

Grandstream Networks, Inc.

UCM62xx/UCM6510 Series

GS Wave WebRTC Video Calling & Conferencing Guide



Table of Contents

INTRODUCTION.....	4
SPECIFICATIONS	5
WEBRTC CONFIGURATION.....	6
VIDEO CONFERENCE CONFIGURATION	7
Basic Settings	7
Video Conference room Configurations	8
Schedule a Conference	9
LOGIN TO GS WAVE PORTAL	13
WEBRTC FEATURES	14
Initiate a Call	14
Transfer	17
<i>Blind Transfer</i>	17
<i>Attended Transfer</i>	18
Screen Sharing	18
Chat	20
VIDEO CONFERENCE FEATURES	22
Join a Video Conference Room	22
Host Privileges	22



Table of figures

Figure 1: Grandstream Wave	4
Figure 2: Enable WebRTC	6
Figure 3: WebRTC Support for Extension	6
Figure 4: Video Conference Basic settings	7
Figure 5: Created Conference Room	9
Figure 6: Video Conference Schedule	9
Figure 7: Video Conference Schedule	11
Figure 8: Invitation Link Schedule	12
Figure 9: GS Wave Portal	13
Figure 10: Contacts Page	13
Figure 11: Initiating a Call	14
Figure 12: Contact Search	15
Figure 13: Recent Calls search	15
Figure 14: Audio call screen	16
Figure 15: Video Call Screen	16
Figure 16: Enable Transfer	17
Figure 17 : Transfer Screen – Blind Transfer	17
Figure 18 : Transfer Screen – Attended Transfer	18
Figure 19: Screen Share Type	18
Figure 20: Plugin Setup	19
Figure 21: Enable Presentation on GXV32xx	19
Figure 22: Enable Presentation on GVC32xx	20
Figure 23: Chat Feature	20
Figure 24: Chat message destination	21
Figure 25: Video Conference screen	22
Figure 26: Host Privileges	22

Table of figures

Table 1: Video Conference Basic Settings	7
Table 2: Video Conference Room Configuration Parameters	8
Table 3 : Video Conference Schedule Parameters	9



INTRODUCTION

The UCM62xx and UCM65xx support HTTP & WebSocket for web browser to register to the UCM and establish calls and participate in web video conferences with other endpoints in real time via WebRTC. With the UCM you can easily create, schedule, manage, and join video conference calls, from your desktop or laptop computer.

UCM Video conferencing uses WebRTC technology, so all the participants don't have to download and install any additional software or plugins.

This document introduces the user portal features and offers step by step instructions to use them.

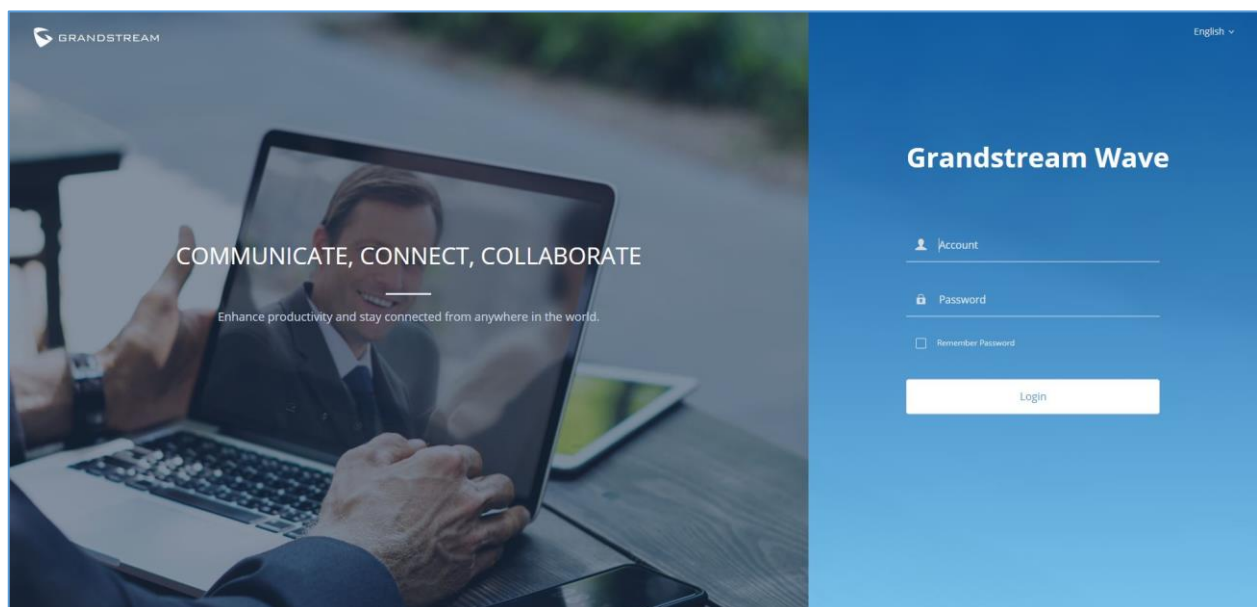


Figure 1: Grandstream Wave

Notes:

- Video conferencing can be resource-intensive and may cause performance issues with the UCM when used.
 - To ensure the best experience, please use Google Chrome (v67 or higher) or Mozilla Firefox (v60).
 - The same IP address cannot log into both the admin/user portal and the GS Wave WebRTC portal at the same time.
 - Concurrent registration limit is 300 for UCM62xx and 500 for UCM6510 starting from firmware 1.0.19.27
-



SPECIFICATIONS

- Up to 4 video feeds in a conference call (one 1080p, others being QVGA) and one screen share feed.
- Up to 300 WebRTC users for UCM62xx and 500 users for UCM6510 that can be logged into the Grandstream Wave page at the same time.
- Participant limit:
 - UCM62xx: **8**
 - UCM6510: **15**
 - **Warning:** A video conference with 4 ongoing video feeds will result in 75-83% CPU usage. UCM62xx and UCM6510 will be able to handle 15 and 30 more concurrent audio calls respectively.
- Audio: PCMU, PCMA, GSM, iLBC, G722.1, G722.1C, G729, G723.1, G726, OPUS
- Video: H264, VP8
- Framerate: 15 FPS video stream, 5 FPS screen share
- Firefox v61 issue: Caller cannot hear the callee when Firefox is used to answer the call.



WEBRTC CONFIGURATION

Web audio and video calls and conferencing can now be achieved through the UCM's new WebRTC page.

UCM Video Conferencing must be enabled by the administrator for the concerned extensions under the extensions level by following below steps:

- 1) Navigate to **Value → Added Features → WebRTC** and enable WebRTC support.

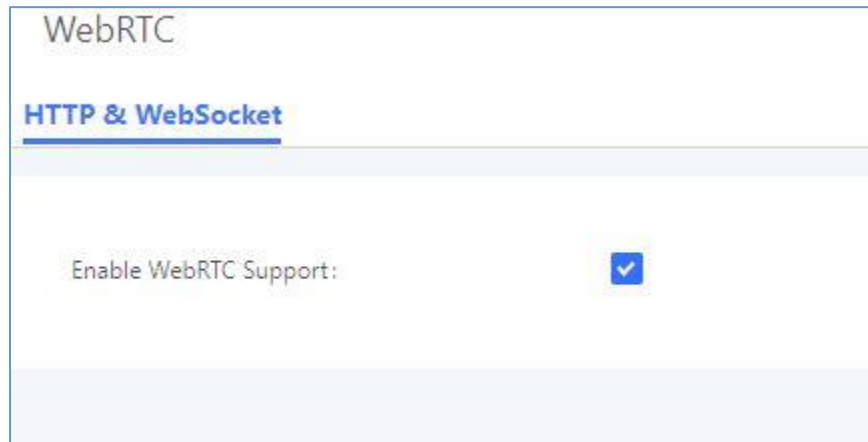


Figure 2: Enable WebRTC

- 2) Select the extensions that would use WebRTC and enable WebRTC support on them under Features section.

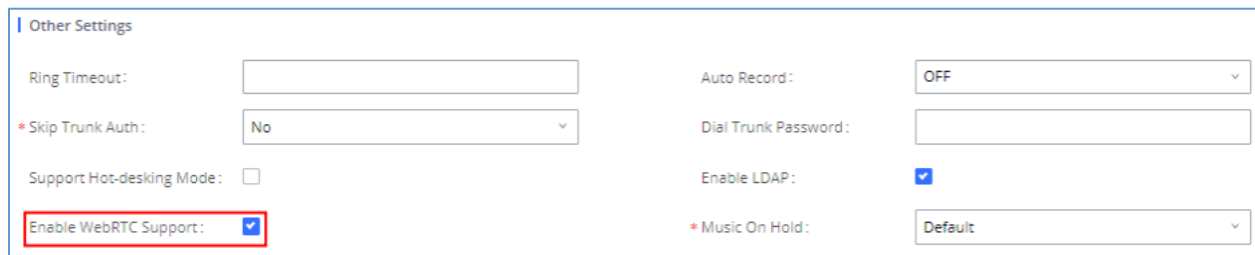


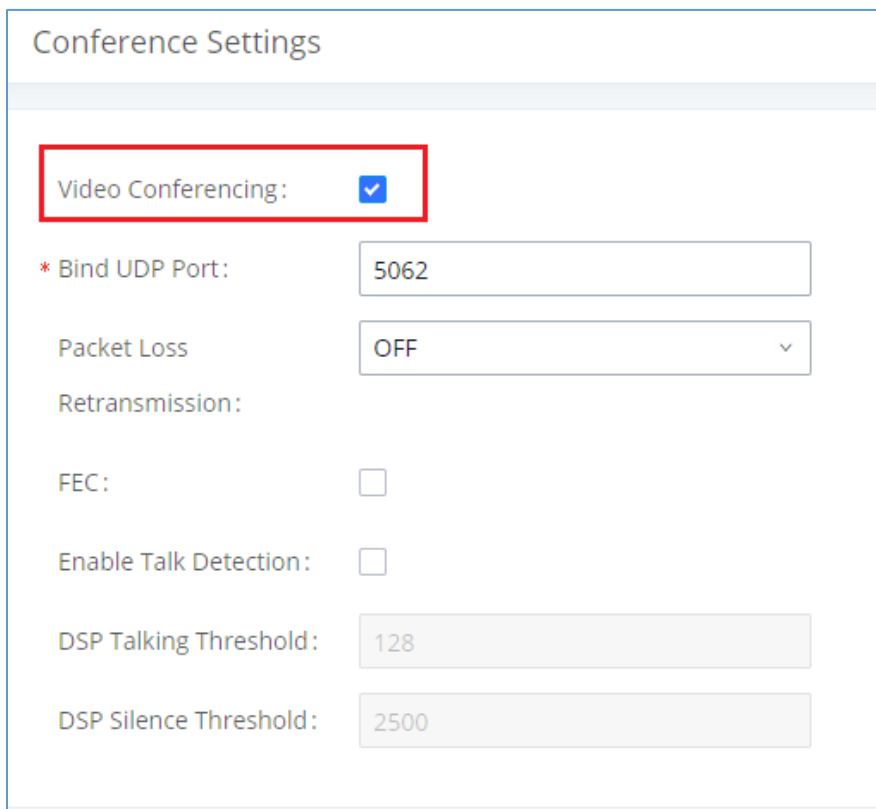
Figure 3: WebRTC Support for Extension



VIDEO CONFERENCE CONFIGURATION

The video conference configurations can be accessed under **Web GUI→Call Features→Video Conference**. In this page, users could enable, set the Basic setting, create, edit, view, manage, delete conference rooms and edit the Conference Schedule.

Basic Settings



Conference Settings

Video Conferencing: ☒

* Bind UDP Port: 5062

Packet Loss: OFF

Retransmission: OFF

FEC: ☐

Enable Talk Detection: ☐

DSP Talking Threshold: 128

DSP Silence Threshold: 2500

Figure 4: Video Conference Basic settings

Table 1: Video Conference Basic Settings

Basic Settings	
Video Conferencing	This option should be enabled to activate the Video Conference feature.
Bind UDP Port	Configure the UDP port number for MCM. The standard UDP port for MCM is 5062.
Packet Loss Retransmission	This option offers whether to enable Packet Loss Retransmission or not. It can be set to NACK, NACK+RTX or OFF. The default setting is "OFF".
FEC	If enabled, the Forward Error Correction (FEC) will be activated. The default setting is "No".



Enable Talk Detection	If enabled, the AMI will send the corresponding event when a user starts or stop talking.
DSP Talking Threshold	The amount of time(ms) that sound exceeds what the DSP has established as the baseline for silence before a user is considered to be talking. This value affects several operations and should not be changed unless the impact on call quality is fully understood.
DSP Silence Threshold	The amount of time(ms) that sound falls within what the DSP has established as the baseline for silence before a user is considered be silent. This value affects several operations and should not be changed unless the impact on call quality is fully understood.

Video Conference room Configurations

Click on the Video Conference tab and create a new video conference room. In this tab, you can:



- Click on "Create New Conference Room" to add a new conference room.
- Click on  to edit the conference room.
- Click on  to delete the conference room.

Table 2: Video Conference Room Configuration Parameters

Extension	Configure the conference number for the users to dial into the conference. Note: Up to 64 characters.
Password	When configured, the users who would like to join the conference call must enter this password before accessing the conference room. Notes: <ul style="list-style-type: none"> • Only digits are allowed. • The password must be at least 4 characters. All repetitive and sequential digits (e.g., 0000, 1111, 1234 and 2345) or common digits (e.g., 111222 and 321321) are not allowed.

When configured, the Conference Room number is the number to dial to join the conference as shown in screenshot below:



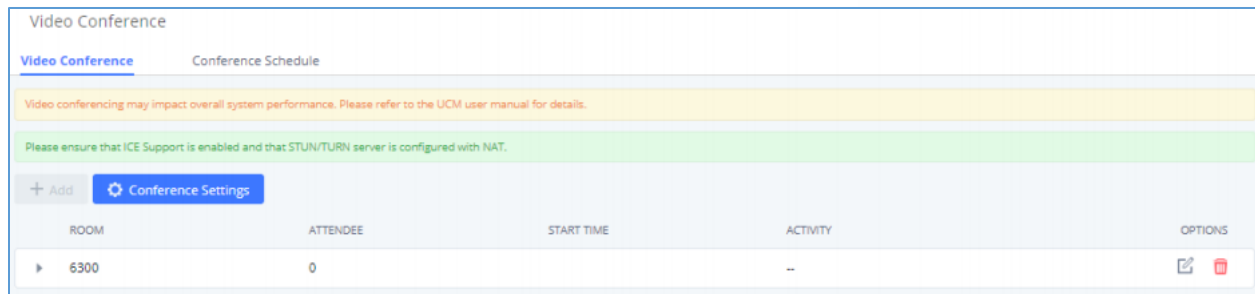


Figure 5: Created Conference Room

Schedule a Conference

Conference Schedule can be found under UCM6510 **Web GUI** → **Call Features** → **Video Conference** → **Conference Schedule**. Users can create, edit, view and delete a Conference Schedule.

- Click on “Schedule New Conference” to add a new Conference Schedule.
- Click on the scheduled conference to edit or delete the event.

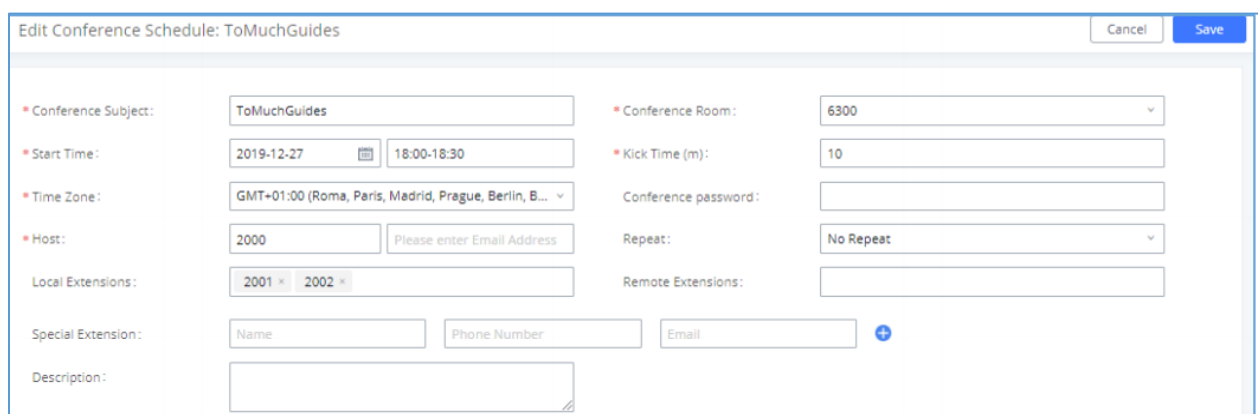


Figure 6: Video Conference Schedule

Table 3 : Video Conference Schedule Parameters

Schedule Options	
Conference Subject	Configure the name of the scheduled conference. Letters, digits, _ and - are allowed.
Conference Room	Select a conference room for this scheduled conference.
Conference Password	Configure conference room password. Please note that if “Public Mode” is enabled, this option is automatically disabled.



Kick Time(m)	Configure the time before the scheduled conference. When this time is reached, a warning prompt will be played, and all attendees currently in the scheduled conference room will be kicked after 5 mins. The conference room will be locked until the scheduled conference begins. Default value is 10 min.
Start Time	Configure the beginning start/end date of scheduled conference. Note: Please pay attention to avoid time conflict on schedules in the same conference room.
Time Zone	Defines the time zone of the scheduled conference
Host	Set the admin of this scheduled conference from the following list of members.
Repeat	Choose when to repeat a scheduled conference: <ul style="list-style-type: none"> • No repeat • Every Day • Weekly
Local Extensions	Select available extensions from the list to attend scheduled conference.
Remote Extensions	The remote extension in the peer PBX connected to the local PBX via LDAP sync.
Special Extension	Add extensions that are not in the list (both local and remote list). If the user wishes to add the special extension, please match the pattern on the outbound route.
Description	Set a description of scheduled conference.

Once created, the **Web GUI** will display scheduled conference in Conference Schedule. Please see figure below:



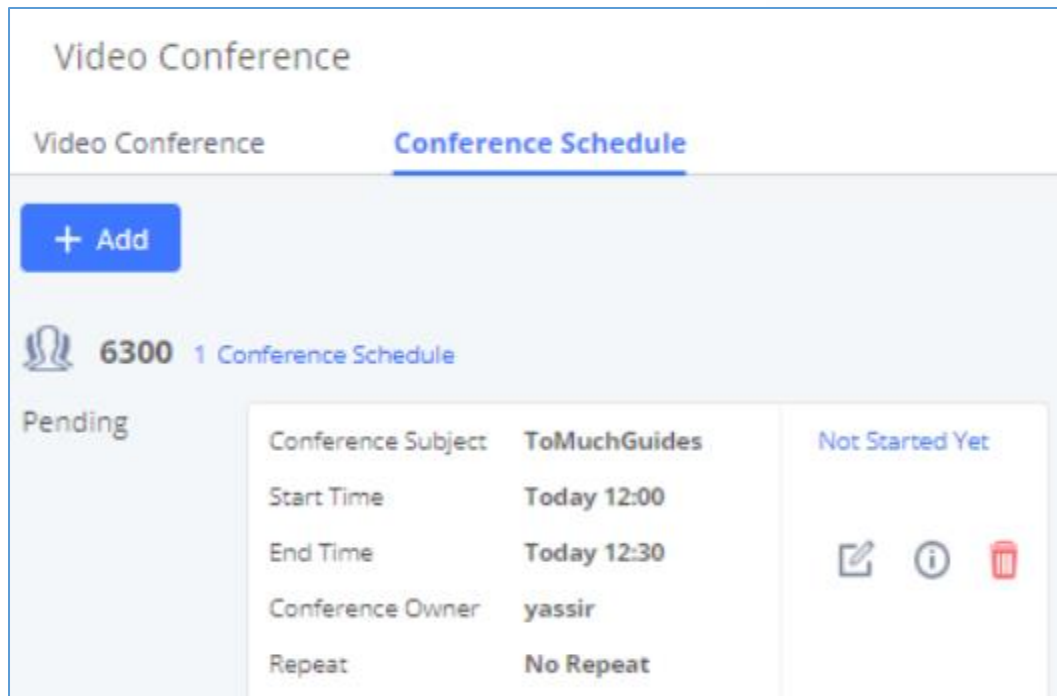


Figure 7: Video Conference Schedule

Once the conference room is scheduled, at the kick time, all users will be removed from conference room and no extension can join the conference room anymore. At the scheduled conference time, UCM6510/UCM62xx will send INVITE to the extensions that have been selected for conference.

Upon scheduling the conference, invites will be sent out to the host and selected participants. These invitations will include the conference details and a link to the conference. Upon clicking the link, participants will be prompted to enter their GS Wave portal passwords to log in and join the conference.





Hi "Michael Jordan",

You are **invited** to attend the following **conference**.

Conference Schedule Details

This is the information of the schedule **conference** which you will attendee.

Topic:
UCMSFU

Description:

Schedule Time:
2018-07-06 15:53 - 2018-07-06 16:08

Conference Info:
Conference Room: 6300
Conference Password: No Password

LAN **Conference** Link Address:
[Click the link to enter Schedule Conference](#)

WAN **Conference** Link Address:
No Address

Notes: please open the link address with Chrome or Firefox browser.

Figure 8: Invitation Link Schedule



LOGIN TO GS WAVE PORTAL

After Enabling WebRTC and creating Conference Rooms, users will be able now to establish WebRTC Calls, and participate/host conferences.

The UCM offers the possibility to login to an extension via Grandstream Wave Portal, where it offers a sleek interface to host conferences; manage contacts and share presentation.

Access the page by adding “/gswave” after the UCM’s server address and port.

(e.g. <https://my.ucm.com:8089/gswave>).



Figure 9: GS Wave Portal

Enter an extension number and its SIP or user portal password. Once logged in successfully, the following page will appear:

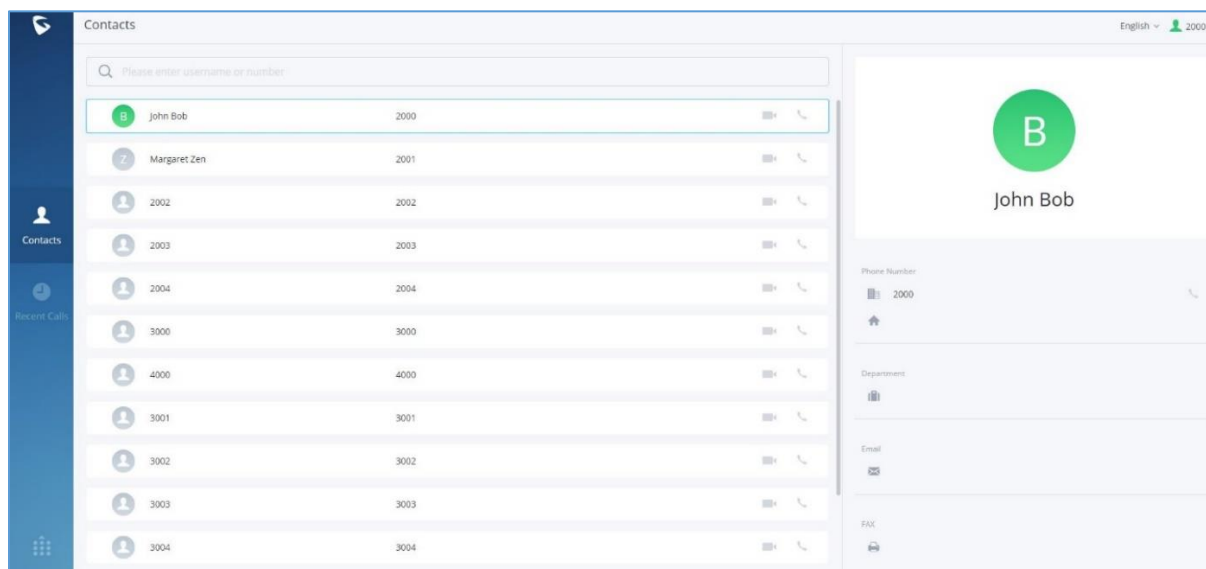


Figure 10: Contacts Page



WEBRTC FEATURES

Initiate a Call

From here on, users can initiate an audio or video call to individual extensions and conference rooms. If starting a call for the first time on a browser, users may be asked to allow the web page to use the device's mic and/or webcam. Users may be required to drop and re-establish the call after allowing webcam/microphone access. Users can initiate a call either by either selecting a contact, a recent call or entering the destination number.

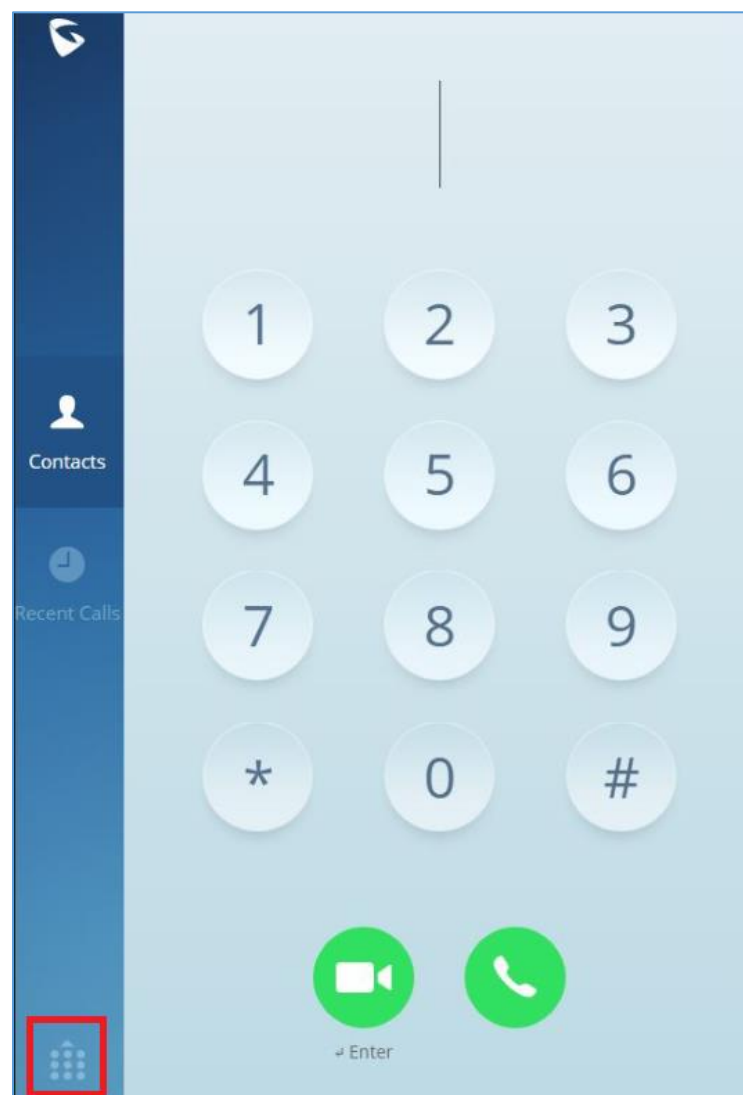


Figure 11: Initiating a Call



Users can also search for a specific contact or a specific number to dial under Contact Tab and Recent Calls Tab.

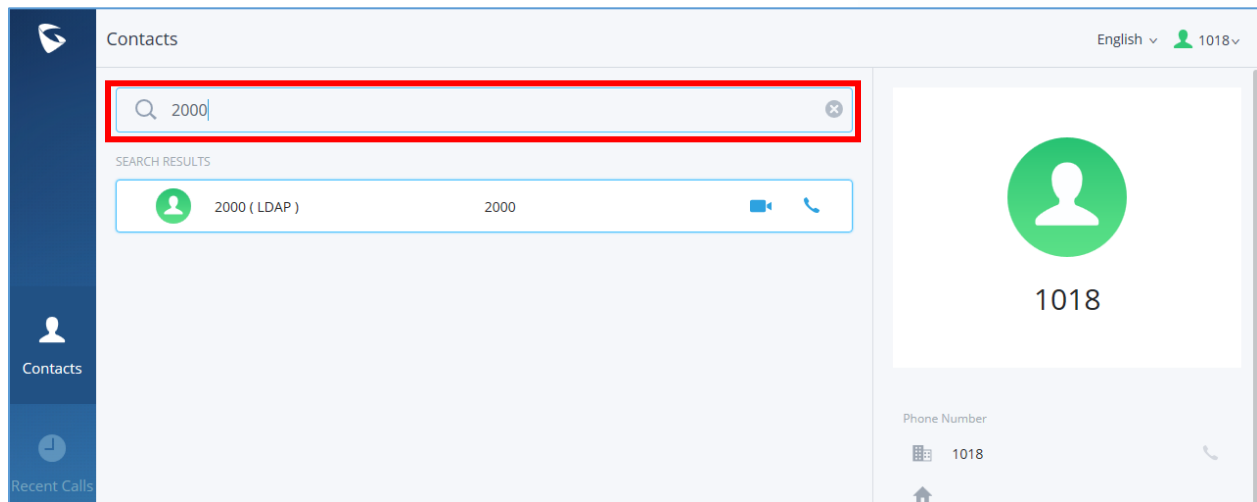


Figure 12: Contact Search

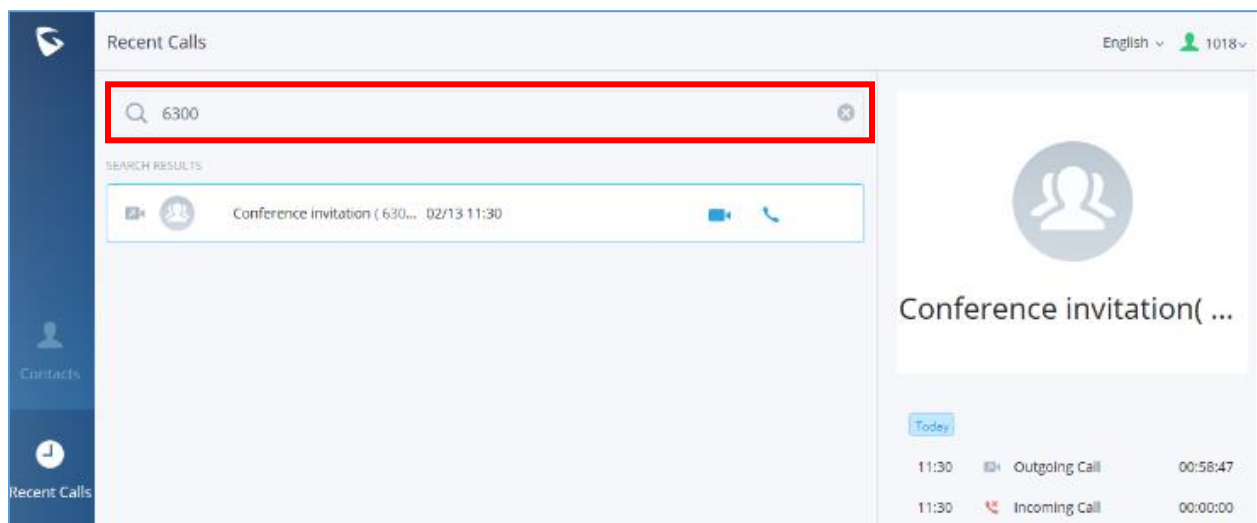


Figure 13: Recent Calls search

Once the call is established, users can mute themselves, enable and disable their video feed, screen share (Video calls only), and transfer the current call.



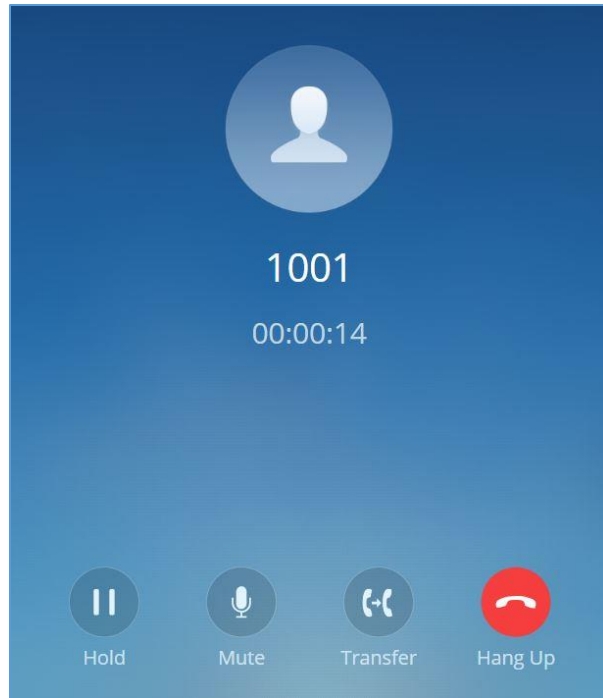


Figure 14: Audio call screen



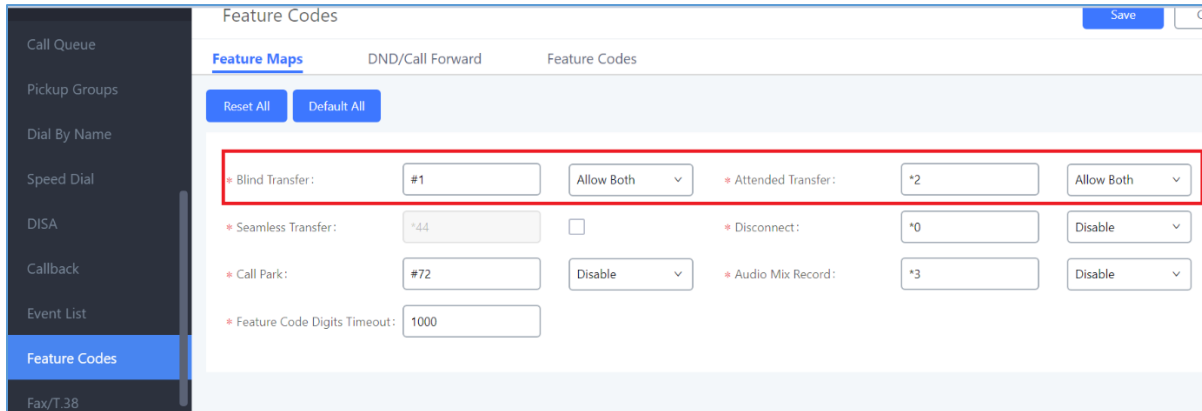
Figure 15: Video Call Screen



Transfer

The call can be transferred using the **Transfer** button to either custom numbers or to already existing extensions. In order to be able to use transfer feature, make sure to set the following:

Access Web UI as admin → Call Features → Feature Codes and set “Blind Transfer” and “Attended Transfer” to “Allow Both” as shown in figure below.



The screenshot shows the 'Feature Codes' configuration page. The 'Blind Transfer' and 'Attended Transfer' settings are highlighted with a red box. Both are set to 'Allow Both'.

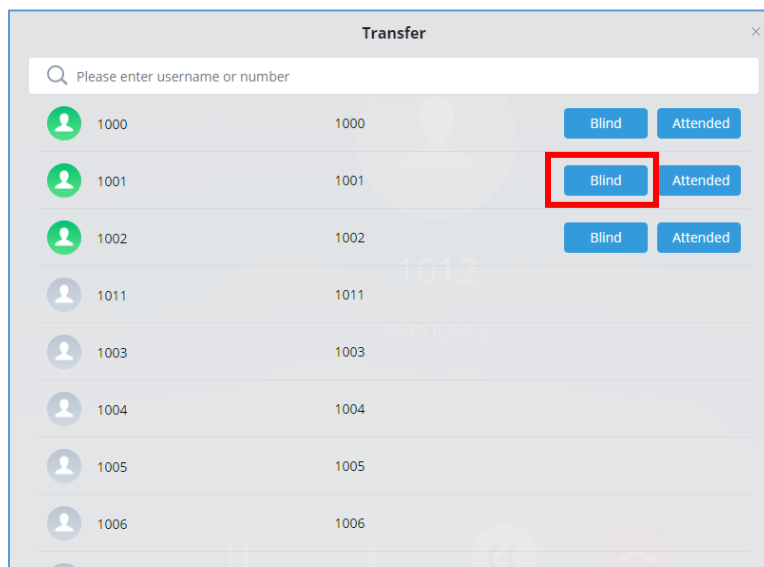
Feature Code	Value	Dropdown
Blind Transfer	#1	Allow Both
Attended Transfer	*2	Allow Both
Seamless Transfer	*44	Disable
Disconnect	*0	Disable
Call Park	#72	Disable
Audio Mix Record	*3	Disable
Feature Code Digits Timeout	1000	

Figure 16: Enable Transfer

Blind Transfer



1. During a call, press **Transfer** button
2. Enter destination number (extension or custom number).
3. Press **Blind** button to complete blind transfer.



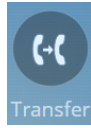
The screenshot shows the 'Transfer' screen. The 'Blind' button is highlighted with a red box.

Search	Extension	Transfer Type
1000	1000	Blind, Attended
1001	1001	Blind, Attended
1002	1002	Blind, Attended
1011	1011	Blind, Attended
1003	1003	Blind, Attended
1004	1004	Blind, Attended
1005	1005	Blind, Attended
1006	1006	Blind, Attended

Figure 17 : Transfer Screen – Blind Transfer



Attended Transfer



1. During a call, press **Transfer** button.
2. Enter destination number (extension or custom number)
3. Press **Attended** button to initiate attended transfer.
4. Press hang up button to complete the transfer.

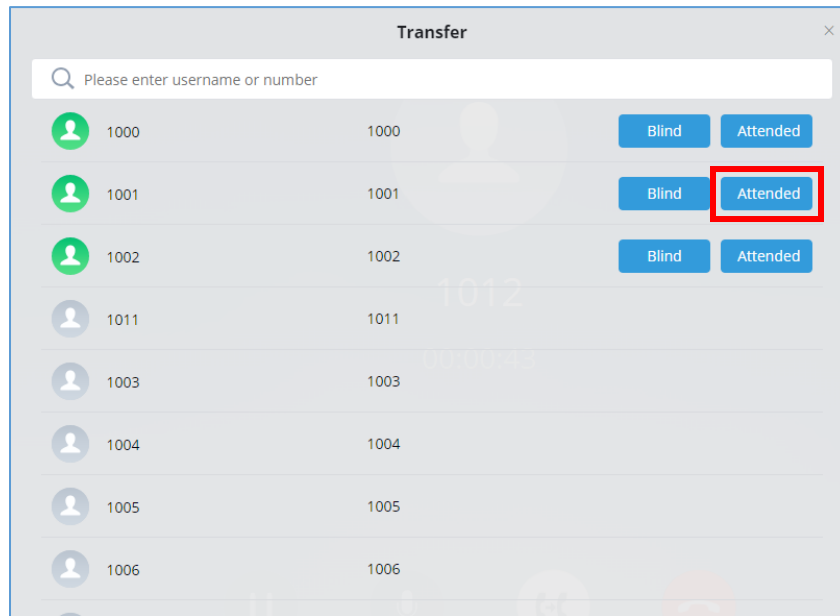


Figure 18 : Transfer Screen – Attended Transfer

Screen Sharing

Screen sharing allows users to share their whole screen or a specific browser tab to all participants:

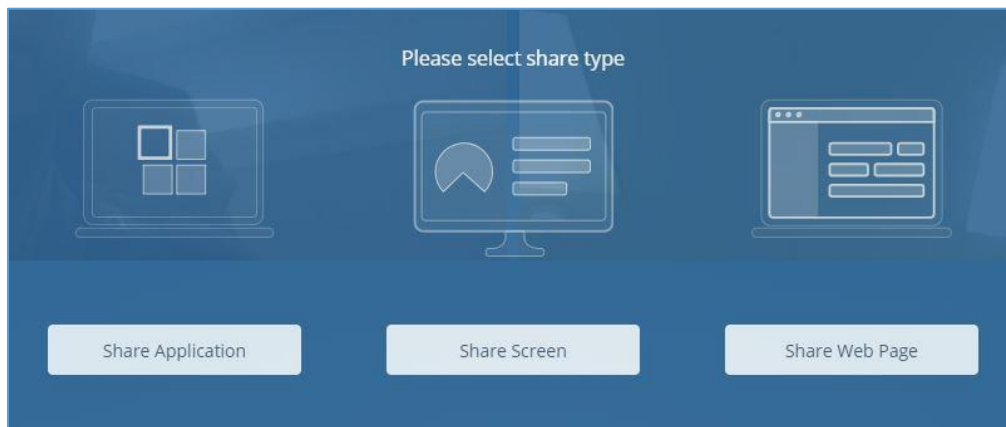


Figure 19: Screen Share Type



Screen sharing requires the IPVT Screen Capturing browser extension/add-on to work properly. If the browser does not have it, a prompt will appear asking to install or allow it. Users may need to disable their ad-blockers to have the prompts appear.

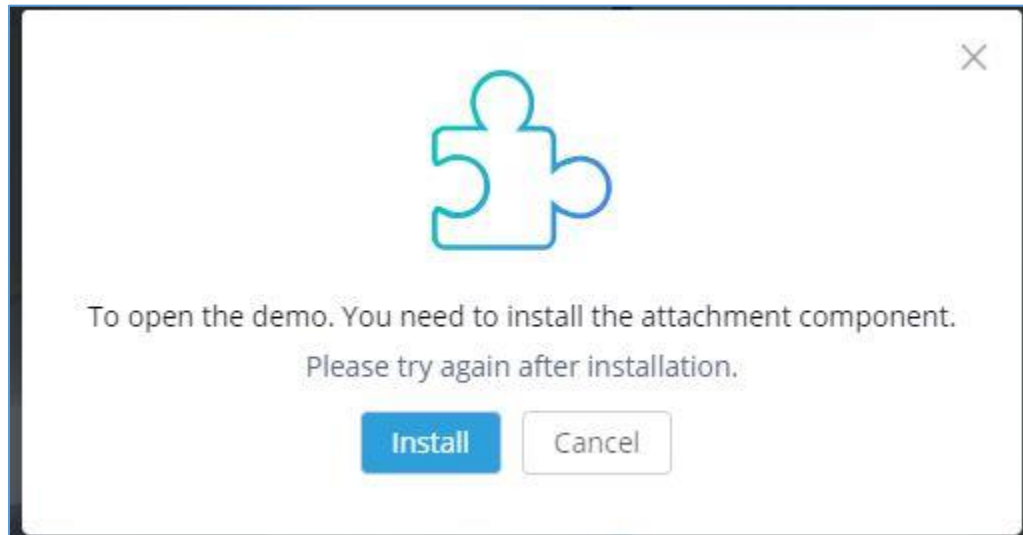


Figure 20: Plugin Setup

GXV and GVC devices will need to enable presentation to have the screen share appear.



Figure 21: Enable Presentation on GXV32xx



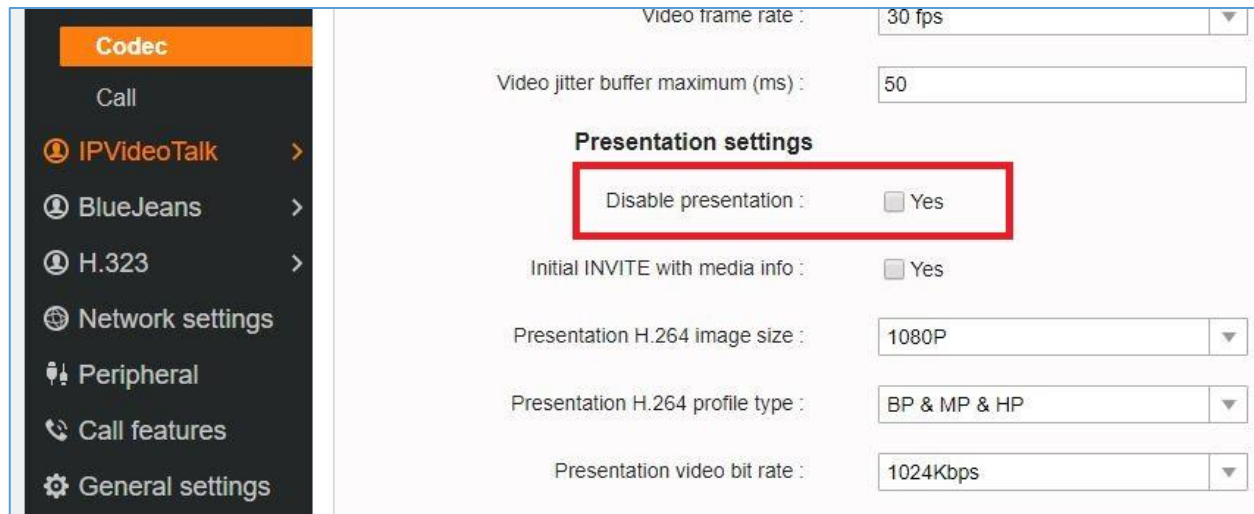


Figure 22: Enable Presentation on GVC32xx

Chat

Video conference attendees via WebRTC can chat directly through the chat section on the upper right corner of the conference page as shown on the figure below:

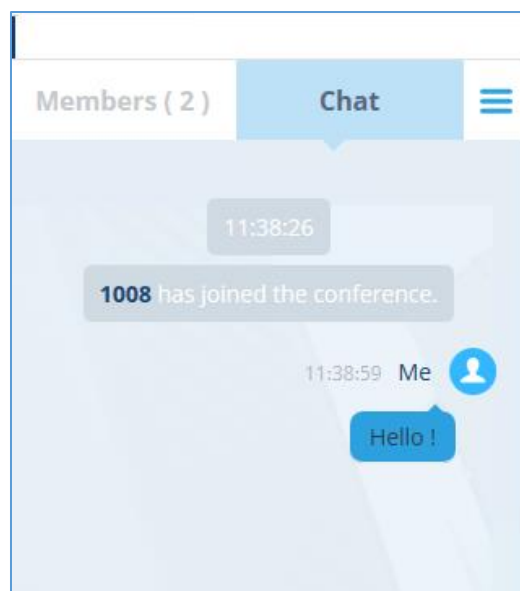



Figure 23: Chat Feature

Users can access the chat window by clicking on the conference members icon on the right lower corner of the conference page  and then clicking on the Chat tab.

The chat messages can either be transmitted to a specific member or to all the members of the conference

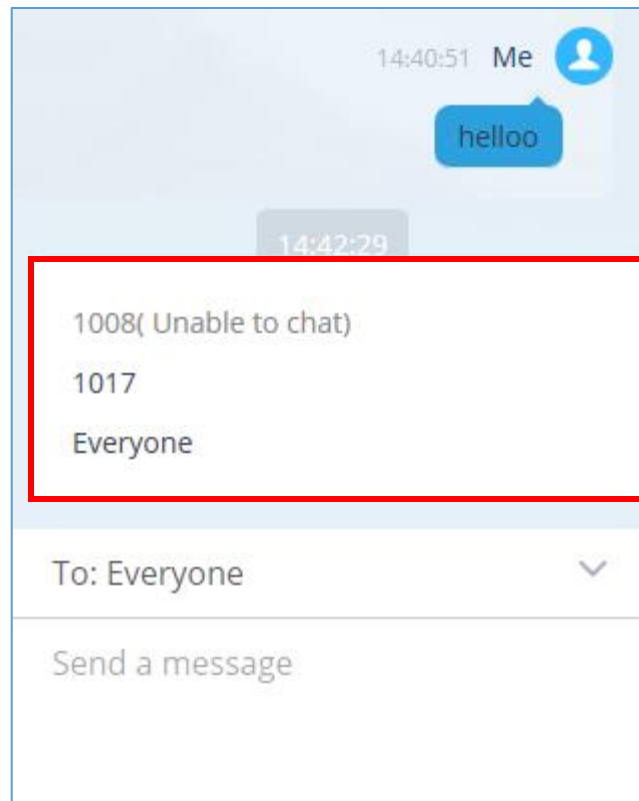


Figure 24: Chat message destination



VIDEO CONFERENCE FEATURES

Join a Video Conference Room

Users can join a video conference room by dialing the room number. Once this is established, users can mute themselves, enable and disable their video feed, screen share, invite other people to the conference, and change the video layout. Currently, only Equal (grid) and Focus video layouts are available.



Figure 25: Video Conference screen

Host Privileges

The host of a conference has the following privileges:

- i. Lock and unlock conferences to control access to it.
- ii. Mute and unmute conference participants
- iii. Transfer host privileges to one of the participants.

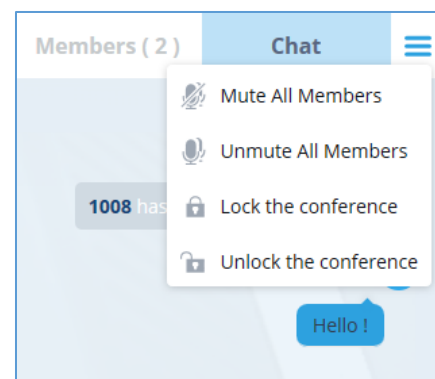


Figure 26: Host Privileges

