imagicle

How to assign a recorded conversation to the final call recipient

to version Application Suite 201x (any version)

Applies to

IAS 2020.Spring.1 or newer

Description

A new Call Recording setting allows to correlate an incoming, recorded call to the final operator who handled the call.

Cause

While using SIPREC recording method, with incoming calls reaching operators through an IVR, it happens that relevant recorded conversation is associated to IVR pilot number or to the first operator involved in the call. Customer wants to see the call associated to the final operator who actually handled the call instead.

Solution

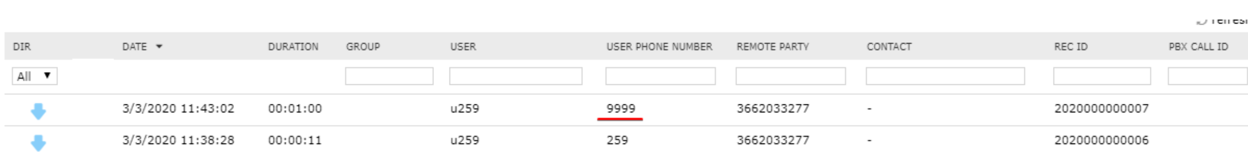
A new parameter is available in this setting file:

C:\Program Files(x86)\StonevoiceAS\Apps\Recorder\Settings**Recorder.ini**

This parameter controls Cisco SIPREC behaviour: **AllowRecordableCallInfoUpdates** and it can assume two different values:

- 0 (default): standard behaviour, assigning recording to first DN who answered incoming call.
- 1: Recording is associated to the original User's call recipient (like previous releases), while "User Phone Number" is populated with the final internal phone line who answered the call.

See below sample, where the first call (REC ID 2020...06) has been answered and handled by User "u259" with ext. #259, while second call (REC ID 2020...07) has been forwarded to User "u259" and eventually answered and handled by ext. #9999.



DIR	DATE	DURATION	GROUP	USER	USER PHONE NUMBER	REMOTE PARTY	CONTACT	REC ID	PBX CALL ID
All									
	3/3/2020 11:43:02	00:01:00		u259	9999	3662033277	-	2020000000007	
	3/3/2020 11:38:28	00:00:11		u259	259	3662033277	-	2020000000006	

Troubleshooting notes

- This setting is relevant only to SIPREC-based recording feature. Other recording methods are not affected.
- In a scenario with multiple call transfers, the recorded conversation will be correlated to the last successful call transfer recipient's DN.
- In case of a conference call, recording is associated to the extension who initiated the conference.
- This feature is taking into account a specific Cisco SIPREC signaling we receive every time the local recipient party changes. This SDP signaling includes the updated list of recipients: IAS is always considering the last recipient, every time there's a new call transfer.
- Above SIPREC signaling does not include custom call headers, like x-incoming, x-outgoing, x-anchor-index, etc.; therefore we are always considering the header included in the original SIP INVITE. This means that Call Direction and participants' order is **always evaluated with original header**, even after several call transfers.