Configure Dial Blast on Cisco Meeting Manager

Contents

Introduction

This document describes how to configure the dial blast feature that enables a User to dial out to multiple meeting participants from a Space.

Prerequisites

Requirements

Cisco recommends that you have knowledge of these topics:

- Cisco Meeting Server (CMS)
- Cisco Meeting Manager (CMM)

Components Used

This document is not restricted to specific software and hardware versions.

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, ensure that you understand the potential impact of any command.

Configure

Workflow

A Space is enabled for dial blast A space owner or a guest dials into blast dial enabled space.

Space dials out to multiple participants at once. Broadcast calls are made to all Endpoint

addresses which are added under the Blast Dial configuration page.

Supports only SIP dial-out (use an interworking gateway to interwork into SIP > H323, like VCS/Expressway).
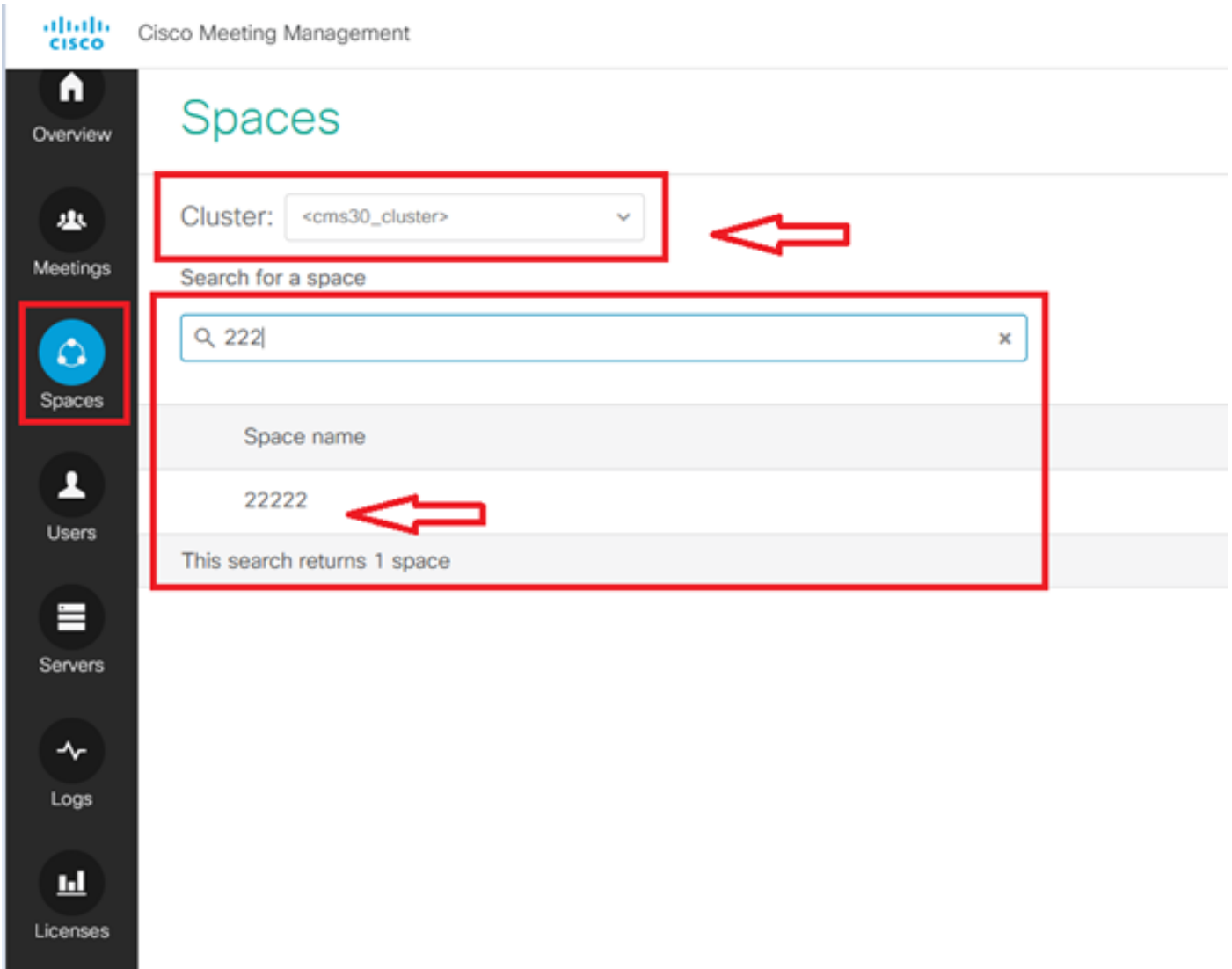
Configurations

Navigate to **Settings > Blast dial monitoring** and select **Primary**.

> **Note**: If you have multiple CMMs in your environment, select your primary CMM to initiate a Dial blast outbound calls. If for some reason, primary CMM is down, and Secondary CMM has to be used to blast dial-out calls, select secondly. In this example, there is 1 CMM in the lab, so make this primary CMM dial out.
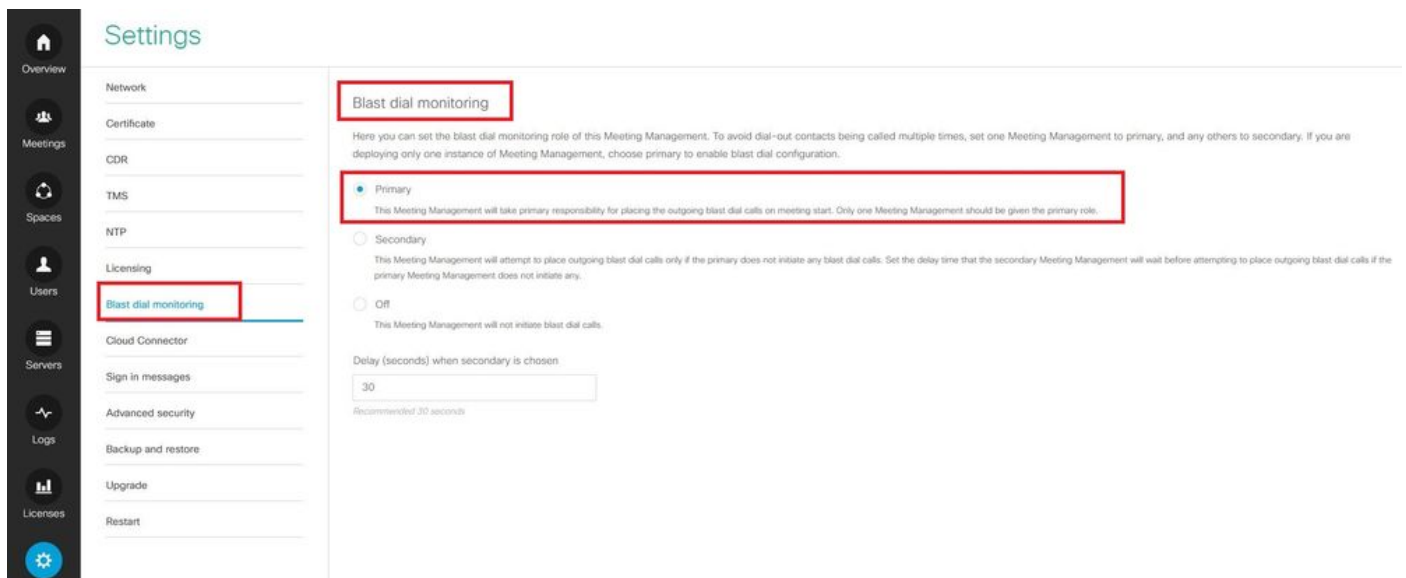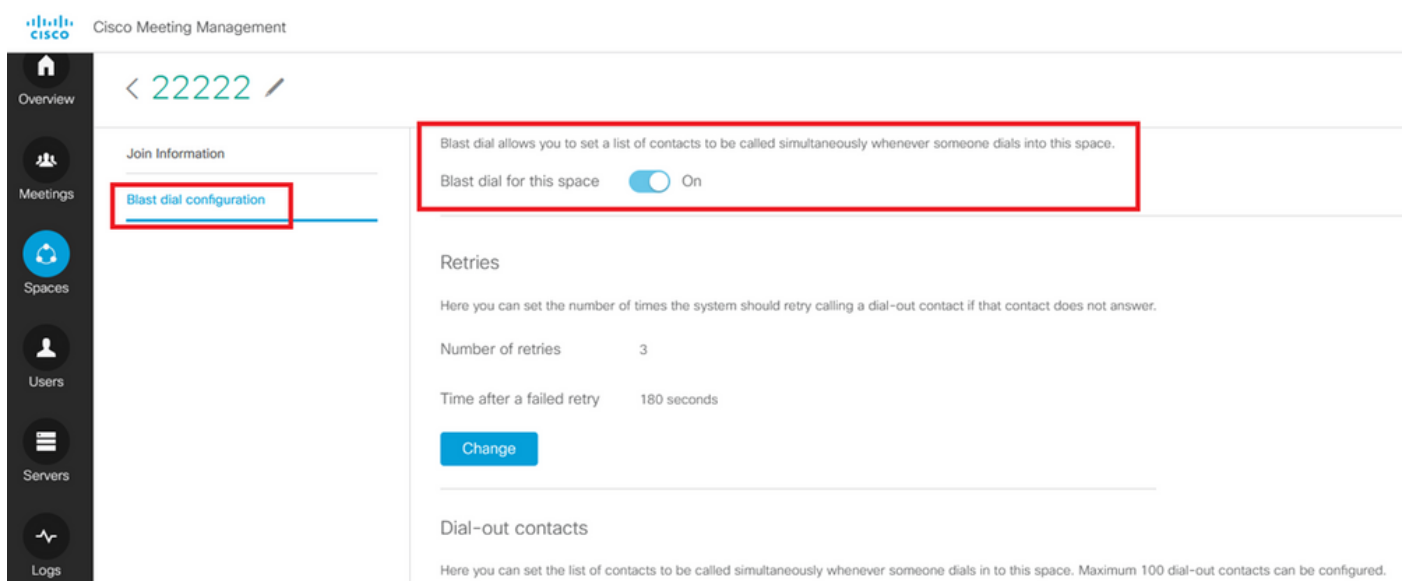

Now, since Blast dial has been enabled, select the CMS cluster (if you have added multiple clusters on 1 CMM) and space from where you want to initiate blast dial.
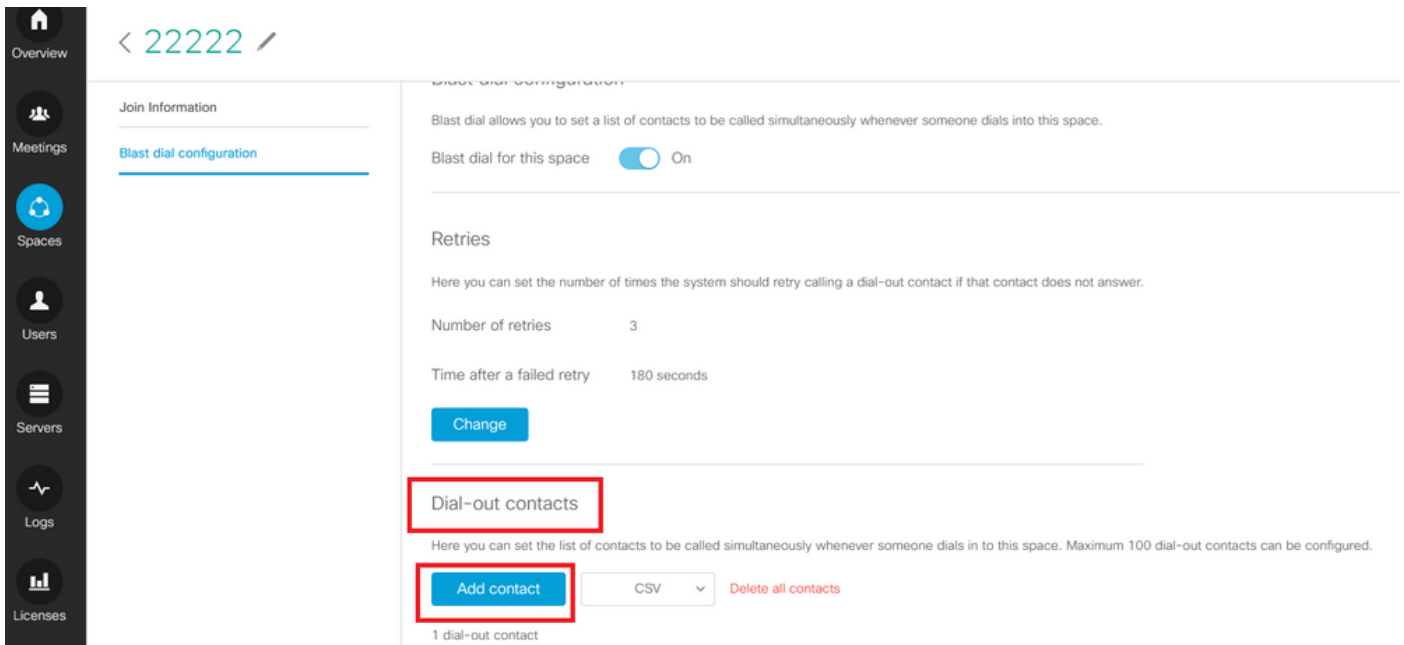
Click on Space name and enable the space for blast dial. Now you have allowed blast dial for a space as shown in the image.

> **Note**: This configuration enables a broadcast dial-out from a chosen Space.

Add Sip address/URL to be dialled out from Space 22222. Click on **Add contact**.

Add contact as shown in the image.
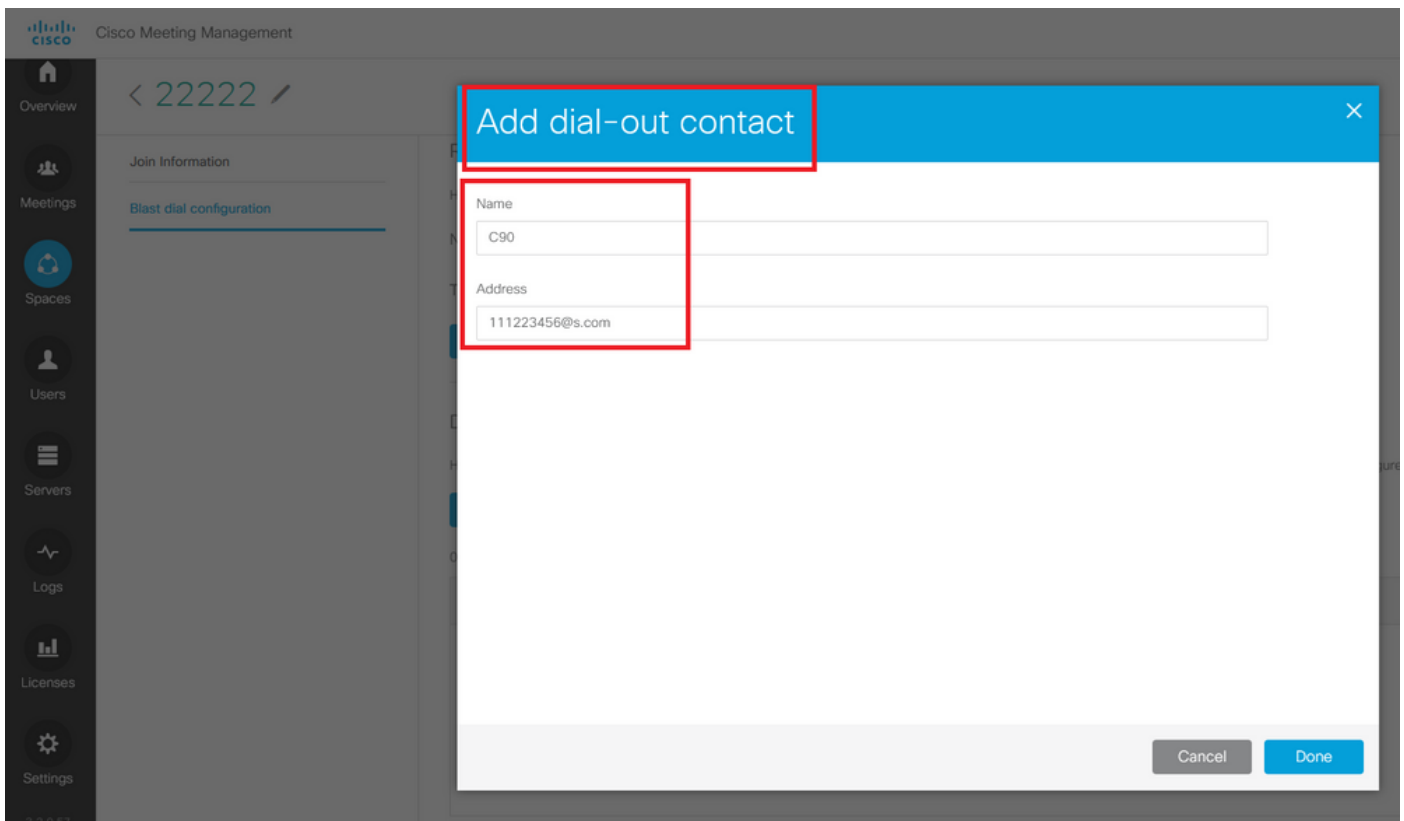
You can add as many as 100 contacts. 100 Endpoints are dialled out as soon as 1 participant dials into this space. Blast dial is initiated only when space gets activated when some user/sip device calls into this space.
Here you can set the list of contacts to be called simultaneously whenever someone dials into this space. A maximum of 100 dial-out contacts can be configured.

Dial-out contacts

Here you can set the list of contacts to be called simultaneously whenever someone dials in to this space. Maximum 100 dial-out contacts can be configured.

Add contact   CSV ⌄   Delete all contacts

2 dial-out contacts

Name	▲	Address	◆	Actions
C90		111223456@s.com		✎ 🗑
Vikdutta		VikramTAC@s.com		✎ 🗑

Another essential configuration for blast dial is Outbound rules. Ensure that the outbound rule is configured correctly to route the call from CMS to the next hop (call control). Calls can fail if outbound rules are missing or incorrectly configured.

For this lab, outbound rules are configured to route the domain s.com to call control 10.106.80.57 (Expressway).

Outbound calls

Filter [   ] [Submit Query]

	Domain	SIP proxy to use	Local contact domain	Local from domain	Trunk type	Behavior	Priority	Encryption	Tenant	
☐	10.106.80.52	10.106.80.52		<use local contact domain>	Standard SIP	Stop	0	Unencrypted	no	[edit]
☐	recorder.com	10.106.80.30:6000		<use local contact domain>	Standard SIP	Stop	0	Unencrypted	no	[edit]
☐	s.com	10.106.80.57		<use local contact domain>	Standard SIP	Stop	0	Unencrypted	no	[edit]
	[ ]	[ ]	[ ]	[ ]	Standard SIP ⌄	Stop ⌄	0	Auto ⌄		Add New Reset

Verify

Use this section in order to confirm that your configuration works properly.

In order to test the blast dial, initiate a webrtc call to space 22222. CMS dials out to add contacts under the dial blast space page on CMM.

2022-09-14	19:49:10.696	Info	call 210: configured - API call leg cceaabdd-4fd7-43c5-85cc-e35a0425e2e2
2022-09-14	19:49:10.696	Info	call 210: setting up combined RTP session for DTLS (combined media and control)
2022-09-14	19:49:10.860	Info	participant "guest1075392600" joined space 87103471-1b6d-41e7-85b0-8ac17eadc6cb (22222)
2022-09-14	19:49:10.860	Info	participant "guest1075392600" (cceaabdd-4fd7-43c5-85cc-e35a0425e2e2) joined conference 6671e198-2384-4b4b-9e36-1d712ec1c9c0 via WB3
2022-09-14	19:49:10.871	Info	call 210: starting DTLS combined media negotiation (as initiator)
2022-09-14	19:49:12.243	Info	call 210: completed DTLS combined media negotiation
2022-09-14	19:49:20.944	Info	API call leg ccf134f7-1801-4e8a-a824-c7823b3244c3 in call 6671e198-2384-4b4b-9e36-1d712ec1c9c0 (API call ffb3c0ee-cdc6-4d05-9929-9e68a586be7f)
2022-09-14	19:49:20.944	Info	10.106.80.60: API user "admin" created new participant ccf134f7-1801-4e8a-a824-c7823b3244c3, call ffb3c0ee-cdc6-4d05-9929-9e68a586be7f
2022-09-14	19:49:20.944	Info	call 211: outgoing SIP call to "111223456@s.com" from space "22222"
2022-09-14	19:49:20.945	Info	call 211: configured - API call leg ccf134f7-1801-4e8a-a824-c7823b3244c3 with SIP call ID "23a266c3-86b9-4c7c-ba4e-50767cba0719"
2022-09-14	19:49:20.947	Info	call 211: setting up UDT RTP session for DTLS (combined media and control)
2022-09-14	19:49:21.168	Info	API call leg e5759c3b-81e0-4fe8-8f50-fb53c1bd8a80 in call 6671e198-2384-4b4b-9e36-1d712ec1c9c0 (API call ffb3c0ee-cdc6-4d05-9929-9e68a586be7f)
2022-09-14	19:49:21.168	Info	10.106.80.60: API user "admin" created new participant e5759c3b-81e0-4fe8-8f50-fb53c1bd8a80, call ffb3c0ee-cdc6-4d05-9929-9e68a586be7f
2022-09-14	19:49:21.168	Info	call 212: outgoing SIP call to "VikramTAC@s.com" from space "22222"
2022-09-14	19:49:21.169	Info	call 212: configured - API call leg e5759c3b-81e0-4fe8-8f50-fb53c1bd8a80 with SIP call ID "8698875e-c551-416f-b1ee-be415b3ba515"
2022-09-14	19:49:21.170	Info	call 212: setting up UDT RTP session for DTLS (combined media and control)
2022-09-14	19:49:21.190	Info	call 211: SIP call ringing
2022-09-14	19:49:21.190	Info	participant "111223456@s.com" joined space 87103471-1b6d-41e7-85b0-8ac17eadc6cb (22222)
2022-09-14	19:49:21.190	Info	participant "111223456@s.com" (ccf134f7-1801-4e8a-a824-c7823b3244c3) joined conference 6671e198-2384-4b4b-9e36-1d712ec1c9c0 via SIP
2022-09-14	19:49:21.227	Info	call 212: SIP call ringing
2022-09-14	19:49:21.228	Info	participant "VikramTAC@s.com" joined space 87103471-1b6d-41e7-85b0-8ac17eadc6cb (22222)
2022-09-14	19:49:21.228	Info	participant "VikramTAC@s.com" (e5759c3b-81e0-4fe8-8f50-fb53c1bd8a80) joined conference 6671e198-2384-4b4b-9e36-1d712ec1c9c0 via SIP
2022-09-14	19:49:28.710	Info	conference "22222": unencrypted call legs now present
2022-09-14	19:49:33.411	Info	call 212: starting DTLS UDT media negotiation (with far end as initiator)

Owner
Start time 09/14/2022 19:49
End time
Duration 00:03:29
Lobby Empty

Pane placement | Move participants | Add participants | Set layout for all | Mute all | Unmute all | Clear all importance | Recording | Streaming | Mute on entry | Lock | Event logs | End meeting

Q Search

Activity Important Speaking Status Connected Moved Disconnected

Showing 3 of 3 participants

	1 ▲	Participant name		Join time 2 ▼	Leave time	Actions
☐ ✗	✓	VD	⋯	09/14/2022 19:51		◌ ▭◦ ✗ ☎ ✕
☐ ✗	✓	Vikram	⋯	09/14/2022 19:49		◌ ▭◦ ✗ ☎ ✕
☐ ✗	✓	VDC90	⋯	09/14/2022 19:49		◌ ▭◦ ✗ ☎ ✕

Troubleshoot

There is currently no specific troubleshooting information available for this configuration.

Pane placement | Move participants | Add participants | Set layout for all | Mute all | Unmute all | Clear all importance | Recording | Streaming | Mute on entry | Lock | Event logs | End meeting