temp failure) for the 2nd call leg
Cause code 47 (resource unavailable) 1st call leg
```

Problem 3. Fast Busy Tone

Call Flow

Internal Extension/PSTN > CUCM > Unity Connection Call Handler (Any Caller Input - Transfer to Alternate Contact Number) > External Number

The Calling Party hears a **Fast Busy** tone. However, the Called Party's phone rings and when the call is answered and there is a **Dead Air**.

Log Analysis

```
// From CCM traces,
|RouteListCdrc::lockOntoDevice|2,100,57,1.134840^192.168.xxx.xx^*
|RouteListCdrc::stopRerouting|2,100,57,1.134840^192.168.xxx.xx^*
|RouteListCdrc::call_proceeding_SdlProcessNE - Cc is not reachable.
|2,100,57,1.134840^192.168.xxx.xx^*
|RouteListCdrc::terminateSelf|2,100,57,1.134840^192.168.xxx.xx^*
|RouteListCdrc::shutting_down_SdlProcessNE - ERROR:
SdlProcessNE is from unknown process|2,100,57,1.134840^192.168.xxx.xx^*
```

Solution

In order to resolve this problem, refer to Cisco bug ID CSCtx96613 and note that the server is affected by Cisco bug ID CSCtx96613

Problem 4. Reorder Tone

This section discusses trace analysis and the solutions when the Calling Party receives a **Reorder Tone** after the Caalling Party provides the caller input.

Trace Analysis

```
// From CCM Logs
// Finds two route groups in the list and two devices/gateways
|RouteList - RouteGroup count=''2''
|RouteListCdrc - RouteGroup count = 2
|RouteListCdrc - Device count = 2
// Tries to route the call through gateway 1 but CUCM considers it down
|RouteListCdrc::null0_CcSetupReq: Execute a route action.
|RouteListCdrc::whichAction -- DOWN (Current Group) = 1
|RouteListCdrc::routeAction --
current device name=aaaaaaaa-xxxx-xxxx-xxxx-xxxxxxxxxxxx, down
|RouteListCdrc::executeRouteAction: SKIP_TO_NEXT_MEMBER
// Tries to route the call through gateway 2 but CUCM considers it down
|RouteListCdrc::null0_CcSetupReq: Execute a route action.
|RouteListCdrc::whichAction -- DOWN (Current Group) = 1
|RouteListCdrc::routeAction --
current device name=bbbbbbb-xxxx-xxxx-xxxx-xxxxxxxxxxxx, down
|RouteListCdrc::executeRouteAction: SKIP_TO_NEXT_MEMBER
// No more Routes in RouteListName XXXX-PSTN-RL causing the reject
|RouteListCdrc::terminateCall -
No more Routes in RouteListName = XXXX-PSTN-RL. Rejecting the call
|RouteListCdrc::terminateCall - Sending CcRejInd, with the cause code
(41), to RouteListControl because all devices are busy/stopped.
// RouteListExhausted alert is also generated.
|GenAlarm: AlarmName = RouteListExhausted, subFac = CALLMANAGERKeyParam = ,
severity = 4, AlarmMsg RouteListName : XXXX-PSTN-RL, Reason=41,
RouteGroups(XXXX-PSTN-noCallID-RG:XXXX-PSTN-RG)
AppID : Cisco CallManager
ClusterID : StandAloneCluster
NodeID : xxxx-cucm-pub
// Reorder tone sent to the VM port
|StationD: (0126489) StartTone tone=37(ReorderTone), direction=0.
// Reorder tone received by Unity
MiuSkinny,12,Receive [Header prefix: length=20 version=18]
StationStartToneMessage (20 bytes) tone=37=DtReorderTone lineInstance=1
callReference=xxxxxxxx|
```

Solution

In order to resolve this problem:

- Reset the Route List

Or

- Restart the Call Manager Service

Problem 5. Music-on-Hold Played

The dialed number does not ring and there is a **Music-on-Hold (MoH)** played after the call has been transferred.

Trace Analysis

```
// From CCM Logs
CCM|Digit Analysis: wait_DaReq: Matching Legacy Numeric, digits=91xxxxxxxxxx|
CCM|Digit analysis: wait_DaReq - cepn=[xxxxxxxx-xxxx-xxxx-xxxx-xxxxxxxxxxx]
BlockFlag=[1]|
```

```
|RouteBlockFlag=BlockThisPattern
|RouteBlockCause=0
CCM|StationD: (0000012) StopTone.|
```

Solution

The RP is set to block the calls with an error code as "No Error" to send calls to PSTN**.** In order to resolve this problem, unblock this pattern or keep the partition of a valid route pattern above the partition of the blocked route pattern in the Voicemail Port's CSS/SIP Trunk's Rerouting CSS.

Problem 6. "I was unable to dial that number" Message

The greeting plays **"**I was unable to dial that number**"** while transferring the call.

Trace Analysis

```
// From CCM Logs
CCM|Digit Analysis: wait_DaReq: Matching Legacy Numeric, digits=91xxxxxxxxxx|
CCM|Digit analysis: wait_DaReq - cepn=[xxxxxxxx-xxxx-xxxx-xxxx-xxxxxxxxxxx]
BlockFlag=[1]|
|RouteBlockFlag=BlockThisPattern
|RouteBlockCause=21
CCM|StationD: (0000013) StopTone.|
CCM|StationD: (0000013) StartTone tone=37(ReorderTone), direction=0.|
```

Solution

The RP is set to block the calls with an error code as **"**Call Rejected**"** to send calls to PSTN**.** In order to resolve this problem, unblock this pattern or keep the partition of a valid RP above the partition of the blocked RP in the Voicemail Port's CSS/SIP Trunk's Rerouting CSS.