



Grandstream Networks, Inc.

UCM630X Series IP PBX

Remote Work Environment Setup Guide

Table of Contents

OVERVIEW.....	4
SIP SIGNALING TRAVERSAL	5
MEDIA TRAVERSAL	9
CONFIGURING FOR WAVE	12



Table of Figures

Figure 1: SIP Signaling Penetration - UDP Port	5
Figure 2: SIP Signaling Penetration - TCP/TLS Port	6
Figure 3: SIP Signaling Penetration - NAT Settings.....	7
Figure 4: RTP Settings – RTP/BFCP Ports.....	9
Figure 5: SIP Settings - STUN/TURN Server	10
Figure 6: HTTP Server - Wave Settings.....	12

Table of Tables

Table 1: NAT Settings.....	7
Table 2: RTP Settings.....	10
Table 3: Wave Settings.....	12



OVERVIEW

Thank you for purchasing the UCM630X Series IPPBX. The Grandstream UCM6300 Series IPPBX is based on Asterisk 16 and provides powerful features, a user-friendly interface for remote management, and a scalable all-in-one communication solution for enterprises of all sizes. Offering up to 3000 extensions, PBX features such as audio/video calling and conferencing, video surveillance support, PBX data management and statistics, and remote device access and management via UCM RemoteConnect, it is the perfect solution to improve communication efficiency and productivity.

The UCM630X Series IPPBX offers UCM RemoteConnect, a service which allows users to quickly set up remote communication and management using Grandstream Wave. Grandstream Wave is a multi-platform application available on web browsers, Windows, macOS, Android, and iOS that supports calling, conferencing, extension management, system alert event monitoring, call/meeting reports, cloud storage access, and more.

UCM RemoteConnect is highly recommended for easy-to-use, secure, and stable network traversal. If UCM RemoteConnect is not used, users can use this guide to set up their remote work environments with the UCM630X. This guide has three sections:

- SIP Signaling Traversal
- Media Traversal
- Configuring for Wave



SIP SIGNALING TRAVERSAL

To allow successful SIP registration and call establishment, users will need to configure SIP signaling NAT traversal by associating their private local network addresses and ports with their public addresses and ports. To do so, please see the following steps:

Step 1: From the UCM630X web UI, users can retrieve or modify their internal UDP port from the PBX Settings -> SIP Settings -> General page. In the same section, the TCP/TLS ports can be found in the *TCP/TLS* tab. Please see the figures below:

SIP Settings

General Session Timer TCP/TLS NAT ToS STIR/SHAKEN Misc

* Realm for Digest Authentication:	<input type="text" value="grandstream"/>
* Bind UDP Port:	<input type="text" value="5060"/>
* Bind IPv4 address:	<input type="text" value="0.0.0.0"/>
* Bind IPv6 address:	<input]"="" type="text" value=":::"/>
Allow Transfer:	<input checked="" type="checkbox"/>
MWI From Header:	<input type="text"/>
Enable Diversion Header:	<input checked="" type="checkbox"/>
Block Collect Calls:	<input type="checkbox"/>

Figure 1: SIP Signaling Penetration - UDP Port



SIP Settings

General Session Timer **TCP/TLS** NAT ToS STIR/SHAKEN Misc

TCP Enable:

TCP Bind IPv4 Address:

TCP Bind IPv6 Address:

TLS Enable:

TLS Bind IPv4 Address:

TLS Bind IPv6 Address:

TLS Do Not Verify:

TLS Self-signed CA:

TLS Cert:

TLS Key:

TLS CA Cert:

TLS CA list: No CA file.

Figure 2: SIP Signaling Penetration - TCP/TLS Port

Step 2: Once the internal port information has been confirmed, users can then map them to the corresponding external ports on the router.

Example: If the internal UDP port is 5060, users would map port 5060 to whatever is designated as the external UDP port for SIP traffic on the router. Please contact your enterprise's IT administrator for this information.

Note:

1. For security purposes, it is not recommended to use an external port number that is the same as the internal port number.
2. If the router supports SIP ALG, it will need to be disabled.

Step 3: Configure NAT settings as shown in the figure below:



SIP Settings

General Session Timer TCP/TLS **NAT** ToS STIR/SHAKEN Misc

If Local Network Address is not configured, External Host will not take effect.

External Host:

Use IP address in SDP:

Get External IP via STUN:

* External UDP Port:

* External TCP Port:

* External TLS Port:

Local Network Address: /

Figure 3: SIP Signaling Penetration - NAT Settings

- Configure *External Host* with the public network address of the UCM630X.
- Configure the *External UDP Port*, *External TCP Port*, and/or *External TLS Port* fields based on the router mappings created in the previous step.
- Configure *Local Network Address* with the UCM630X's local network subnets.

Table 1: NAT Settings

NAT Settings	
External Host	Configure the IP address or domain that would be used in outbound SIP messages if the UCM is behind a NAT. Typically, this would be the public network address.
Use IP Address in SDP	If enabled, the SDP connection will use the IP address resolved from the external host.
Get External IP via STUN	If enabled, the external IP address will be retrieved via STUN. Please confirm the configured STUN server is working properly.
External UDP Port	Configure the externally mapped UDP port when the UCM is behind a static NAT or PAT.
External TCP Port	Configure the externally mapped TCP port when the UCM is behind a static NAT or PAT.



External TLS Port	Configure the externally mapped TLS port when the UCM is behind a static NAT or PAT. The default setting is 5061.
Local Network Address	Configure a list of subnets under the internal network. Multiple entries are supported. This must be configured for successful NAT traversal. Example: 192.168.0.0/16

If properly configured, users can then register their extensions to the UCM630X's public network address and port. If one-way audio issues are encountered, please confirm your NAT settings (SIP signaling traversal) and SIP/RTP ports (media traversal) on the UCM and/or router. For details about media traversal configuration, please see the Media Traversal section below.



MEDIA TRAVERSAL

To ensure correct audio and video exchange between call parties on different networks, proper media traversal configurations are required. For IP endpoints, users will need to configure RTP and BFCP port mappings. For Wave, users will need to configure STUN and TURN server settings instead. Please see the following steps:

Step 1: From the UCM630X Web UI, navigate to the *PBX Settings* -> *RTP Settings* page as shown in the figure below:

RTP Settings

RTP Settings Payload Type Settings

* RTP Start:	10000
* RTP End:	20000
Strict RTP:	<input type="checkbox"/>
RTP Checksums:	<input type="checkbox"/>
ICE Support:	<input checked="" type="checkbox"/>
STUN Server:	
* BFCP UDP Start:	50000
* BFCP UDP End:	52999
* BFCP TCP Start:	53000
* BFCP TCP End:	55999
TURN Server:	
TURN Server Name:	
TURN Server Password:	
Connection Protocol:	UDP
Number of ICE Candidates:	4

Figure 4: RTP Settings – RTP/BFCP Ports

Step 2 (for IP endpoints): Confirm the starting and ending ports for RTP and BFCP.



- For proper audio/video exchange, create an RTP port range mapping on your router. From the figure above, this would be 10000-20000.
- For proper presentation sharing, create a port range mapping on your router. From the figure above, this would be 53000-55999.

Step 3 (for Wave): Configure the STUN and TURN server settings as highlighted in the figure below. Without UCM RemoteConnect, users would need to confirm their own STUN/TURN server information.

RTP Settings

RTP Settings Payload Type Settings

* RTP Start:	10000
* RTP End:	20000
Strict RTP:	<input type="checkbox"/>
RTP Checksums:	<input type="checkbox"/>
ICE Support:	<input checked="" type="checkbox"/>
STUN Server:	<input type="text"/>
* BFCP UDP Start:	50000
* BFCP UDP End:	52999
* BFCP TCP Start:	53000
* BFCP TCP End:	55999
TURN Server:	<input type="text"/>
TURN Server Name:	<input type="text"/>
TURN Server Password:	<input type="text"/>
Connection Protocol:	UDP ▼
Number of ICE Candidates:	4 ▼

Figure 5: SIP Settings - STUN/TURN Server

Table 2: RTP Settings

RTP Settings	
RTP Start	Configure the starting RTP port. Default is 10000.



RTP End	Configure the ending RTP port. Default is 20000.
Strict RTP	Enable strict RTP protection. If enabled, RTP packets that do not originate from the RTP stream source will be dropped. Disabled by default.
RTP Checksums	Toggles UDP checksums on RTP traffic. Disabled by default.
ICE Support	Configure ICE support. ICE is the integrated use of STUN and TURN structures to provide a reliable VoIP or media transmission via a SIP request/response model for multiple candidate endpoints. These endpoints exchange IP addresses and ports such as private addresses and TURN server addresses. Enabled by default.
STUN Server	<p>Configure the STUN server address. STUN protocol is both a client-server protocol and a request-response protocol that is used to check the connectivity between two endpoints and maintain a NAT-binding Keep-Alive agreement between entries.</p> <p>If blank, STUN monitoring is disabled. Default is blank.</p> <p>Supported format: [hostname/IP address]: [port]</p> <p>If the port number is not specified, 3478 will be used by default.</p>
BFCP UDP Start	Configure the starting BFCP UDP port. Default is 50000.
BFCP UDP End	Configure the ending BFCP UDP port. Default is 52999.
BFCP TCP Start	Configure the starting BFCP TCP port. Default is 53000.
BFCP TCP End	Configure the ending BFCP TCP port. Default is 55999.
TURN Server	Configure the TURN server address. TURN is an enhanced version of the STUN protocol and is dedicated to processing symmetric NAT problems.
TURN Server Name	Configure the TURN server account name.
TURN Server Password	Configure the TURN server account password.
Connection Protocol	Protocol used to connect to the TURN server.
Number of ICE Candidates	Configure the number of ICE candidates to gather and send to remote peers. Increasing this value slightly increases network traffic to and from the TURN server. High network traffic to the TURN server may negatively affect call quality.



CONFIGURING FOR WAVE

To successfully log in and use Wave, please see the following steps:

Step 1: From the UCM630X Web UI, navigate to the *System Settings* -> *HTTP Server* -> *Wave Settings* section to configure *External Host* with the UCM630X's public network address and retrieve/modify the *Wave Port* information.

Step 2: Map the internal Wave port to an external port number on your router.

HTTP Server

UCM Web Settings

Redirect from Port 80:

External Host:

* Port:

Enable IP Address Whitelist:

Permitted IP (s): / Add IP Address +

Wave Settings

External Host:

* Port:

Certificate Settings

Options:

TLS Private Key:

TLS Cert:

Figure 6: HTTP Server - Wave Settings

If properly configured, users should now be able to access the Wave login page via the public network address and external port.

Table 3: Wave Settings

Wave Settings	
External Host	Configure the URL and port (optional) used to access the Wave portal or UCM meeting rooms if the UCM is behind NAT. If UCM RemoteConnect is used, this field will be configured automatically.



Port	The port to access the UCM's Wave portal. If behind NAT, please confirm that a mapping to this port has been configured on the router.
-------------	--

