

Video Terminal Configuration Guide

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1 Introduction

Audio and video conferencing is a set of video communication and audio communication in the new generation of interactive multimedia communications system. For the enterprise, there are maybe different types of terminals, and this document is just used for the VIDEO terminals configuration guide for the end user.

Note:

Different vendors have different types of the video terminals; this document is a general guide for the end user.

2 General Configuration for the Terminal

If the enterprise administrator wants to use the video terminal to join the video conference, the terminal should register in IMS, then the end user can dial the access number to join the conference.

2.1 Register Server

ims.omantel.om

2.2 Proxy Server

If the terminal supports domain configuration, then enter the proxy server domain:

imsreg.omantel.om

If the terminal only support IP configuration, then enter the IP address of the proxy server:

212.72.5.136/212.72.5.138

Note:

These two IPs must be reachable from the terminal; otherwise the terminal can't register in IMS;

2.3 User Name and Password

When the enterprise subscribes the video conference, Omantel will assign one access number for the enterprise and together 5 video terminal numbers to register in IMS. For example:

User Name: +96824243496@ims.omantel.om

Password: XXXXX

3 Configuration example

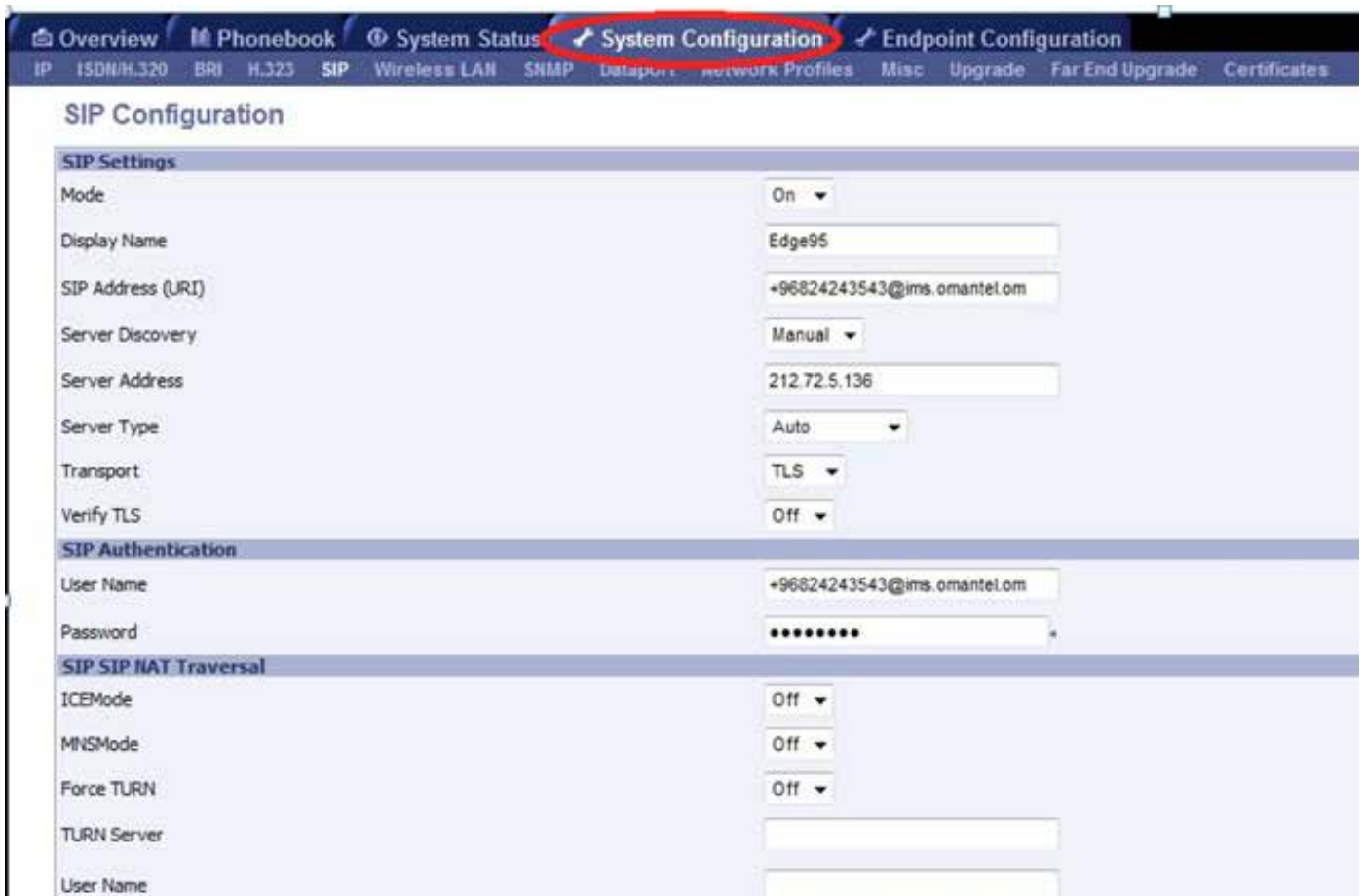
3.1 Disable H.323 protocol and enable SIP protocol

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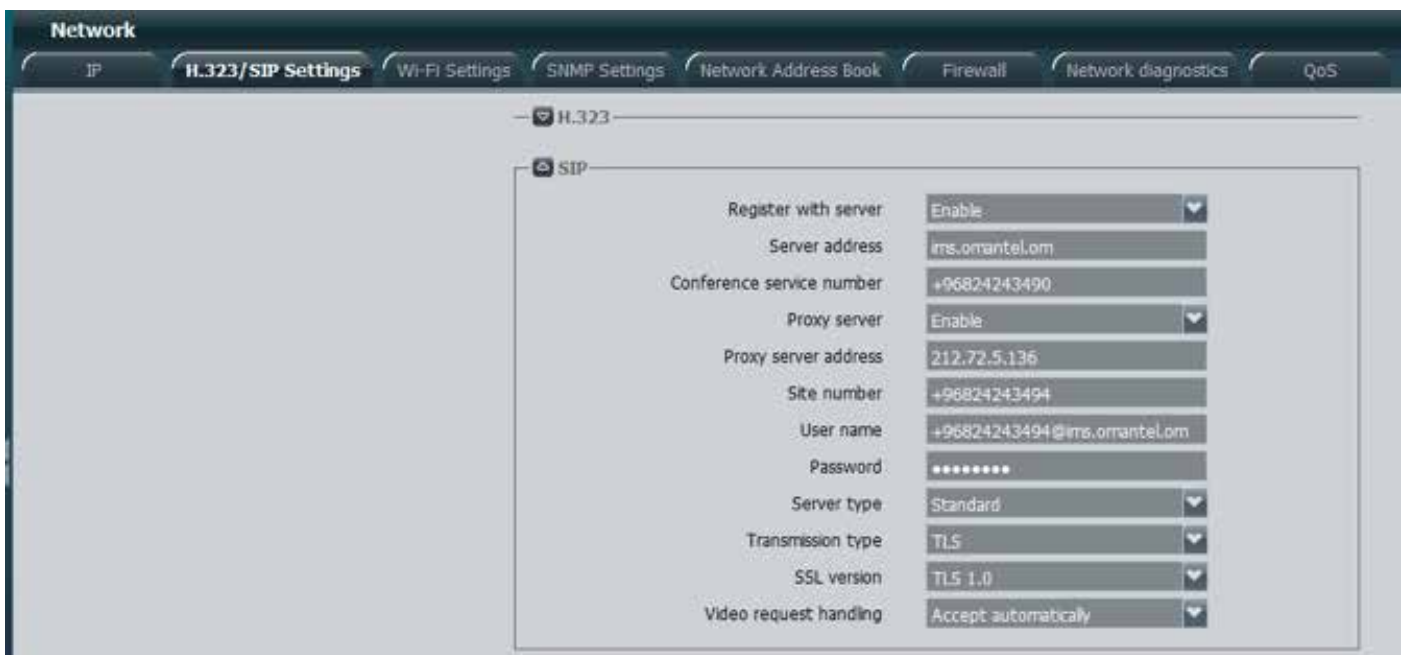
- 2) Cisco video terminal configuration guide

Go to the H.323 label switch the mode of H.323 off, and go to the SIP label to switch the SIP mode on



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Go to the H.323 label disable H.323 server, and go to the SIP label to enable the SIP server.



3.2 Configure the SIP register server and proxy server.

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IP Network

Update

SIP Settings

Enable SIP:

SIP Server Configuration: Specify

Registrar Server: ims.omantel.om

Proxy Server: 212.72.5.136

Transport Protocol: Auto

User Name: +96824243494@ims.omantel.om

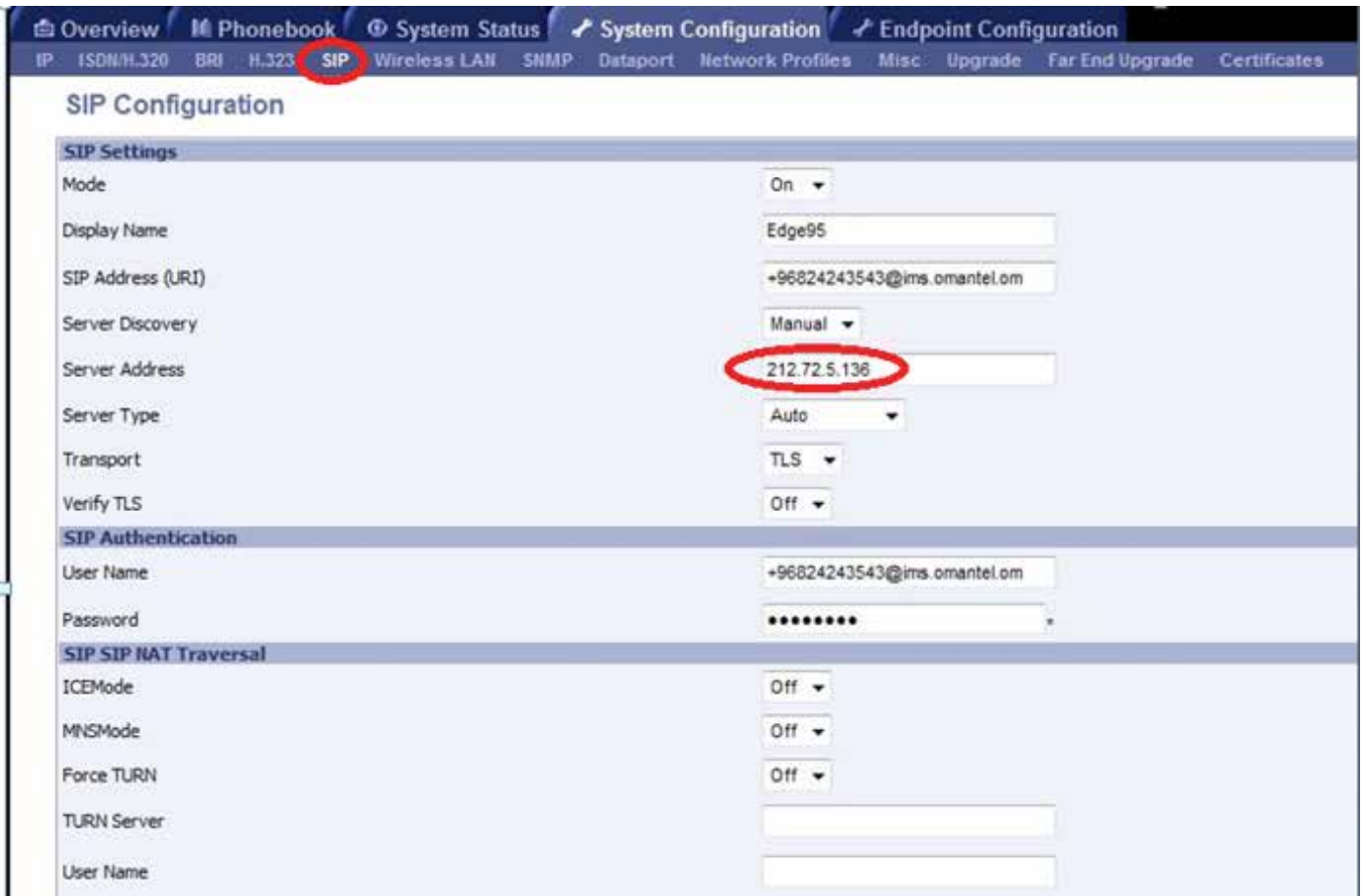
Domain User Name: +96824243494@ims.omantel.om

Password:

Directory:

Microsoft Lync Server 2010:

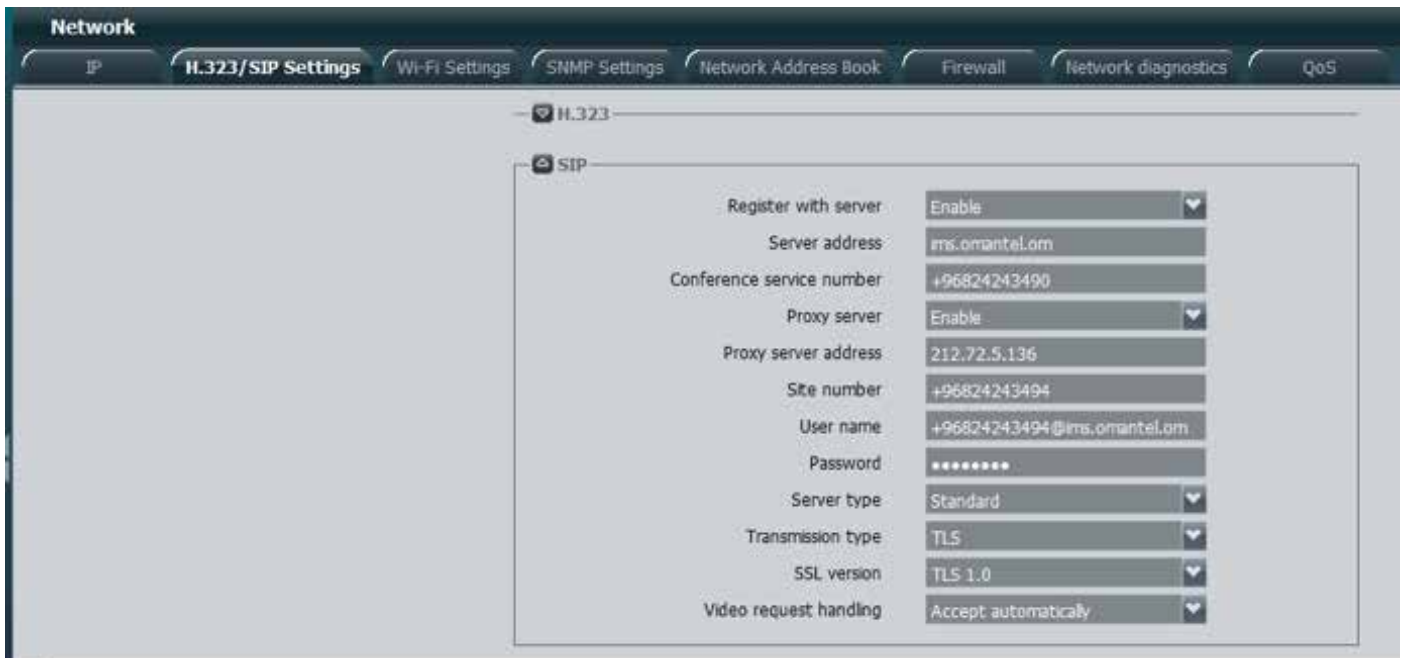
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The screenshot shows the Cisco SIP Configuration page. The 'SIP' tab is highlighted in red. The configuration is organized into three sections: SIP Settings, SIP Authentication, and SIP SIP NAT Traversal. In the SIP Settings section, the 'Server Address' field is circled in red and contains the IP address '212.72.5.136'. Other fields include Mode (On), Display Name (Edge95), SIP Address (URI) (+96824243543@ims.omantel.om), Server Discovery (Manual), Server Type (Auto), Transport (TLS), and Verify TLS (Off). The SIP Authentication section includes User Name (+96824243543@ims.omantel.om) and Password (masked with dots). The SIP SIP NAT Traversal section includes ICEMode (Off), MNSMode (Off), Force TURN (Off), and TURN Server (empty).

| Section | Field | Value |
|-----------------------|-------------------|-----------------------------|
| SIP Settings | Mode | On |
| | Display Name | Edge95 |
| | SIP Address (URI) | +96824243543@ims.omantel.om |
| | Server Discovery | Manual |
| | Server Address | 212.72.5.136 |
| | Server Type | Auto |
| | Transport | TLS |
| Verify TLS | Off | |
| SIP Authentication | User Name | +96824243543@ims.omantel.om |
| | Password | ***** |
| SIP SIP NAT Traversal | ICEMode | Off |
| | MNSMode | Off |
| | Force TURN | Off |
| | TURN Server | |
| | User Name | |
| | | |

3) Huawei video terminal configuration guide



The screenshot shows the Huawei SIP Settings page. The 'SIP' tab is selected. The configuration is organized into a single section: SIP. The fields and their values are: Register with server (Enable), Server address (ims.omantel.com), Conference service number (+96824243490), Proxy server (Enable), Proxy server address (212.72.5.136), Site number (+96824243494), User name (+96824243494@ims.omantel.com), Password (masked with dots), Server type (Standard), Transmission type (TLS), SSL version (TLS 1.0), and Video request handling (Accept automatically).

| Field | Value |
|---------------------------|------------------------------|
| Register with server | Enable |
| Server address | ims.omantel.com |
| Conference service number | +96824243490 |
| Proxy server | Enable |
| Proxy server address | 212.72.5.136 |
| Site number | +96824243494 |
| User name | +96824243494@ims.omantel.com |
| Password | ***** |
| Server type | Standard |
| Transmission type | TLS |
| SSL version | TLS 1.0 |
| Video request handling | Accept automatically |

Note:

1. Transport protocol should be TLS, port is 5061, but some POLYCOM terminals should be configured as Auto, so first use Auto, if the POLYCOM terminal can't register in IMS, then uses TLS to register.
2. Proxy Server: If the terminal supports the domain name, then enter imsreg.omantel.om, if the terminal only supports IP input, then choose one of the IPs to enter: 212.72.5.136 or 212.72.5.138.

3.3 Configure the username and password to register in IMS

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IP Network Update

SIP Settings

Enable SIP:

SIP Server Configuration: Specify ▾

Registrar Server:

Proxy Server:

Transport Protocol: Auto ▾

User Name:

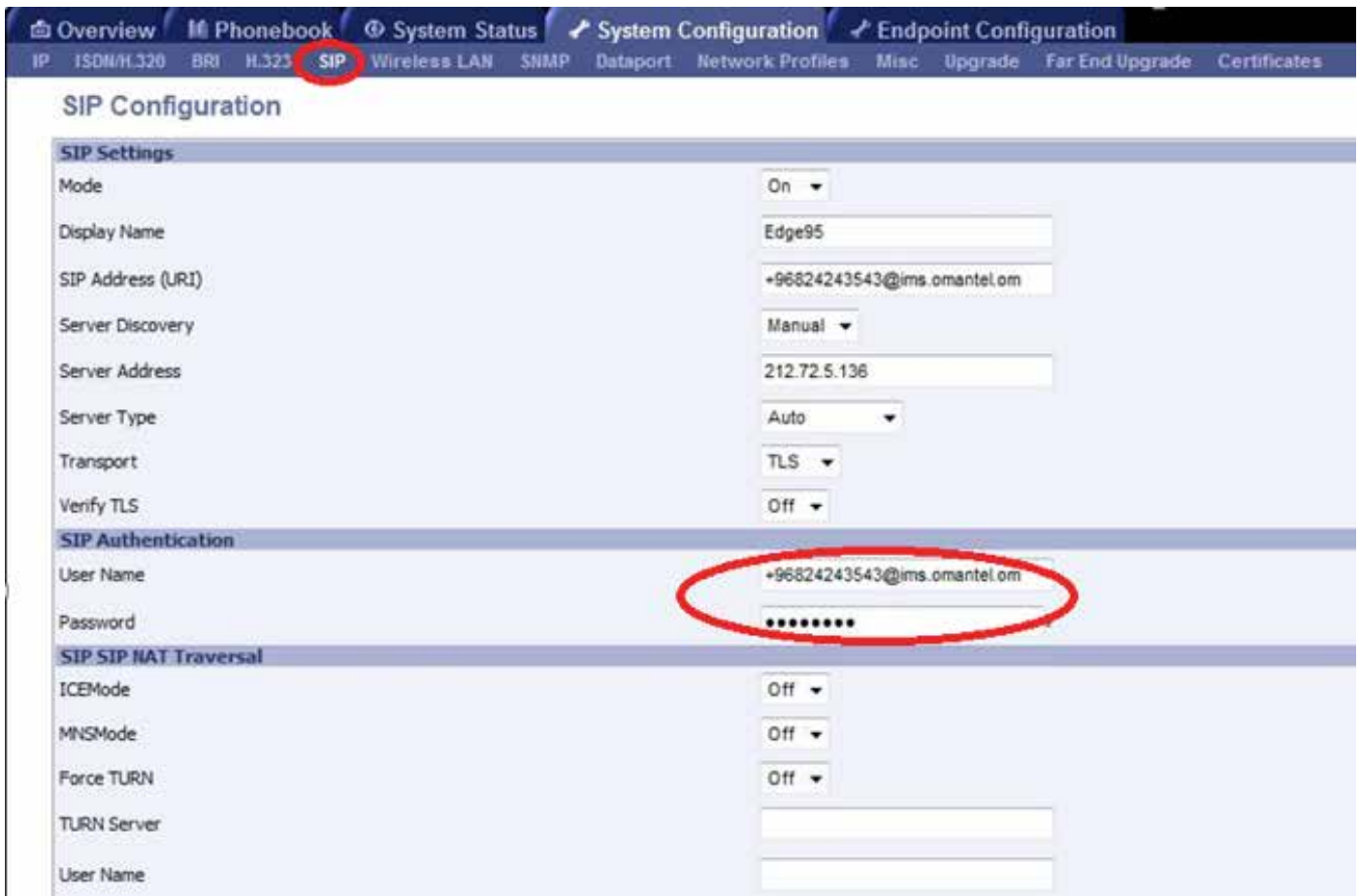
Domain User Name:

Password:

Directory:

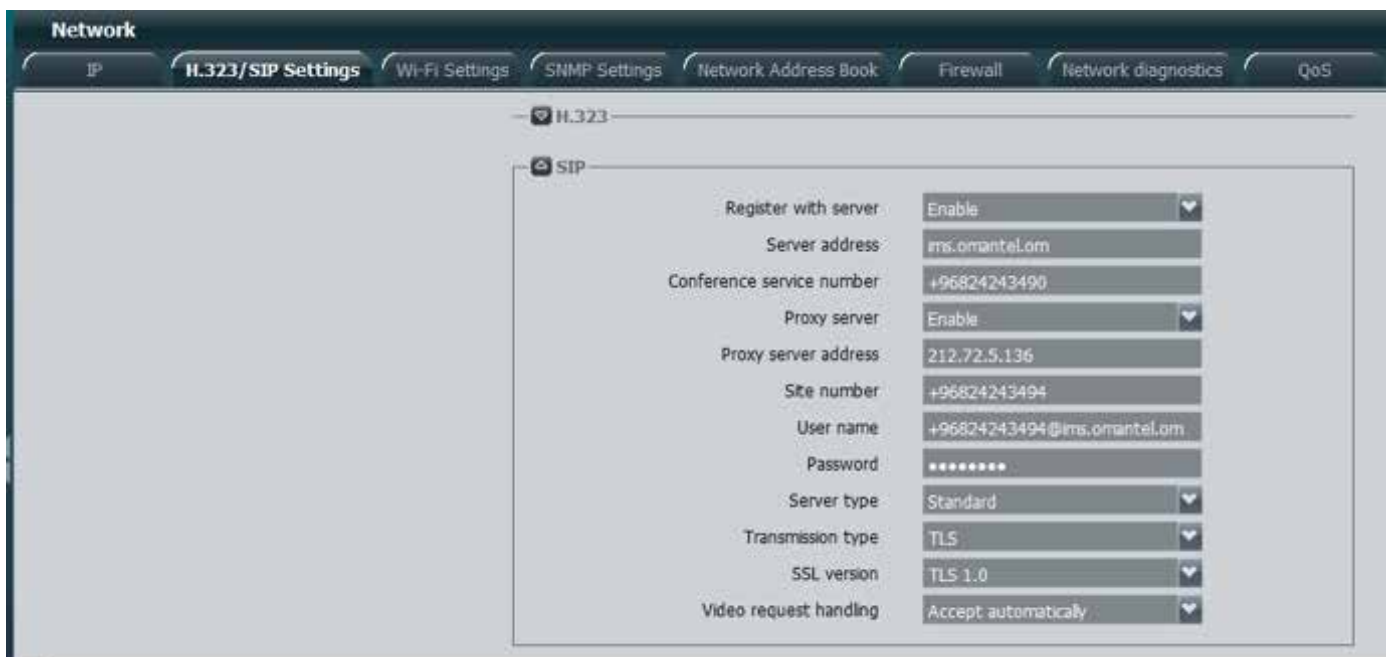
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The screenshot shows the Cisco SIP Configuration web interface. The 'SIP' tab is selected in the top navigation bar. The 'SIP Settings' section is expanded, showing fields for Mode (On), Display Name (Edge95), SIP Address (URI) (+96824243543@ims.omantel.om), Server Discovery (Manual), Server Address (212.72.5.136), Server Type (Auto), Transport (TLS), and Verify TLS (Off). The 'SIP Authentication' section is also expanded, with the 'User Name' field containing '+96824243543@ims.omantel.om' and the 'Password' field containing masked characters (*****). The 'SIP SIP NAT Traversal' section is partially visible at the bottom, with fields for ICE Mode, MNS Mode, Force TURN, TURN Server, and User Name.

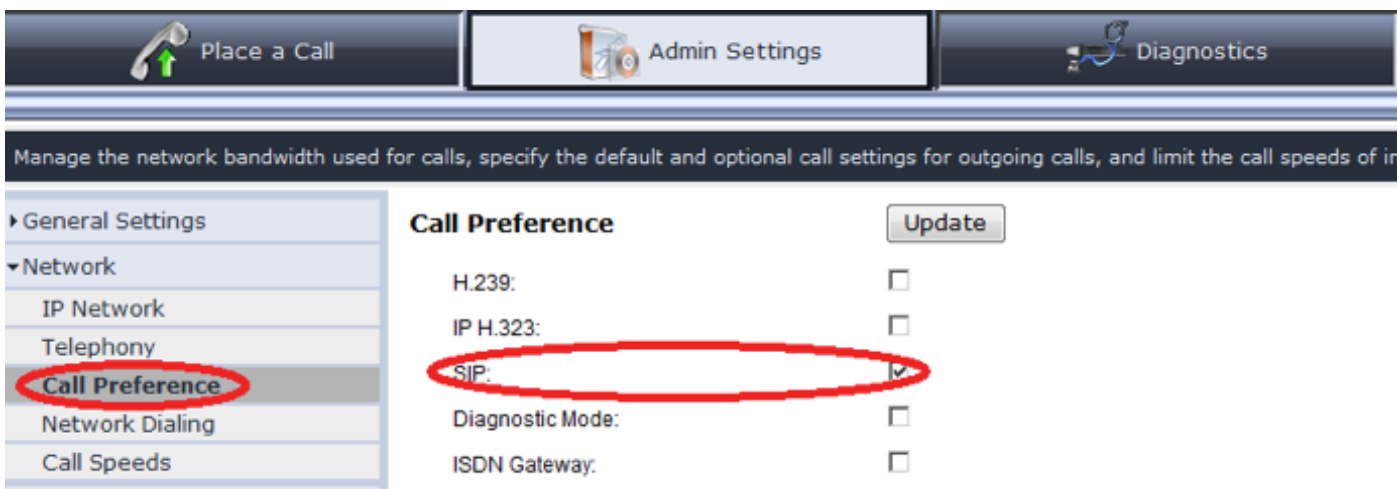
3) Huawei video terminal configuration guide



Note:

The username and password should be provided by Omantel, and the administrator should keep the password well, if the administrator forget the password, please ask Omantel for help.

3.4 Configure the call preference



Note:

If you can choose the call preference protocol, please disable other protocols, only enable the SIP protocols.

4 Network IP and Ports Requirements

There are 8 IPs for the video conference, if there is a firewall in the enterprise, please open these ports of the IPs.

| NO | Public IP | Ports |
|----|--------------|--|
| 1 | 212.72.5.136 | 443(TCP,TLS,UDP) 5060(UDP) 5061(TCP) 80(TCP,UDP) |
| 2 | 212.72.5.137 | 10001-61000(UDP) 5061(TCP) 443,80(TCP,UDP) |
| 3 | 212.72.5.138 | 443(TCP,TLS,UDP) 5060(UDP) 5061(TCP) 80(TCP,UDP) |
| 4 | 212.72.5.139 | 10001-61000(UDP) 5061(TCP) 443,80(TCP,UDP) |
| 5 | 212.72.5.147 | 443(TCP,TLS,UDP) 80(TCP,UDP) 5061(TCP) |
| 6 | 212.72.5.148 | 443(TCP,TLS,UDP) 80(TCP,UDP) 5061(TCP) |
| 7 | 212.72.5.149 | 443(TCP,TLS,UDP) 80(TCP,UDP) 5061(TCP) |
| 8 | 212.72.5.150 | 443(TCP,TLS,UDP) 80(TCP,UDP) 5061(TCP) |

5 Network QoS Requirements

5.1 Signaling QoS Requirements

Due to large packets generated by some conference services, the bearer network is required to be capable of delivering a packet of at least 1500 bytes to ensure the smooth deployment of the conference system, that is, the Maximum Transmission Unit (MTU) of the bearer network should be at least 1500 bytes. The MTU with 1500 bytes of the bearer network is the universal standard for the industry.

5.2 Media QoS Requirements of Audio Conference

The Following table describes the media QoS requirements of different audio conference experiments.

| Quality Class | Expected Result | Codec | Delay | Jitter | Unexpected Packet Loss |
|---------------|-------------------------------|-------|------------------|------------------|------------------------|
| A (clear) | The voice quality is good. | G.711 | ≤ 100 ms | ≤ 20 ms | |
| | | G.729 | Not deteriorated | Not deteriorated | |
| | | AMR | ≤ 100 ms | ≤ 20 ms | |
| | | iLBC | | | |
| B (fair) | The voice quality is average. | G.711 | ≤ 200 ms | ≤ 40 ms | |
| | | G.729 | ≤ 50 ms | ≤ 10 ms | Not deteriorated |
| | | AMR | ≤ 200 ms | ≤ 30 ms | |
| | | iLBC | | | |
| C (available) | The voice quality is poor. | G.711 | | | |
| | | G.729 | | | |
| | | AMR | | | |
| | | iLBC | | | |

5.3 Media QoS Requirements of HD Conference

The Following table describes the media QoS requirements of different HD conference experiments.

| Quality Class | Expected Result | Main Stream Image Format | Delay | Jitter | Unexpected Packet Loss | Bandwidth Without Auxiliary Streams (Media) | Bandwidth Without Auxiliary Streams (Network) |
|---------------|---|--------------------------|----------|---------|------------------------|---|---|
| A (clear) | 1. Both parties can hear each other clearly. 2. Voice and video are clear and smooth. 3. Voice and video are synchronized. | 720p@30fps | ≤ 100 ms | ≤ 30 ms | 1% | 2 Mbit/s | 2.6 Mbit/s |
| | | 720p@60fps | ≤ 100 ms | ≤ 30 ms | 1% | 2 Mbit/s | 2.6 Mbit/s |
| | | 1080p@30fps | ≤ 100 ms | ≤ 30 ms | 1% | 3 Mbit/s | 3.6 Mbit/s |
| | | 1080p@60fps | ≤ 100 ms | ≤ 30 ms | 1% | 6 Mbit/s | 7.2 Mbit/s |
| B (fair) | 1. Subscribers can perceive delay in some calls but the delay does not affect the communication between two parties. 2. Video is smooth and clear. Voice is clear. 3. Voice and video are synchronized. | 720p@30fps | ≤ 150 ms | ≤ 50 ms | %5 | 1 Mbit/s | 1.3 Mbit/s |
| | | 720p@60fps | ≤ 150 ms | ≤ 50 ms | %5 | 1.5 Mbit/s | 2 Mbit/s |
| | | 1080p@30fps | ≤ 150 ms | ≤ 50 ms | %5 | 2.5 Mbit/s | 3 Mbit/s |
| | | 1080p@60fps | ≤ 150 ms | ≤ 50 ms | %5 | 4 Mbit/s | 4.8 Mbit/s |
| C (available) | 1. Subscribers perceive obvious delay, which may slightly affect the communication between two parties. 2. Video is not clear and smooth enough, which does not affect the communication. 3. Voice quality is not affected. 4. Voice and video are synchronized. | 720p@30fps | ≤ 200 ms | ≤ 50 ms | %10 | 1 Mbit/s | 1.3 Mbit/s |
| | | 720p@60fps | ≤ 200 ms | ≤ 50 ms | %10 | 1.5 Mbit/s | 2 Mbit/s |
| | | 1080p@30fps | ≤ 200 ms | ≤ 50 ms | %10 | 2.5 Mbit/s | 3 Mbit/s |
| | | 1080p@60fps | ≤ 200 ms | ≤ 50 ms | %10 | 3 Mbit/s | 3.6 Mbit/s |

Notes:

1. This above data is based on the scenario that TEX0 series HD conference terminal attend the SD conference with supporting SEC3.0 function.