



# Grandstream Networks, Inc.

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UCM63xx Series

## **Remote Connect EndPoint Configuration Guide**



## Table of Contents

<b>INTRODUCTION.....</b>	<b>3</b>
<b>PREREQUISITES .....</b>	<b>4</b>
<b>CONFIGURE IP PHONES VIA GDMS.....</b>	<b>5</b>
<b>CONFIGURE UCM REMOTECONNECT SERVICE FOR IP PHONES .....</b>	<b>7</b>
<b>MAKE CALLS USING IP PHONES .....</b>	<b>10</b>

## Table of figures

Figure 1: UCM RemoteConnect Plan Information .....	4
Figure 2: Add New SIP Server on GDMS .....	5
Figure 3: Batch Configure SIP Server for SIP Accounts .....	6
Figure 4: Assign Device to SIP Account.....	6
Figure 5: GXV3370 Account Configuration Page .....	7
Figure 6: GXV3370 Account →SIP Settings .....	8
Figure 7: GXV3370 Phone Settings →General Settings .....	8
Figure 8: GXV3370 TLS Configuration .....	9



## INTRODUCTION

Thank you for purchasing the UCM6301/UCM6302/UCM6304/UCM6308 IP PBX. The Grandstream UCM6300 series IP PBX is based on Asterisk 16 system. It provides powerful functions, friendly interface for remote management and easy-to-expand all-in-one communication solution to enterprises of all sizes. The UCM6300 series IP PBX supports up to 3000 extensions with PBX features including audio/video calling, video conferencing, video surveillance, PBX data management and analysis, UCM RemoteConnect, and device remote access. It is an ideal choice for enterprises looking for an all-in-one solution for users to communicate efficiently and work productively.

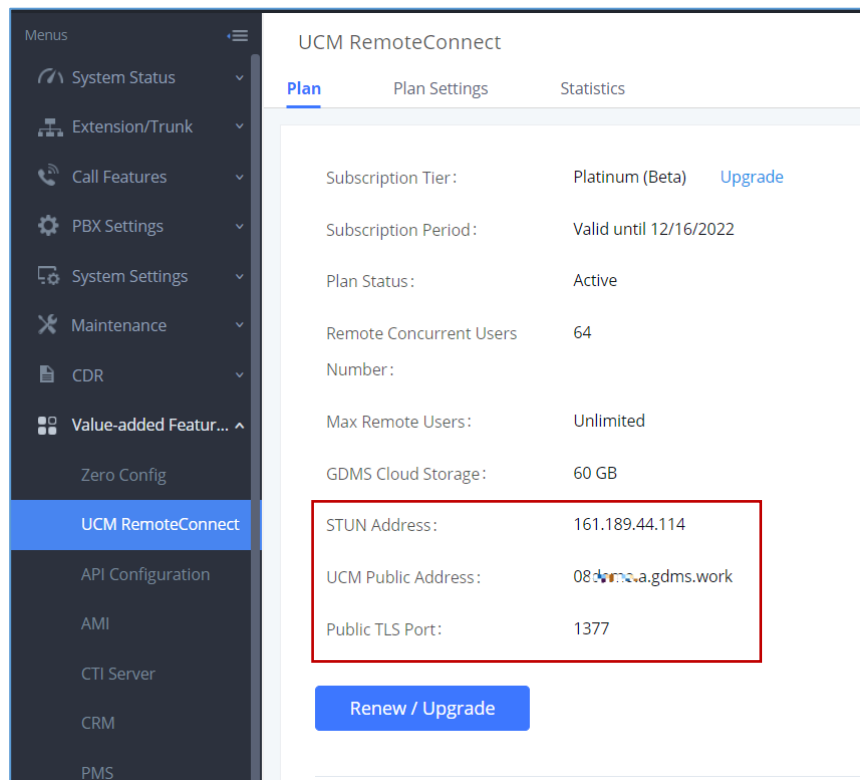
The UCM6300 series IP PBX provides UCM RemoteConnect service which offers users a quick setup to start working remotely including GS Wave web app using WebRTC and Wave mobile app on Android and IOS system to communicate and join meetings, sync up and manage extension, receive alerts and reports, view and manage storage via cloud, and much more. The UCM6300 UCM RemoteConnect service is offered via Grandstream Device Management System (GDMS). Please visit GDMS platform for UCM RemoteConnect service plan information and purchasing plan, device remote management, cloud storage management and etc.

This document describes how to configure end users' IP phones to register to UCM6300 series IP PBX UCM using RemoteConnect service. With RemoteConnect service, IP phones behind NAT can register to UCM6300 series and communicate with other devices without additional settings in your network.



## PREREQUISITES

The UCM RemoteConnect service on UCM6300 series must be used with Grandstream Device Management System (GDMS). After the UCM is connected with GDMS, the RemoteConnect information displays as below on UCM6300 web **GUI→Value-added Features→UCM RemoteConnect** page. In this page, STUN Address, UCM Public Address and Public TLS Port information are needed for IP phone to register to UCM6300 series.



**Figure 1: UCM RemoteConnect Plan Information**

Users can configure the IP phone to register to UCM6300 series and manage the IP phone remotely via GDMS, or directly configure the account information on IP phone manually.

## CONFIGURE IP PHONES VIA GDMS

The GDMS admin can configure IP phone remotely with the settings required for RemoteConnect. To do so please follow the steps below:

**Step 1:** In your web browser, open the GDMS address and log in with your GDMS account:  
<http://www.gdms.cloud/login>

**Step 2:** After logging in GDMS as admin, navigate to **VoIP Account→SIP Server** page. Click on “Add Server” and configure the following:

- **Server name:** enter the server name for identification purpose.
- **SIP Server:** enter the UCM Public Address: Public TLS Port.
- **NAT Traversal:** STUN
- Other settings are optional and can be configured as needed.

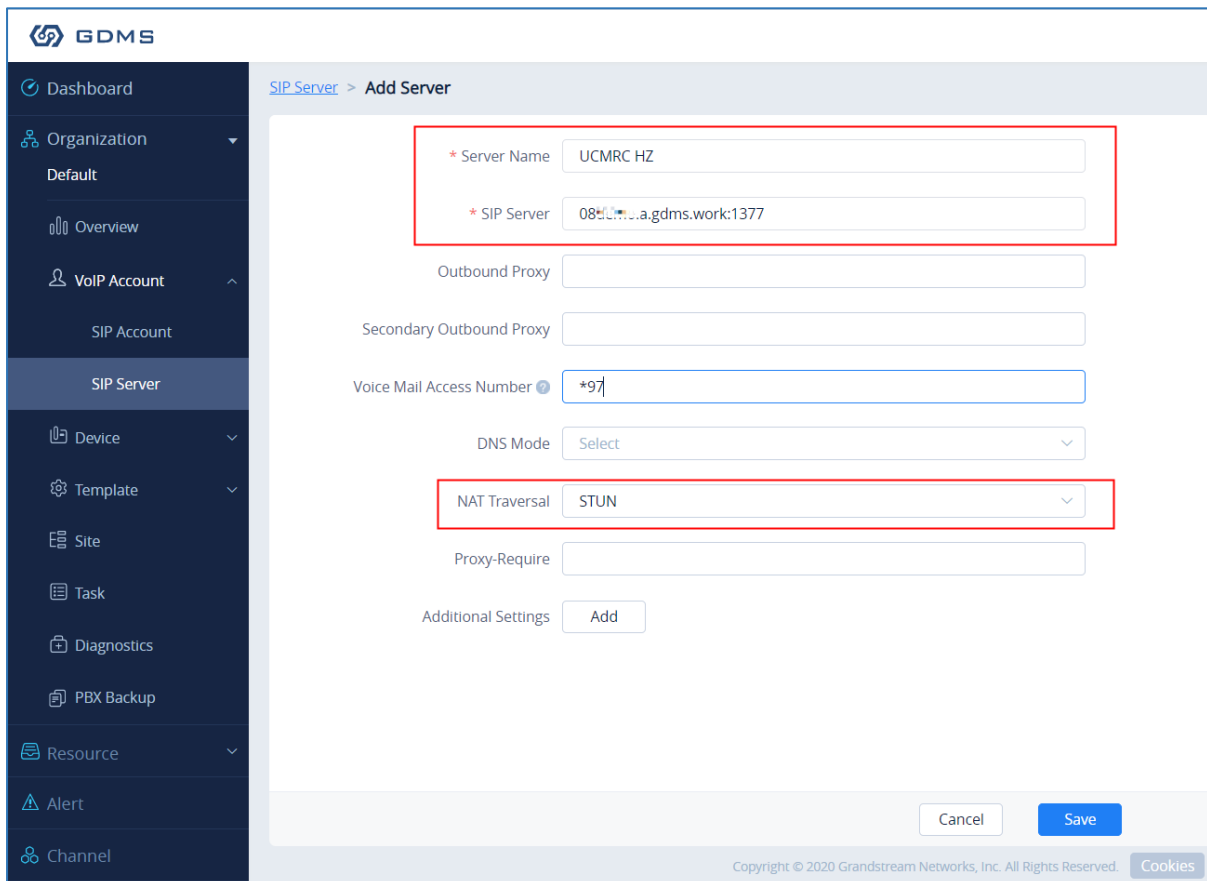
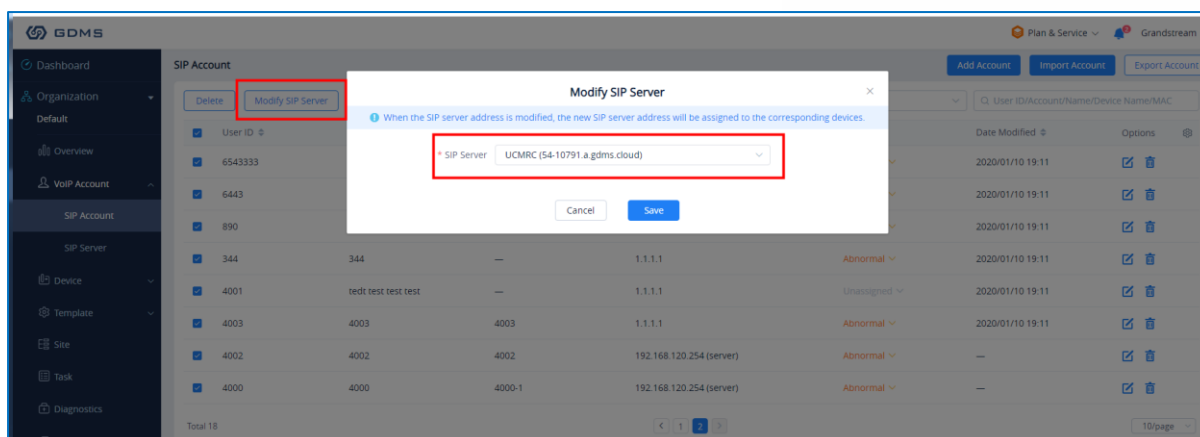


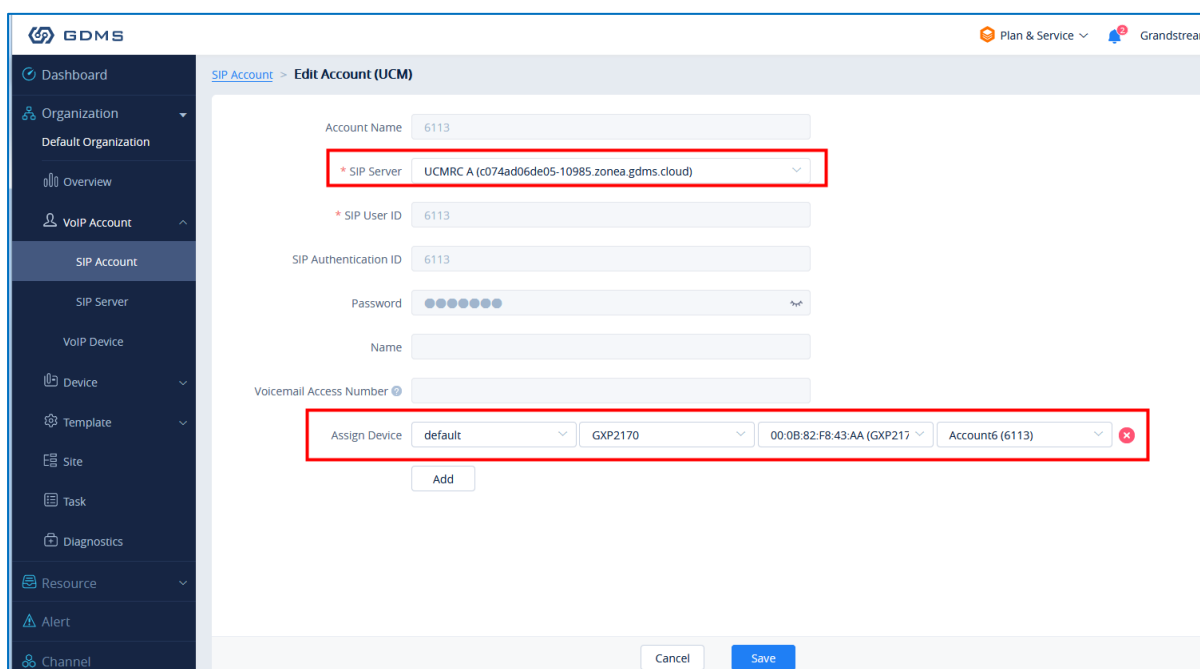
Figure 2: Add New SIP Server on GDMS

**Step 3:** Navigate to **VoIP Account→SIP Account**, select the accounts that need to use RemoteConnect Public UCM address as SIP server, then click on “Modify SIP Server”. Select the newly created SIP server address in step 2 to be used here. Click on “Save”.



**Figure 3: Batch Configure SIP Server for SIP Accounts**

**Step 4:** Nagivate to **VoIP Account->SIP Account**, assign the account which has its SIP server as the UCM public address to the device. If the account has device assigned before, GDMS will send the update account information to the assigned device.



**Figure 4: Assign Device to SIP Account**

**Note:**

1. If the IP phone device is currently not supported by GDMS, this device cannot be configured and managed by GDMS.
2. After GDMS assigns SIP account to the IP phone device, if the device cannot register or experiences call issues, please check device configuration in below section [**CONFIGURE UCM REMOTECONNECT SERVICE FOR IP PHONES**] to see it's configured properly.



## CONFIGURE UCM REMOTECONNECT SERVICE FOR IP PHONES

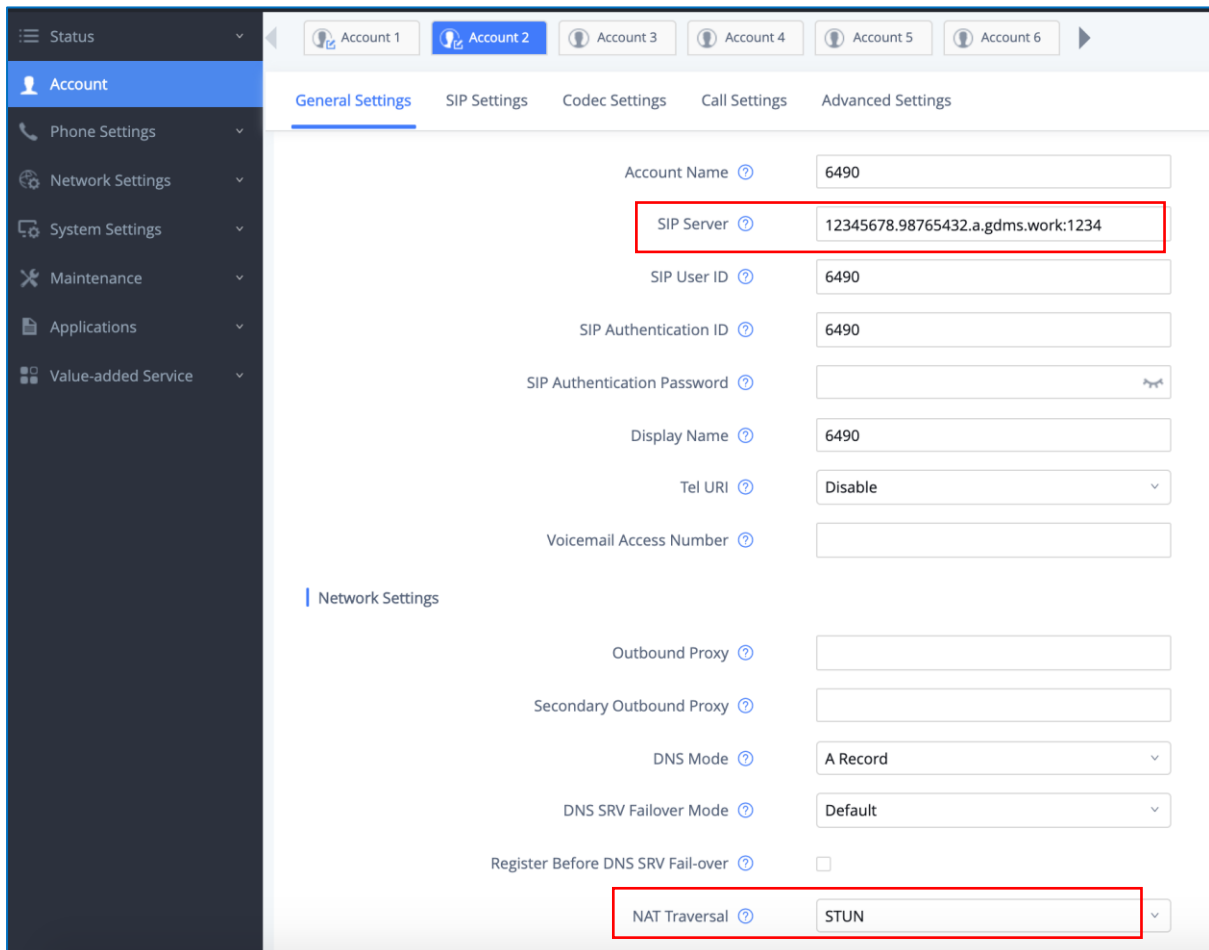
IP phone end devices behind NAT in external network can register to the UCM6300 series for remote work purpose.

Please refer to below configuration example on GXV3370.

1. Log in GXV3370 web UI as admin, navigate to **Account→Basic Settings** page and configure the following:

**SIP Server:** Enter the UCM Public Address: Public TLS Port. This information can be found under UCM **web UI→ Value-added Features→UCM RemoteConnect→Plan** page.

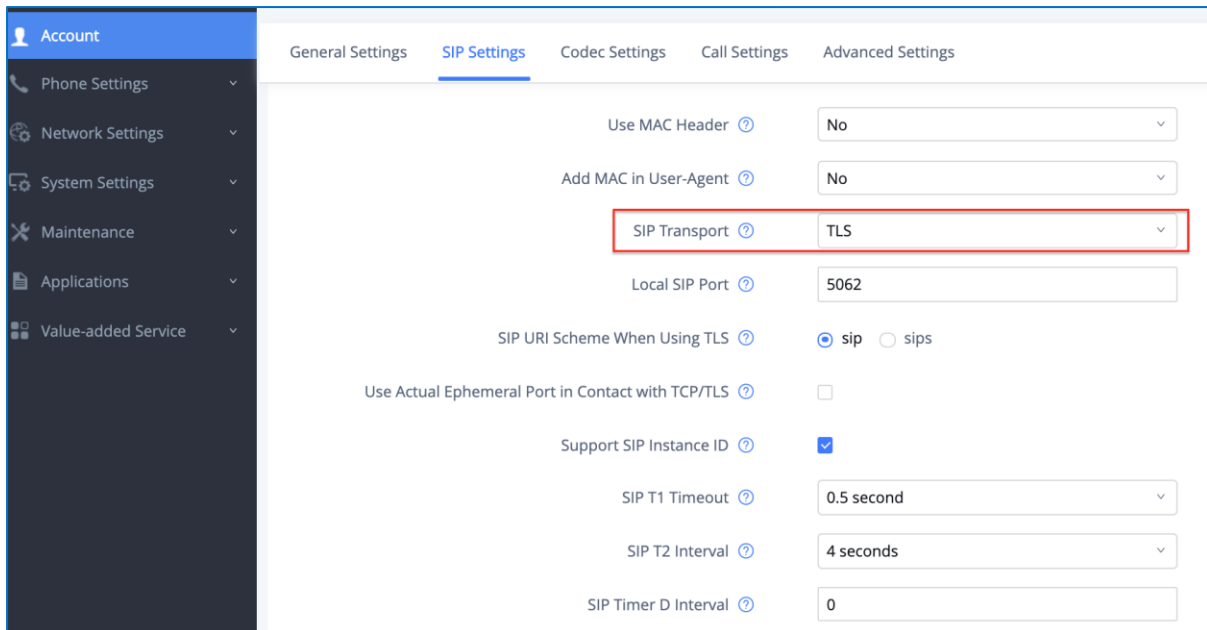
**NAT Traversal:** STUN



The screenshot displays the 'Account Configuration' page for 'Account 2' in the GXV3370 web UI. The 'General Settings' tab is active. The 'SIP Server' field is highlighted with a red box and contains the value '12345678.98765432.a.gdms.work:1234'. The 'NAT Traversal' field is also highlighted with a red box and is set to 'STUN'. Other fields include 'Account Name' (6490), 'SIP User ID' (6490), 'SIP Authentication ID' (6490), 'SIP Authentication Password' (empty), 'Display Name' (6490), 'Tel URI' (Disable), 'Voicemail Access Number' (empty), 'Outbound Proxy' (empty), 'Secondary Outbound Proxy' (empty), 'DNS Mode' (A Record), 'DNS SRV Failover Mode' (Default), and 'Register Before DNS SRV Fail-over' (unchecked).

Figure 5: GXV3370 Account Configuration Page

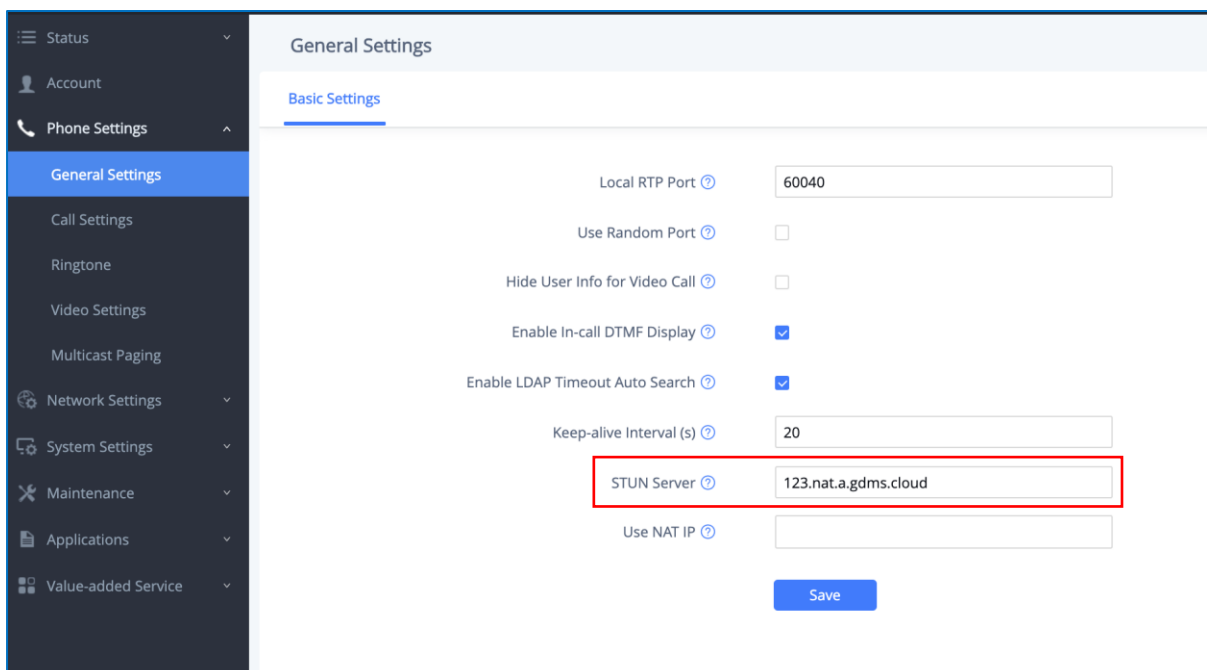
2. Go to **Account→SIP Settings** and configure SIP transport to “TLS”.



The screenshot shows the 'Account' settings page for a GXV3370 device. The 'SIP Settings' tab is selected. The 'SIP Transport' dropdown menu is highlighted with a red box and set to 'TLS'. Other settings include 'Use MAC Header' (No), 'Add MAC in User-Agent' (No), 'Local SIP Port' (5062), 'SIP URI Scheme When Using TLS' (sip), 'Use Actual Ephemeral Port in Contact with TCP/TLS' (unchecked), 'Support SIP Instance ID' (checked), 'SIP T1 Timeout' (0.5 second), 'SIP T2 Interval' (4 seconds), and 'SIP Timer D Interval' (0).

**Figure 6: GXV3370 Account→SIP Settings**

- Go to phone's Web UI→**Phone Settings**→**General Settings**, configure the STUN server to be same as the one under UCM Web UI→ **Value-added Features**→**UCM RemoteConnect**→**Plan** page.



The screenshot shows the 'Phone Settings' page for a GXV3370 device. The 'General Settings' tab is selected. The 'STUN Server' text input field is highlighted with a red box and contains the value '123.nat.a.gdms.cloud'. Other settings include 'Local RTP Port' (60040), 'Use Random Port' (unchecked), 'Hide User Info for Video Call' (unchecked), 'Enable In-call DTMF Display' (checked), 'Enable LDAP Timeout Auto Search' (checked), 'Keep-alive Interval (s)' (20), and 'Use NAT IP' (empty). A 'Save' button is at the bottom right.

**Figure 7: GXV3370 Phone Settings→General Settings**

- Go to the phone's web UI→**System Settings**→**Security Setting**→**TLS** page, configure "Minimum TLS Version" and "Maximum TLS Version" to be 1.2



- Status
- Account
- Phone Settings
- Network Settings
- System Settings
  - Time and Language
  - Security Settings**
  - Preferences
  - TR069
- Maintenance
- Applications
- Value-added Service

### Security Settings

- Web/SSH Access
- User Info Management
- TLS**
- Certificate Management

Minimum TLS Version ? 1.2

Maximum TLS Version ? 1.2

Enable Weak TLS Cipher Suites ? Disable Symmetric Encryption RC4/DES/3DES

SIP TLS

SIP TLS Certificate ?

SIP TLS Private Key ?

SIP TLS Private Key Password ?

Save

**Figure 8: GXV3370 TLS Configuration**

## MAKE CALLS USING IP PHONES

After configuring the IP phones with UCM RemoteConnect service, users can use the phone to make audio/video calls and join GS Wave audio/video conferences.



**Note:** Presentation on end device IP Phones is currently not supported.

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Below are the Grandstream devices that supports RemoteConnect services with UCM6300 series:

- GXV3350/GXV3370/GXV3380
- GXP series
- GRP series
- WP820
- DP750
- GVC series.

