



# CCE Campaign Notifier

For Branded Calling Solutions

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# Abstract

With this solution we are trying to address the market requirement for

## Branded Calling Solutions

The solution focus is for UCCE Outbound Dialer and how to increase answer rate of voice outbound Campaigns



# The CNAM issue

- Carriers rely on CNAM databases to help identify outbound calls that are assigned to a specific number. But once the call reaches its intended destination, there's no telling what the person on the other end sees.
- If your customer has caller ID services enabled, they may see a name and number. But the name on the Caller ID is not something a business can completely control, which means they could receive some weird variation of the business name.
- That's because there is no universal CNAM standard; there are multiple CNAM databases across the country and carriers may subscribe to one or multiple databases. Without a sole source of truth, the company's name could be listed inaccurately or with misspellings
- Also, Businesses often relies on outsourcers to run Marketing Campaigns, therefore the Caller ID could be re-used for multiple Businesses

We need a consistent mechanism to allow the business to brand the CLID reaching the end customer.

# Branded Calling Solutions with CCE Campaign Manager

## Requirement:

- When Cisco Campaign Manager reaches the Customer Phone through the SIP Dialer, the calling number is usually considered unknown, therefore ignored. This takes the business to a low answer rate.
- Often CLIDs are sold multiple times to several businesses as part of the SIP Trunk service, reason why customer do not know who is calling.
- For this reason, businesses who tries to reach their own customers through outbound campaigns are asking for a mechanism for Branded Text Display, which makes the incoming number known to the reached end customer.
- If the end customer receives a call from their own bank service for example, they are more willing to answer.

# Cisco Proposed solution



# How the solution works

The objective is to make the Dialer CLID number a known number from the end customer side so that, when the end customer phone rings, the call is presented as from a known business/service.

This can be achieved by inserting a new item in the customer phone Contact list just before the call gets received, and then remove it just after the call has been released.

In this way the customer will see a familiar contact when phone rings.

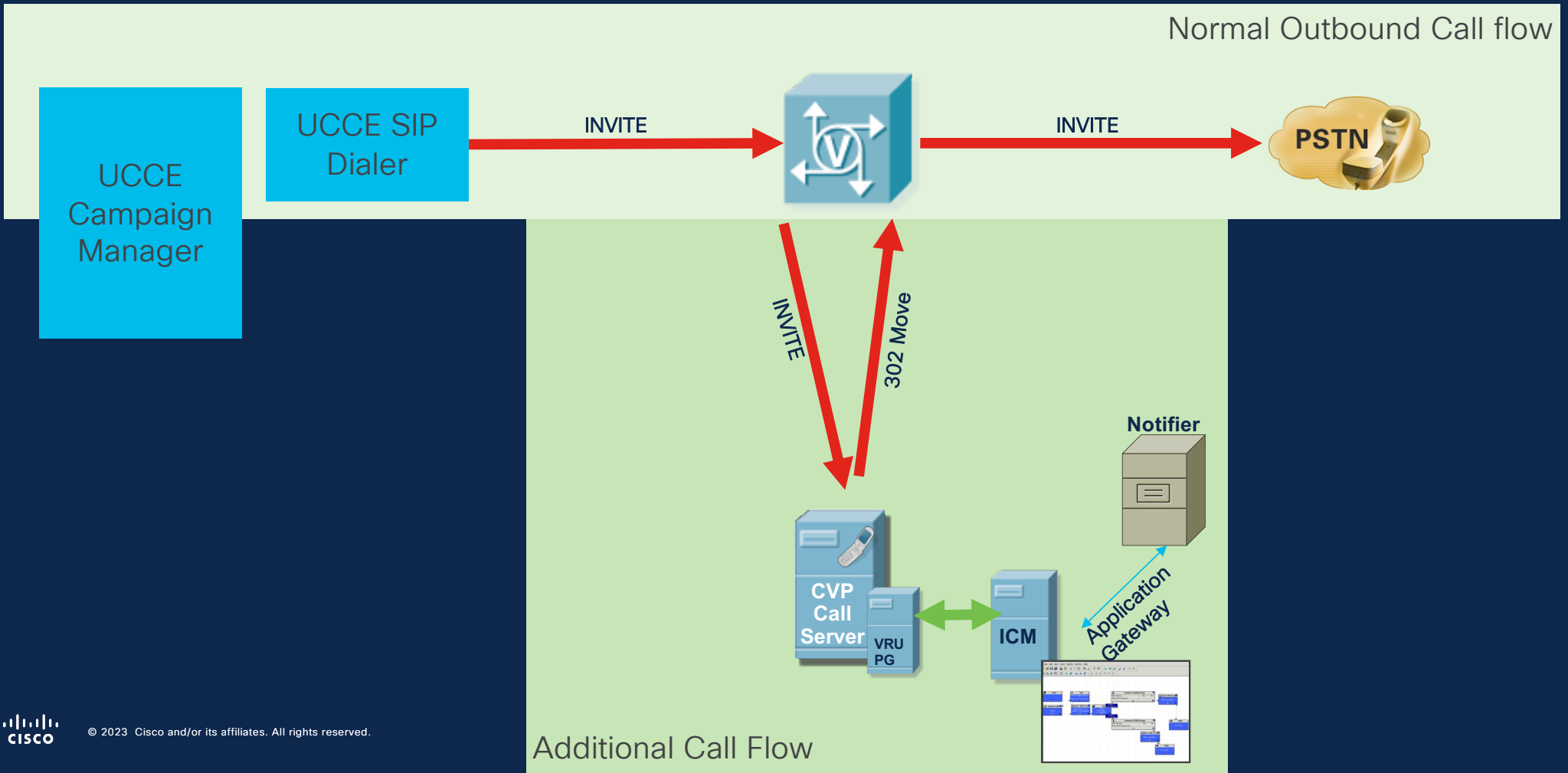
Requirement is that the Customer has the business mobile app installed on the phone: this app is used to receive pre-alert notification through Cisco Dialer and insert the new item in the phone Contact list. The new Contact will be removed when the call is released or not answered.

# How to trigger the pre-dial event

We need to intercept this event before the call reach the customer phone

- Option 1: check Dialing List for records in “D” state: not enough reactive, no scalable in case dialing list has millions of records
- Option 2: Use GW Service API to monitor CUBE activity, subscribe/approve dial in progress: it requires custom development. Dependent on IOS sw release/Gwservice
- Option 3: Intercept outgoing call at the edge, before it is sent to the PSTN, and submit that to CVP Call Server: easy solution, reliable/monitorable/selective. This solution is preferable because it does not modify the end-to-end call flow and simplify the integration using ICM Application Gateway.

# Proposed Call Flow





# Configuration



# How to engage CVP CallServer/ICM

- You need to capture a range of numbers (campaign Prefix) and send it to CVP through a single DN
- You need to pass the original DN (from the range) to ICM in order to invoke the Notifier service through the Application Gateway
- CUBE/CVP/ICM needs to deal with Custom SIP Header containing the original called number
- You need CVP to respond with the LABEL containing the PSTN customer number
- You need CVP to instruct a Redirect

## Few considerations:

- CVP does not answer the call, therefore there is no REFER but a REDIRECT
- Since Call gets not answered by CVP, the flow does not have impact on CPA, which will take place when it will catch the customer answer.
- From dialer standpoint this call is terminated into CUBE
- From CVP standpoint, the call comes from CUBE and gets redirected according to final LABEL

# CUBE Configuration

```
voice service voip
no ip address trusted authenticate
address-hiding
mode border-element
allow-connections sip to sip
no supplementary-service sip handle-replaces
signaling forward unconditional
trace
sip
header-passing
asymmetric payload full
error-passthru
early-offer forced
pass-thru headers un supp
```

// This is to replace the original dialed number with Campaign prefix with ICM DN

```
voice translation-rule 2
rule 1 /^88.* / 8812345678/
```

```
voice translation-profile RANGEtoSINGLE
translate called 2
```

// This is to copy the original DN of the Dialing List into a custom SIP Header for CVP

```
voice class sip-profiles 101
request INVITE peer-header sip To copy "sip:(.*)@" u01
request INVITE sip-header X-CISCO-DN add "X-CISCO-DN: REPLACE"
request INVITE sip-header X-CISCO-DN modify "X-CISCO-DN: (.*)" "X-CISCO-DN: \u01"
```

// This is the dialpeer to be added to CUBE to intercept the dialer and engage CVP/ICM

```
dial-peer voice 7099 voip
description Dialer Call Submitted to CVP ICM AppGw
translation-profile outgoing RANGEtoSINGLE
voice-class sip profiles 101
destination-pattern 88T
session protocol sipv2
session target ipv4:198.18.133.13
session transport tcp
incoming called-number 88T
dtmf-relay rtp-nte
codec g711ulaw
```

// required to consume 302 inside CUBE and to prevent to send it back to Dialer  
**no supplementary-service sip moved-temporarily**



# CVP Configuration

You can add Custom SIP Header through PCCE SPOG or CVP OAMP

This is the custom SIP Header introduced in CUBE in order to make the original called number available to ICM

The screenshot displays the Cisco Unified Contact Center Enterprise Management interface. The main title is "Unified Contact Center Enterprise Management". The page is titled "Device Configuration" and is for a "CVP Server" at the "Main" site. The left sidebar contains navigation options: Overview, Infrastructure, Organization, Users, Desktop, and Capacity. The main content area is divided into tabs for ICM, SIP, IVR, VXML Server, and Infrastructure. The SIP tab is active, showing various configuration options:

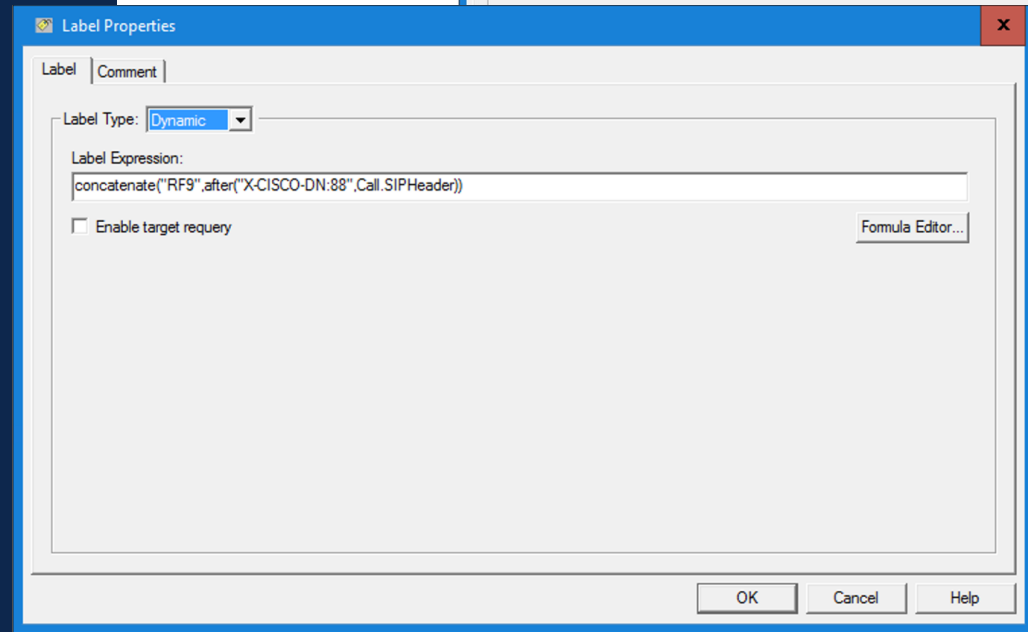
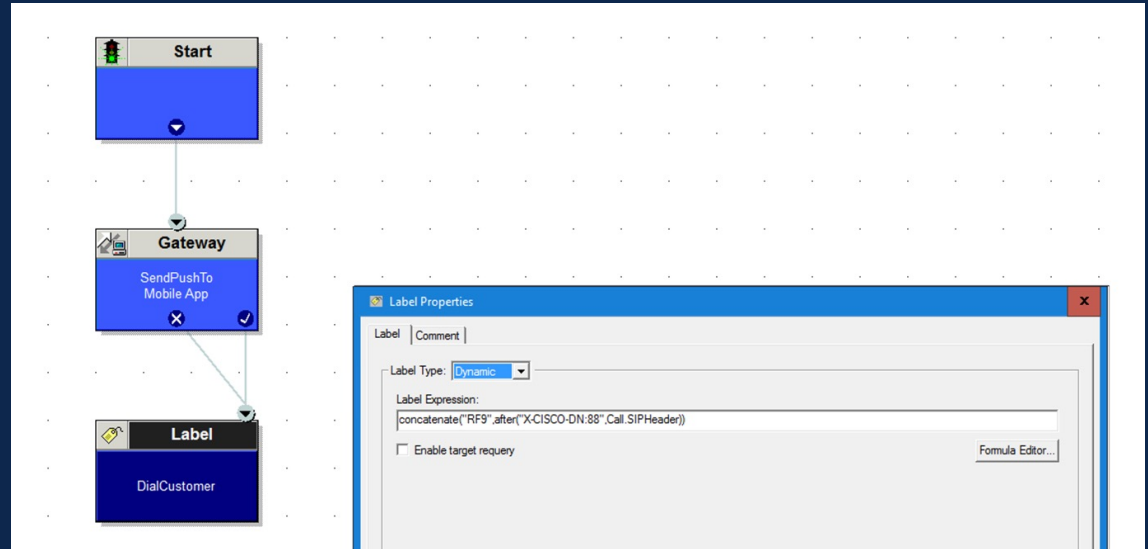
- Enable Outbound Proxy:** A toggle switch is turned off, with a warning icon and text: "Change in value requires Call Server restart".
- Outbound Proxy Host:** A dropdown menu showing "Select..."
- Outbound Proxy Port:** A text input field containing "5060".
- DNS SRV:** A warning icon and text: "Change in value requires Call Server restart".
- Enable DNS SRV Type Query:** A toggle switch is turned off.
- Resolve DNS SRV Locally:** A toggle switch is turned off.
- Outgoing Transport Type:** A dropdown menu showing "TCP".
- Port Number for Incoming SIP Requests:** A text input field containing "5060".
- Prepend Digits:** A dropdown menu showing "0".
- Use Error Refer:** A toggle switch is turned on.
- SIP Info Tone Duration:** A text input field containing "100" with "milliseconds" as a unit.
- SIP Info Comma Duration:** A text input field containing "100" with "milliseconds" as a unit.
- SIP Header Passing to ICM:** A section with a sub-header "Available" and a green plus icon. Below it is a list of headers, currently containing "X-CISCO-DN" with a close button (x) next to it.

# ICM Configuration

ICM can extract the original called number from the custom SIP Header and use that to build a Redirect response using RF as prefix, and replacing the campaign prefix (88 in this example) with the outgoing prefix (9 in this example)

The Application Gateway node invokes the Notifier service allowing a max timeout of 5 secs.

This max delay must be added to the total NoAnswerTimeout of the SIP Dialer.



# Scalability

When deploying this kind of solution, please consider the amount of traffic that SIP Dialer could generate toward CVP Call Server/ICM

- SIP Dialer can support 3000 ports and 60 calls per second (CPS)
- The most powerful Voice Gateway supports about 12 calls per second, even under the most favourable conditions
- CVP Call Server support up to 15 CPS

You should plan your design considering such limits

# Supportability

The proposed solution falls within scenarios supported at the various points of the call flow:

- Sip Dialer interfaces with a CUBE or a SIP Proxy. They are the elements of the flow that manage the Redirect as the Dialer does not support it. The integration Dialer/CUSP already works in this way: CUSP does Redirects (302) to final targets.
- CVP receives the SIP INVITE from the CUBE. In this case the call flow is typical of the Call Director mode. It might be advised to use CVP Call Servers dedicated to Call Director function. You can refer to the official documentation describing Call Director and REFER mechanism:

[https://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cust\\_contact/contact\\_center/customer\\_voice\\_portal/cvp\\_12\\_5/configuration/guide/ccvp\\_b\\_configuration-guide-12-5-1/ccvp\\_b\\_configuration-guide-12-5-1\\_chapter\\_010.html#CCVP\\_RF\\_R05E73A7\\_00](https://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cust_contact/contact_center/customer_voice_portal/cvp_12_5/configuration/guide/ccvp_b_configuration-guide-12-5-1/ccvp_b_configuration-guide-12-5-1_chapter_010.html#CCVP_RF_R05E73A7_00)

- You may have unexpected Dialer behavior if you don't consider the delay introduced by the ICM Application Gateway, or if the built ICM LABEL is incorrect. The call flow is also easy to monitor and debug. You might have outgoing calls hitting time out before being able to reach the customer, but this would fall under configuration error and bad tuning

# Conclusion





Leveraging Cisco CUBE to intercept Dialer calls could open to several integration scenarios.

The use of CVP CallServer, ICM and Application Gateway provides a valid monitoring/reporting mechanism to verify the outcome of each individual interaction

Besides integrating with 3<sup>rd</sup> party Branded Calling Solutions (aircall.io, firstorion.com, hiya.com) the Application Gateway could be used to engage a Webex Connect webhook and drive the Contact Add/Removal using Push Notification framework from Webex Connect mobile SDK framework



The bridge to possible