

Grandstream Networks, Inc.

UCM6300 Series WebRTC Trunk - Guide



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OVERVIEW

WebRTC (Web Real-Time Communications) is a technology that allows establishing calls from a web browser. It is supported by all the major web browsers like Mozilla Firefox, Chrome and other Chromium based web browsers. WebRTC supports peer-to-peer infrastructure which makes it a very flexible technology to deploy and expand.

The UCM6300 series offers creating a WebRTC trunk. After the creation, an html code of the call button is generated, and the user can embed the code into the website/web application from which calls will be initiated. Once the call reaches the UCM, it will be forwarded according to the destination set in the inbound route. E.g., the destination can be set to a call queue, if you wish to use this feature for pre-sale and after-sale customer service.

Please note that this feature is offered only when a UCM RemoteConnect plan is purchased. For more information please refer to <https://ucmrc.gdms.cloud/plans>

For more information about WebRTC please refer to the following RFC reference. RFC8835

WEBRTC TRUNK CONFIGURATION

First, create a WebRTC trunk, and to do that please access the web GUI of the UCM and then navigate to **Extension/Trunk -> WebRTC Trunks** then click on "Add".

WebRTC Trunks > Create New WebRTC Trunks

*Trunk Name ⓘ

Disable This Trunk ⓘ

☐

Auto Record ⓘ

☐

Allow Video Calling

☐

One-Click Dialing ⓘ

☐

Monitor Concurrent Calls ⓘ

☐

Incoming Concurrent Call
Threshold ⓘ

WebRTC Inbound Link
Address ⓘ

(Automatically generated after saving configuration)

Learn more about configuring WebRTC Trunks with the [WebRTC Trunk Guide](#)

Cancel

Save

WebRTC Trunk Configuration

Parameter	Description
Trunk Name	Enter the name of the trunk. Use a unique name that can be easily identifiable in the list of trunks.
Disable This Trunk	Enable this option to create the trunk but to keep it disabled.
Auto Record	When enabled, all the calls coming into the trunk will be recorded.
Allow Video Calling	Enable video calling on this trunk.
One-click Dialing	Users can dial directly after accessing the link, without entering the call interface to trigger the call again.
Jitter Buffer	Select jitter buffer method. The default option is NetEQ. Disable: Jitter buffer will not be used. Fixed: Jitter buffer set to the value configured in the "Jitter Buffer Size" field. Adaptive: Jitter buffer with a adaptive size that will not exceed the value configured in the "Max Jitter Buffer" field. NetEQ: Dynamic jitter buffer via NetEQ. Note: This option is displayed only when the administrator edits a WebRTC Trunk that has been created previously.
Monitor Concurrent Calls	If enabled, the number of concurrent calls on this trunk will be monitored. If the "Trunk Concurrent Calls" system alert is enabled, alert notifications will be generated if the number of concurrent calls exceeds this trunk's configured concurrent call thresholds.
Incoming Concurrent Call Threshold	Threshold of all incoming concurrent calls passing through this trunk.

WebRTC Inbound Link Address	This link can be embedded onto a web page. Clicking the link will connect to a pre-configured WebRTC trunk destination. You can also enter this link in the browser address bar to directly access and test WebRTC calls.
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After the creation of the WebRTC trunk, it will be displayed as follows.

WebRTC Trunks			
Add			
Trunk Name	Trunk Status	Inbound Link	Options
WebRTC_Live	● Available	https://...gdms.cloud/c...	
Total: 1 < 1 > 10 / page			

WebRTC Trunks

INBOUND ROUTE CONFIGURATION

Now that the WebRTC Trunk is created, let's configure an inbound route for the trunk so we can route all our calls to a specific destination of our choice.

To create an inbound route, please access the web GUI of the UCM and navigate to **Extension/Trunk -> Inbound Routes** then click on "Add".

Inbound Routes > Create New Inbound Rule

General

* Trunks ⓘ

WebRTC Trunks -- WebRTC_Live ▼

* Inbound Route Name ⓘ

Inbound-Route-WebRTC

Disable This Route ⓘ

☐

Call Setting

Ringback Tone ⓘ

None ▼

Alert-info ⓘ

None ▼

Auto Record ⓘ

☐

Play Holiday Prompt During Holiday ⓘ

☐

CallerID Setting

Prepend Trunk Name ⓘ

☐

Inbound Mode

Inbound Multi-Mode ⓘ

☐

Cancel

Save

Inbound Route Rule Configuration

On "Trunks" select the WebRTC trunk that has been created, then specify a name for the Inbound Route. Scroll down to "Default Destination" and select the default destination to which the inbound calls will be forwarded.

For more information about other option in "Inbound Route" please refer to the table below.

Trunks	Select the trunk to configure the inbound rule.
Inbound Route Name	Configure the name of the Inbound Route. For example, "Local", "LongDistance" etc.
Disable This Route	After creating the inbound route, users can choose to enable and disable it. If the route is disabled, it will not take effect anymore. However, the route settings will remain in the PBX. Users can enable it again when it is needed.
Call Settings	
Ringback tone	Choose the custom ringback tone to play when the caller reaches the route.
Auto Record	If enabled, calls using this route will automatically be recorded.
Play Holiday Prompt During Holiday	During holidays, the system will play the prompt configured in Time Settings > Holiday to the caller before the call is routed to the destination.
Alert-Info	Configure the Alert-Info, when the PBX receives an INVITE request, the Alert-Info header field specifies an alternative ring tone to the UAS.
CallerID Setting	
Prepend Trunk Name	If enabled, the trunk name will be added to the caller id name as the displayed caller id name.
Inbound Mode	
Inbound Multi-Mode	Gives uses the ability to configure inbound mode per individual route. When enabled two fields will show allowing to set the Inbound mode and the Inbound mode Suffix.Note: Global inbound mode must be enabled before users can configure route-level inbound mode.
Custom Inbound Mode	If enabled, this route will use the inbound mode specified in this page's Inbound Mode setting instead of the configured Global Inbound Mode.
Inbound Mode	Choose the inbound mode for this route.Note: Toggling the global inbound mode will not affect routes that have Route-level Inbound Mode enabled. If all routes have the option enabled, toggling the global inbound mode via BLF will trigger a voice prompt indicating that none of the routes will be affected by the global inbound mode change.
Inbound Mode Feature Code	Assigns a suffix that can be appended to the Global Inbound Mode feature code to switch to the corresponding mode on this route. This suffix can also be dialed directly without the feature code to toggle between the current mode and the default mode of this route.Example: If Global inbound Mode > Mode 1 feature code is *62, and this route's feature code is set to 88, dialing *6288 would change this specific route's inbound mode to Mode 1. Dialing 88 directly toggles between this route's Default Mode and Mode 1.
Default Destination	Select the default destination of the call. Extension Multimedia Meeting Voicemail Voicemail Group IVR Ring Group Queue Paging/Intercom Call Flow Fax DISA Dial By Name External Number Announcement Time Condition Routing
Time Condition	
Time Condition	Select the time condition. Office Time Out of Office Time Holiday Out of Holiday Out of Office Time or Holiday Office Time and Out of Holiday Custom Time
Destination	Inbound calls will be routed to this destination if time conditions are satisfied. Extension Multimedia Meeting Voicemail Voicemail Group IVR Ring Group Queue Paging/Intercom Call Flow Fax DISA Dial By Name External Number Announcement Time Condition Routing

WebRTC Trunk Usage Methods

You can use two different methods to provide the call terminal to your clients, either through a link, or by embedding a button into your website that the clients click to call the WebRTC trunk.

One-Click Dial Link

You can copy the link of the page of the call terminal and put it on your website.

On the WebRTC Trunk page, click on the copy button.

WebRTC Trunks

Add

Trunk Name	Trunk Status	Inbound Link	Options
WebRTC_Live	Available	https://...b.gdms.clou... <div>Copy</div>	<div>Edit</div> <div>Share</div> <div>Download</div> <div>Delete</div>

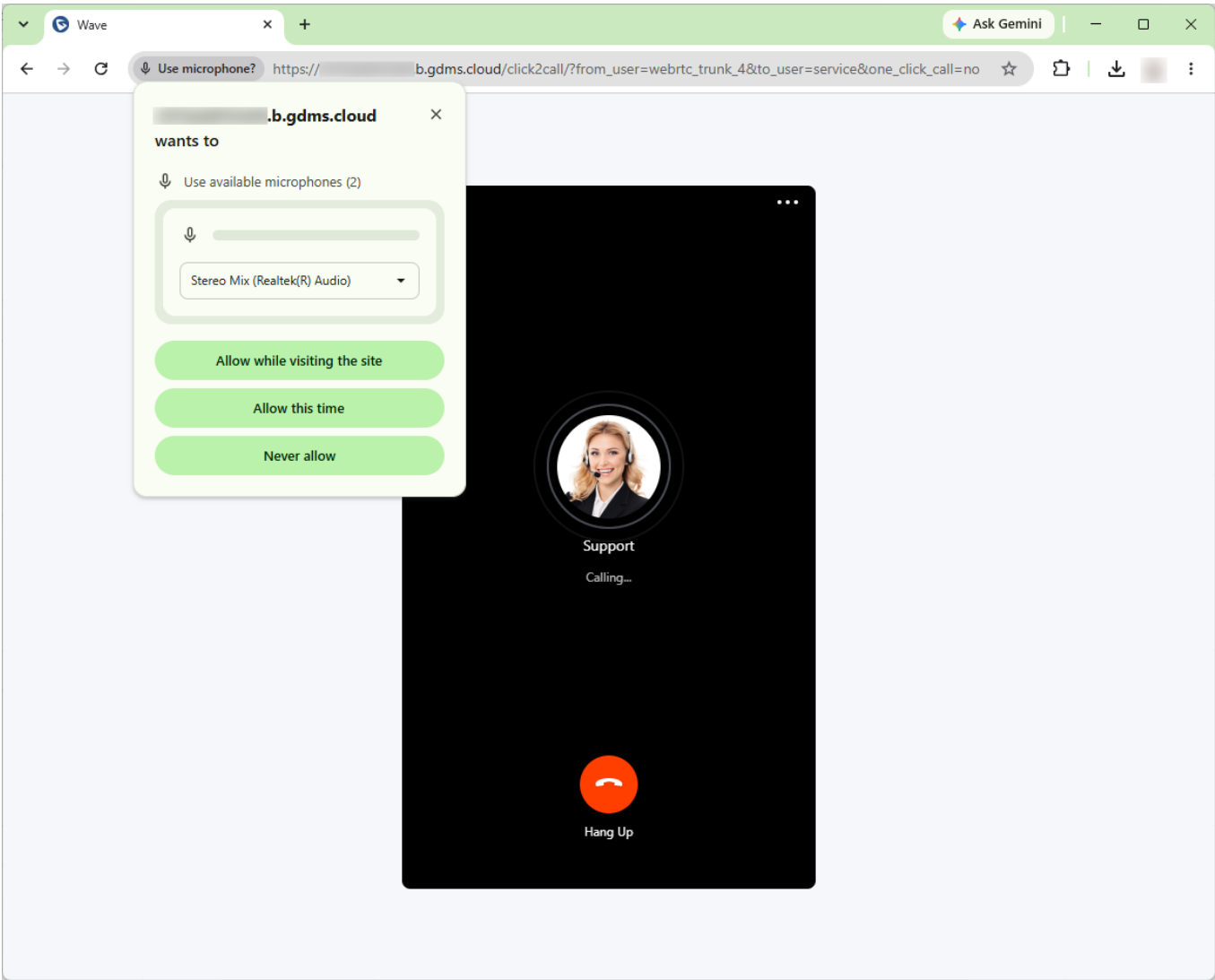
Total: 1

<1>

10 / page

When a client clicks on the link, the call will be immediately triggered.

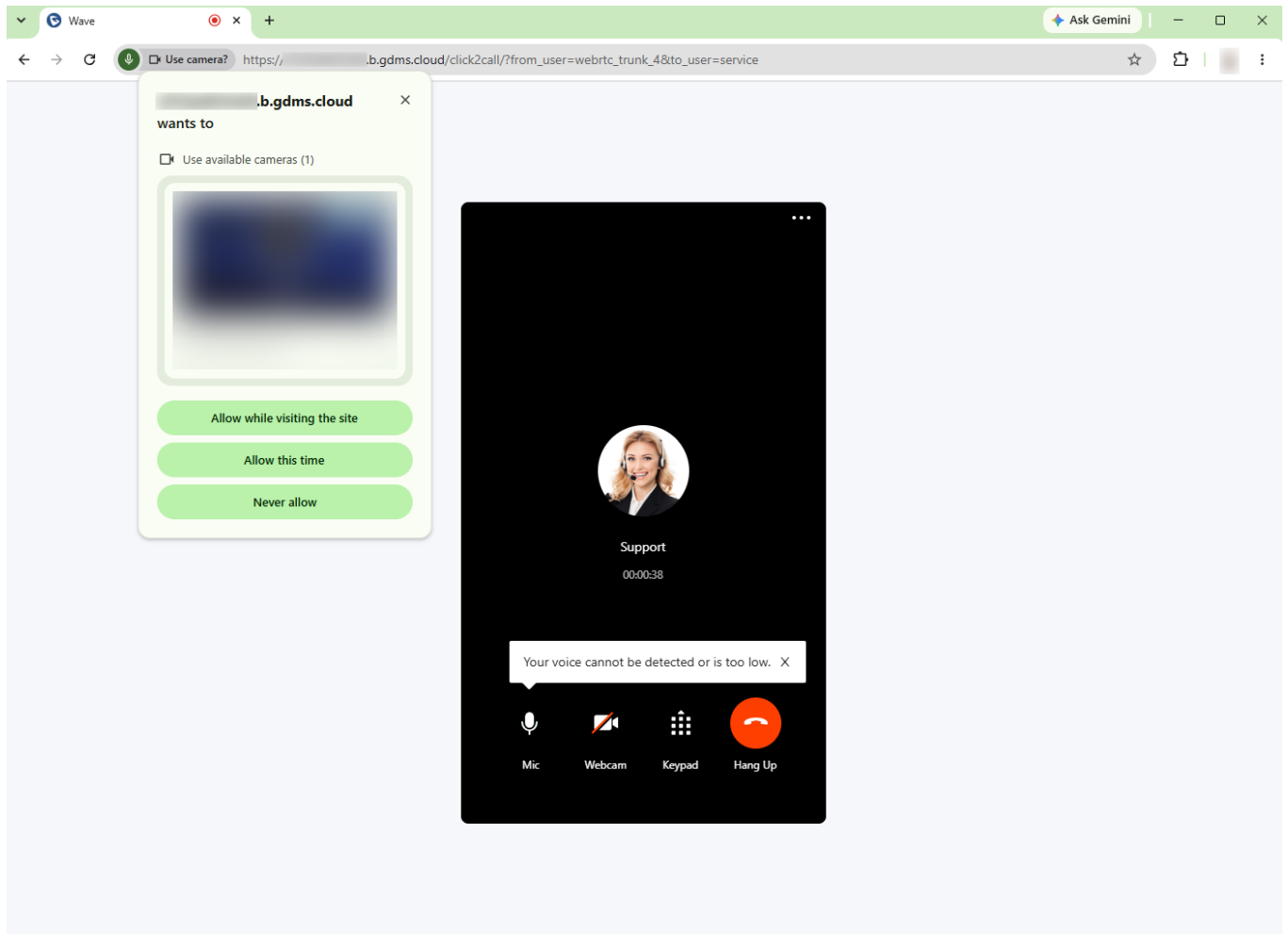
When the call is initiated for the first time, the client will be prompted to allow the use of a microphone, the follow prompt will be displayed.



Click on "Allow while visiting" or "Allow this time" to permit the microphone usage.

To start a video, you need first to establish an audio call, then the PBX will verify if video calls are permitted in the configuration of the WebRTC trunk. If enabled, a button to enable video calling will be displayed.

When video call is initiated by the client, they need to grant the permission camera use by the browser. A prompt will appear when the video call is initiated. The screenshot below shows the prompt that appears, click on "Allow this time" or "Allow" to grant the permission of camera use to the browser.



Camera Permission Prompt

If you click "Never allow" accidentally, you will have to reset the site permissions to be able to get the prompt again.

Click-to-Call Button Embedding

The code generated by the PBX for embedding click-to-call allows adding a floating button to your website's page in a neat design that provides a quick way to initiate a call to your service agents, without obstructing or changing the layout of the webpages of your site. The code provided can be edited to increase and decrease the size of the button, change the position of the button in your site's layout, or change the color of the button to match the color palette of your website, etc...

Add

Trunk Name ▾	Trunk Status ▾	Inbound Link ▾	Options
WebRTC_Live	● Available	https://...b.gdms.clou...	

Total: 1 < 1 > 10 / page ▾

WebRTC Trunks

Click on "Copy Code" to copy it as text, or click "Download Code" to download the code as a Javascript file.

Embed the code snippet on your website



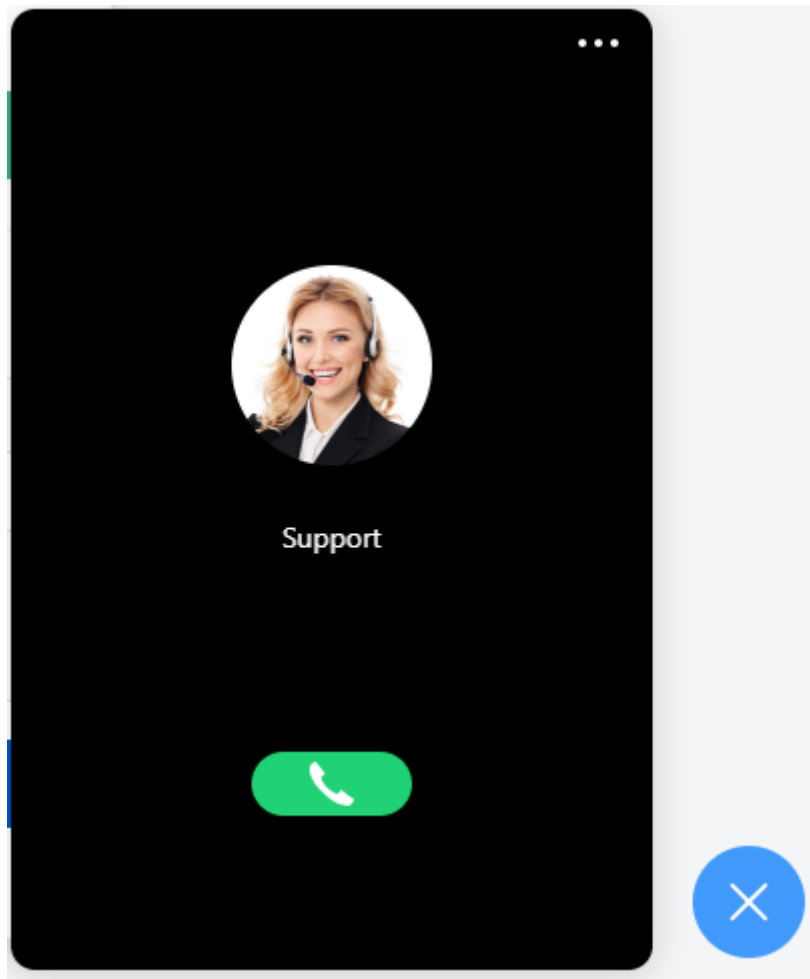
To add online call to websites, users can

1. Embed the *Online Call Link Address* into the web page.
2. Embed the following JavaScript code snippet into the web page, which will create a online call button in the lower right corner of the page:



DownloadCopy The Code

When the code is embedded, the follow button will appear , which the user can click on, and the following window appears where the user can initiate the call.



Embedded Call Window

For more information about WebRTC please refer to the following RFC reference. RFC8835

Live Chat Integration

WebRTC Trunk is used alongside Live Chat feature to provide support through call if necessary.

* Name

LiveChat1

* Destination ⓘ

Extension ▾

1001 ▾

Web Page Language ⓘ

Use Browser Language ▾

Disable ⓘ

☐

Call Settings

Allow Visitor to Call ⓘ

☒

Allow Calling Before Chat ⓘ

☒

* WebRTC Trunks

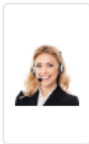
https://.b.gdms.cloud/cl... ▾

[Go to Page](#)

The default destination of the inbound route bound to this WebRTC trunk will be changed to the same destination as this Live Chat.

Customer Service Agent Information

Avatar



Please select a file in png, jpg, jpeg format.

Cancel

Save

Live Chat WebRTC Trunk

For more information regarding Live Chat, please refer to the following guide:
<https://documentation.grandstream.com/knowledge-base/live-chat-user-guide/>