

# Use Direct IP Address Calls Between Two Endpoints to Troubleshoot Call Quality Issues



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## Introduction

This document describes how to place a point-to-point IP video call between two devices that are in the same network in order to isolate the possibility of WAN or infrastructure device issues.

## Background Information

At times there can be issues with call quality (audio/video), call connectivity, or one-way audio/video. There are multiple devices that are involved in the complete call setup, so the issue can be difficult to troubleshoot.

The WAN can be one of the reasons for the poor call quality if it is not properly configured or if the Quality of Service (QoS) values are not marked properly along the call route. There is also the possibility that the Cisco TelePresence Video Communication Server (VCS) or the Cisco Unified Communications Manager (CUCM) can cause the issue if there is a configuration problem.

This document describes a method that you can use in order to test whether the device itself is the cause of the issue, or whether there are other factors (such as the WAN or the infrastructure devices) that cause the issue.

## Troubleshoot

It is possible to place calls between two devices in these two scenarios:

- When both devices are in the same network.
- When both devices are on public IPs and they are reachable from one another.

These calls can only be placed when the IP address of one device is dialed from the other device. The IP

address can be dialed via H323 or Session Initiation Protocol (SIP).

## IP-to-IP Direct Call Through H323

This section provides the standard H323 configurations that allow calls to be placed directly between two endpoints. You can choose to implement these configurations through the use of either the device CLI or the GUI.

*Note:* These configurations are required on both of the devices.

### CLI Configuration

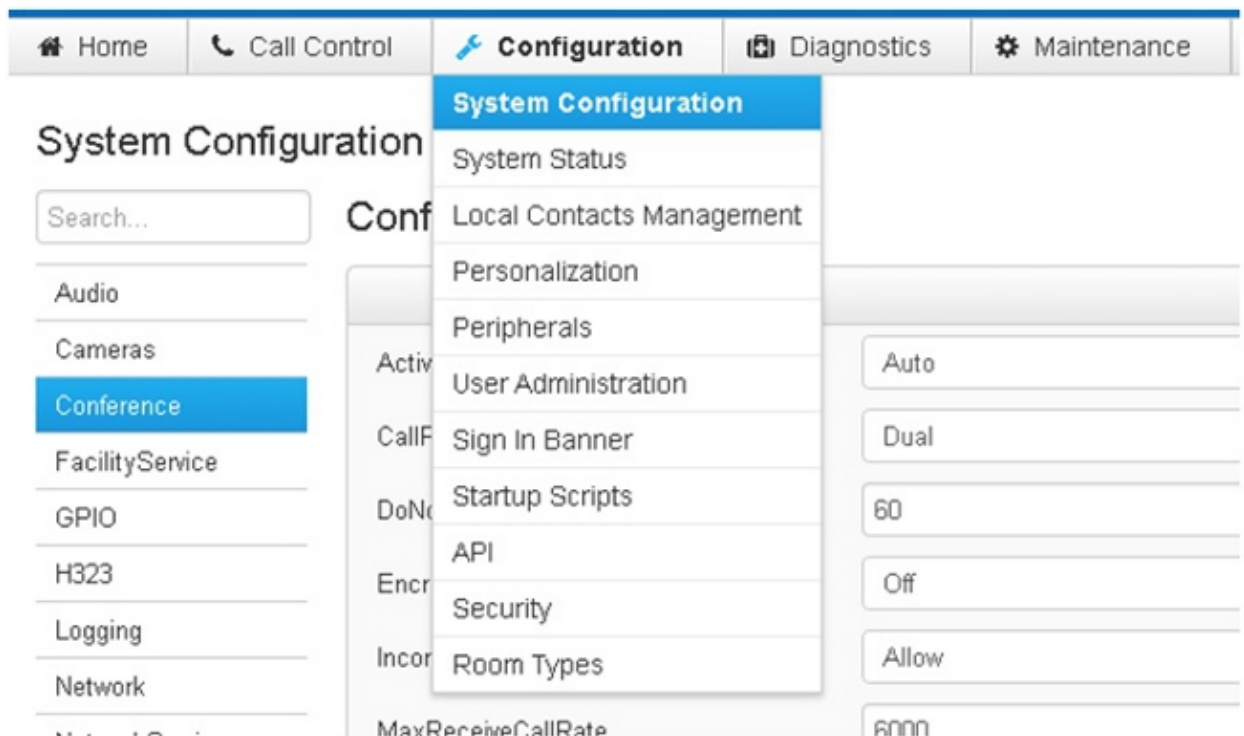
In order to implement this configuration via the CLI, establish a Secure Shell (SSH) session to the device and log in with Administrator credentials. Once logged in, use this information in order to configure the device:

```
xConfiguration H323 Profile 1 CallSetup Mode: Direct
xConfiguration Conference 1 DefaultCall Protocol: H323
xConfiguration NetworkServices H323 Mode: On
```

### GUI Configuration

Complete these steps in order to implement this configuration via the GUI:

1. Log into the endpoint GUI.
2. Navigate to *Configuration > System Configuration* and click the *Conference* tab on the left side of the screen:



3. In the *DefaultCall* section, choose *H323* from the *Protocol* drop down menu and click *Save*:

DefaultCall	
Protocol	Auto ▼
Rate	(64 to 6000)

Auto  
**H323**  
 Sip  
 H320

4. Navigate to **Configuration > System Configuration** and click the **H323** tab on the left side of the screen. In the **Profile 1** section, choose **Direct** from the **CallSetup Mode** drop down menu, and then click **Save**:

Home | Call Control | Configuration | Diagnostics | Maintenance | admin

### System Configuration

H323

- Audio
- Cameras
- Conference
- FacilityService
- GPIO
- H323
- Logging
- Network
- NetworkServices
- Peripherals
- Phonebook Server

NAT

Address  (0 to 64 characters)

Mode

Profile 1

CallSetup Mode

Encryption KeySize

PortAllocation

5. Navigate to **Configuration > System Configuration** and click the **Network Services** tab on the left side of the screen. Choose **On** from the **H323 Mode** drop down menu and click **Save**:

## System Configuration

Search...

### NetworkServices

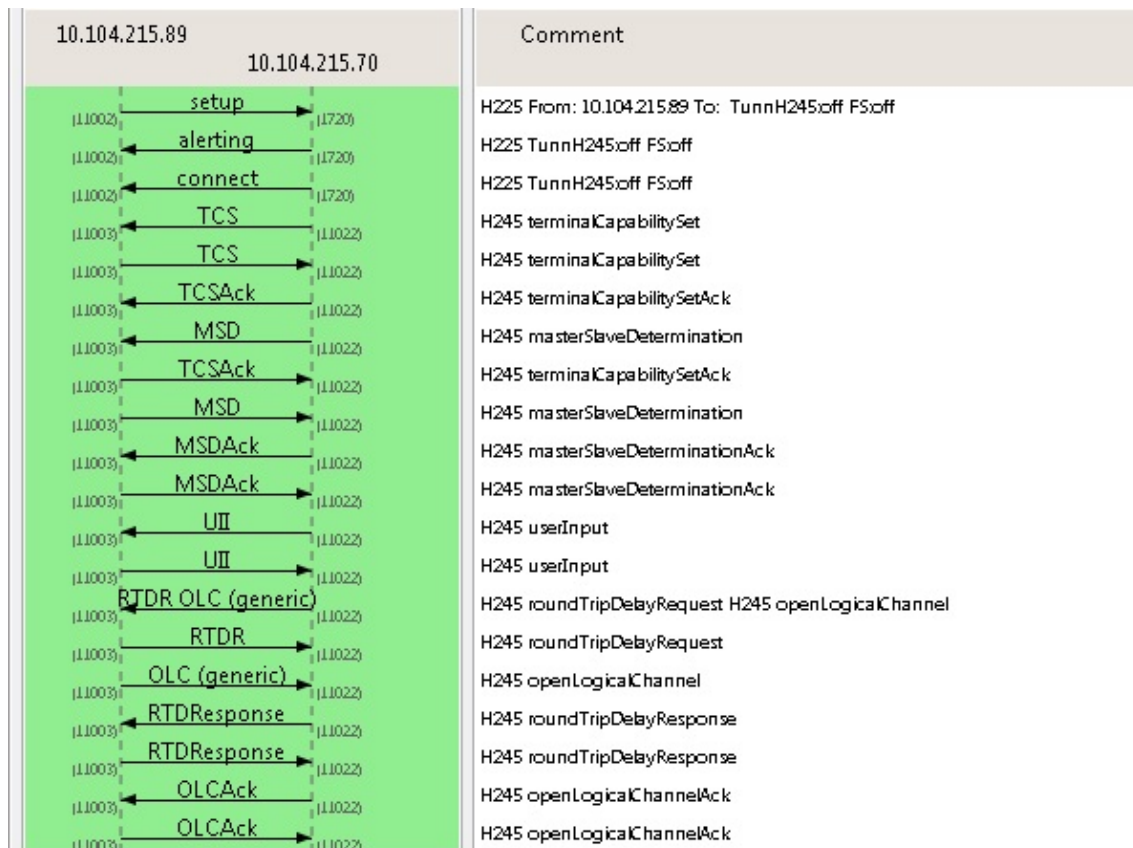
Refresh

- Audio
- Cameras
- Conference
- FacilityService
- GPIO
- H323
- Logging
- Network
- NetworkServices**
- Peripherals

CDP Mode	On	
H323 Mode	On	Undo
	Off	
HTTP Mode	On	
Medianet Metadata	Off	
SIP Mode	Off	
Telnet Mode	On	
WelcomeText	On	

### H323 Call Flow

This image illustrates the call flow when H323 is used:



# IP-to-IP Direct Call Through SIP

This section provides the standard SIP configurations that allow calls to be placed directly between two endpoints. You can choose to implement these configurations through the use of either the device CLI or the GUI.

*Note:* These configurations are required on both of the devices.

## CLI Configuration

In order to implement this configuration via the CLI, establish an SSH session to the device and log in with Administrator credentials. Once logged in, use this information in order to configure the device:

```
xConfiguration NetworkServices SIP Mode: On
xConfiguration SIP Profile 1 Proxy 1 Address: ""
xConfiguration SIP Profile 1 Proxy 1 Discovery: Manual
xConfiguration Conference 1 DefaultCall Protocol: SIP
```

## GUI Configuration

Complete these steps in order to implement this configuration via the GUI:

1. Log into the endpoint GUI.
2. Navigate to **Configuration > System Configuration** and click the **Network Services** tab on the left side of the screen. Choose **On** from the **SIP Mode** drop down menu and click **Save**.
3. Navigate to **Configuration > System Configuration** and click the **SIP** tab on the left side of the screen. Ensure that the **Proxy 1** section is left blank and click **Save**:

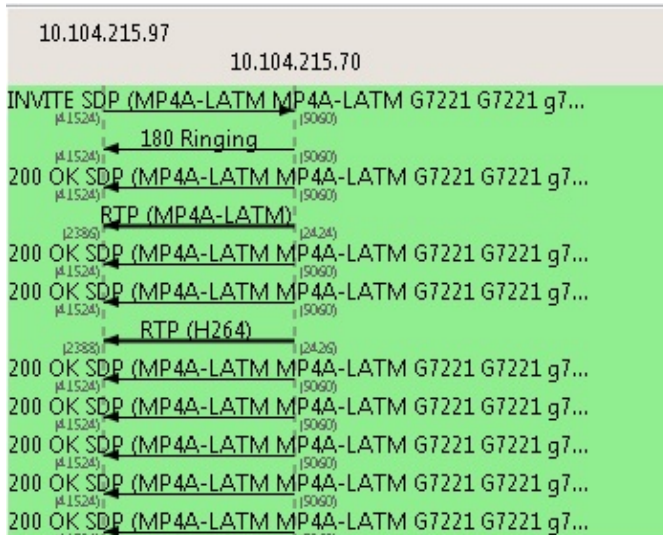
The screenshot shows the GUI configuration page for SIP. On the left, there is a navigation menu with the following items: SIP (highlighted in blue), Standby, SystemUnit, Time, UserInterface, and Video. The main content area is divided into several sections, each with a title and a list of configuration fields. The 'Proxy 1' section is highlighted with a red border. The fields and their values are as follows:

Section	Field	Value	Character Limit
SIP	DefaultTransport	Tls	
	DisplayName	Network	0 to 255 characters
	Line	Private	
	Mailbox		0 to 255 characters
	Outbound	Off	
	TlsVerify	Off	
	Type	Standard	
	URI		0 to 255 characters
Authentication 1	LoginName		0 to 128 characters
	Password		0 to 128 characters
Ice	DefaultCandidate	Host	
	Mode	Auto	
Proxy 1	Address		0 to 255 characters
	Discovery	Manual	

4. Navigate to *Configuration > System Configuration* and click the *Conference* tab on the left side of the screen. In the *DefaultCall* pane, choose *SIP* from the *Protocol* drop down menu and click *Save*.

## SIP Call Flow

This image illustrates the call flow when SIP is used:



## Diagnosis

At this point, you can dial the IP address of one device from the other device and verify whether the call is connected properly. If the call goes through as expected, then there is no need to further concentrate on the infrastructure device settings in order to isolate the issue.

If the same problem persists, then the issue is with either the device itself or the with the network (if the call is placed over the WAN).