

Grandstream Networks, Inc. UCM6XXX WebRTC Demo Guide





Table of Contents

OVERVIEW	. 3
CONFIGURING HTTP & WEBSOCKET	. 3
ENABLING WEBRTC FOR EXTENSION	. 5
USING USER PORTAL DEMO	. 6
REFERENCES	. 9

Table of Figures

Figure 1: HTTP & WebSocket	3
Figure 2: HTTP & WebSocket Status	5
Figure 3: Enable WebRTC Support	6
Figure 4: Extension with WebRTC Enabled	6
Figure 5: Extension User Portal Password	7
Figure 6: User Portal->Value-added Features: Register	7
Figure 7: User Portal->Value-added Features: Connected	. 8
Figure 8: Dial Number	. 8
Figure 9: Call Control	9
Figure 10: Click on "Enjoy our live demo"	9
Figure 11: Click on "Expert mode"	10
Figure 12: WebSocket Server URL	11
Figure 13: Registration and Call Control	11





OVERVIEW

The UCM6xxx supports HTTP & WebSocket for web browser to register to the UCM and establish calls with other endpoints in real time via webRTC. This document describes how to set up an easy WebRTC connection through UCM6xxx webUI User Portal.

Note: UCM6xxx series include UCM6100 series (UCM6102, UCM6104, UCM6108 and UCM6116), UCM6200 series (UCM6202, UCM6204 and UCM6208) and UCM6510.

CONFIGURING HTTP & WEBSOCKET

On the UCM6xxx web UI, the users need firstly configure HTTP and WebSocket in order for the webRTC client to establish connection with the UCM.

- 1. Log in UCM6xxx webUI and navigate to PBX->Value-added Features->HTTP & WebSocket.
- 2. Select the checkbox for "Enable HTTP".
- 3. Enter the UCM IP address in "HTTP Bind Address" field, e.g., 192.168.40.214.
- 4. Enter the port number for "HTTP Bind Port". The default "HTTP Bind Port" is 8088. Once configured, the UCM6xxx will listen on HTTP connection through port 8088.
- 5. If the users would like to use TLS for security purpose, select the checkbox for "TLS Enable". Then enter the "TLS Bind Address" of the UCM IP Address with port number, e.g., 192.168.40.214:8445. The default port number is 8445.

	Status PBX	Settings Maintenance
	PBX >> Value-added Features	>> HTTP & WebSocket
Basic/Call Routes	HTTP & WebSocket	
Call Features		
Internal Options	(i) Enable HTTP:	
IAX Settings	(i) HTTP Bind Address:	192.168.40.214
SIP Settings	(i) HTTP Bind Port:	8088
Ports Config	I TLS Enable:	~
Zero Config	() TLS Bind Address:	192.168.40.214:8445
Value-added Features	() WebSocket Interface:	ws://192.168.40.214:8088/ws
- Fax Sending	 Secure WebSocket Interface: 	wss://192.168.40.214:8445/w
- Announcement Center		Cancel
- HTTP & WebSocket		
PMS		
CRM		







- Once the above are configured, the WebSocket Interface and Secure WebSocket Interface will be automatically filled up. For example: WebSocket Interface: ws://192.168.40.214:8088/ws Secure WebSocket Interface: wss://192.168.40.214:8445/ws
- 7. Save and apply the change.

The following table shows the detailed description for all the options in HTTP & WebSocket section.

Enable HTTP	Enable it to allow WebRTC user (such as Chrome or Firefox) to register to UCM6xxx via HTTP.
HTTP Bind Address	Configure IP address for HTTP server on the UCM61xx to bind to. When HTTP Bind Address set to 0.0.0.0, it means UCM6xxx will listen on all the socket connections through HTTP Bind Port.
HTTP Bind Port	Configure the port for the HTTP server on the UCM6xxx that binds to. By default, it is set to 8088.
TLS Enable	Enable it to allow WebRTC user (such as Chrome or Firefox) to securely register to UCM6xxx via HTTPS, by default it is disabled.
TLS Bind Address	Configure IP address for TLS server on the UCM6xxx to bind to. When TLS Bind Address set to 0.0.0, it means the UCM6xxx will listen on all the socket connections through the configured port. By default, the port number is set to 8445. Note: By default the TLS Bind address is 0.0.0.0:8445, which means binding to all interfaces via port 8445. CDR API using TLS is using the same port number. To avoid conflict, please manage different port number accordingly when both WebSocket TLS and CDR API TLS feature are enabled.
WebSocket Interface	Indicates WebRTC URL that will be used to connect to UCM6xxx via HTTP. WebSocket Interface is associated with HTTP Bind Address and Port. For example, if HTTP Bind Address and Port set to 192.168.1.2:8088, WebSocket Interface will be ws://192.168.1.2:8088/ws
Secure WebSocket Interface	Indicates secure WebRTC URL that will be used to connect to UCM6xxx with HTTPS. Secure WebSocket Interface is associated with TLS Bind Address. For example, if TLS Bind Address set to 192.168.1.2:8445, Secure WebSocket Interface will be wss://192.168.1.2:8445/ws

8. Check the UCM HTTP & WebSocket status using the following link:

For HTTP:

http://UCM_IP_ADDRESS:PORT/httpstatus

UCM_IP_ADDRESS should be your UCM's IP address, PORT number is the same as configured in "HTTP Bind Port". For example, <u>http://192.168.40.214:8088</u>.





If TLS is enabled, use the following link instead:

https://TLS_BIND_ADDRESS/httpstatus

TLS_BIND_ADDRESS should be the same as configured in "TLS Bind Address". For example, <u>https://192.168.40.214:8445</u>.

A warning prompt will pop up for you to confirm the security risk. Once confirmed, the following Asterisk information will be displayed.

Please note: Since UCM6xxx is using self-signed certificate, the WebRTC connection will be dropped by browser by default unless the user confirms the security risk.

AsteriskTM HTTP Status

Server	Asterisk/13.4.0					
Prefix	192.168.40.214 8088					
Bind Address						
Bind Port						
SSL Bind Port	8445					
Cookie 'jumpMenu'	sip_webrtc.html					
Cookie 'locale'	en-US					
Cookie 'localeDirection'	ltr					
Cookie 'position'	home					
Cookie 'first_login'	no					
Cookie 'is_strong_password	₇ , 0					
Cookie 'role'	privilege_0					
Cookie 'html'	%7B%22upgrade%22%3A1%2C%22backup%22%3A1%2C%22backup_network%22%3A1%2C%22auto_cleaner cmpil%22%3A1%2C%22uuto_creatives%22%3A1%2C%22backup%22%3A1%2C%22backup%22%3A1%2C%22backup%22%3A1%2C%22backup%22%					

Cookie 'html'	$\label{eq:2} \end{tabular} \begin{tabular}{lllllllllllllllllllllllllllllllllll$
Cookie 'user_id'	0
Cookie 'username'	admin
Cookie 'session- identify'	sid1985282665-1484355500
Cookie 'TRACKID'	c96cea1303432ff88fcb324a5a207002

Asterisk and Digium are registered trademarks of Digium, Inc.

Figure 2: HTTP & WebSocket Status

ENABLING WEBRTC FOR EXTENSION

Before using the demo in UCM user portal, the users need to enable WebRTC for this extension.

- 1. Log in UCM6xxx web UI and navigate to **PBX->Basic/Call Routes->Extensions**.
- 2. Select the extension to edit and open tab "Features".
- 3. Select the checkbox for "Enable WebRTC Support".
- 4. Save and apply the setting.





C GRANDSTREAM	Edit Extension: 1000				x
CONNECTING THE WORL	Basic Settings Media Fea	tures Specific Time			*
	Simultaneously:				
	Monitor privilege control				
Basic/Call Routes	(i) Allowed to call-barging:				
- Extensions		Available Extensions	Selected Extension	ons	
- Analog Trunks	1000 1001	*	0	*	
- VolP Trunks			0		
- SLA Station		-	0	*	I.
- Outbound Routes	Other Settings				
- Inbound Routes	(i) Ring Timeout:		Auto Record:		I.
Call Features	(i) Skip Trunk Auth:	No *	Dial Trunk Password:		I.
Internal Options	 Support Hot-desking Mode: 		(i) Enable LDAP:		I.
IAX Settings	Enable WebRTC Support:		(i) Music On Hold*:	default 💌	I.
SIP Settings	Enable Seamless Transfer:		() Call Duration Limit:		I.
Ports Config	 Custom Call-info for Auto 				I.
Zero Config	Answer:				Ŧ
Value-added Features		Cancel	Save		

Figure 3: Enable WebRTC Support

5. Check the extension in **PBX->Basic/Call Routes->Extensions** page. The above extension will have its terminal type shown as "SIP(WebRTC)".

	PBX >> Basic/Call Ro	outes >> Extensions					
Basic/Call Routes	Manage Extensior	IS					
 Extensions Analog Trunks 	Extension:		(i) CallerID Name:		Search	Show All Exte	ensions
- VolP Trunks - SLA Station - Outbound Boutes	Create New Extension	n Modify Select	ted Extensions Del	ete Selected Extensio	ons Batch Add	Extensions	Import Extensi View: 30
 Inbound Routes 	Status	Extension ⊘	CallerID Name	Terminal Typ	IP and Port	Email Status	Options
Call Features	•	1000		SIP(WebRT C)		To Be Sent	/ 心 前
Internal Options	•	1001		SIP		To Be Sent	/ 心 🏛
IAX Settings	Total: 2 Show: 1/1	Go to: Go				First Prev	v Next Last



USING USER PORTAL DEMO

1. On the UCM6xxx web UI, log in user portal with the extension number the user password.

Please note the user password for the extension to log in user portal is initially created with admin access under UCM6xxx **web UI->PBX->Basic/Call Routes-**>Edit extension. Please consult with your UCM admin for the user password for your extension to log in user portal.





C GRANDSTREAN	Edit Extension: 1000						x
CONNECTING THE WORL	Basic Settings Media Fea	Basic Settings Media Features Specific Time					
	General						
	(i) Extension*:	1000		(j)	CallerID Number:]
Basic/Call Routes	(i) Permission:	Internal 🔻		(i) :	SIP/IAX Password*:	••••••	0
- Extensions	(i) AuthID:			(i)	Enable Voicemail:	\checkmark	
- Analog Trunks				(i) :	Skip Voicemail Password		
- VoIP Trunks	Voicemail Password*:	••••••	0	,	Verification:		
- SLA Station	 Disable This Extension: 						
- Outbound Routes	User Settings						
- Inbound Routes	First Name:]	(j)	Last Name:]
Call Features	Email Address:			(j)	User Password [*] :	****	
Internal Options	(i) Language:	Default •		(j)	Concurrent Registrations:	1]
IAX Settings	 Mobile Phone Number: 						

Figure 5: Extension User Portal Password

2. In the user portal web UI->Value-added Features->WebRTC, WebRTC settings will be automatically filled up with the previously configured information in above steps.

Please make sure the password is the one used to register the SIP extension. Also, the WebSocket Server URL is in the format of <u>ws://192.168.40.214:8088/ws</u>, or <u>wss://192.168.40.214:8445/ws</u> if TLS is enabled.

User Portal						
User Portal >> Value-added Features >> WebRTC 😷						
Basic Information	WebRTC					
My files						
Value-added Features	Advanced Options					
- WebRTC	Need Registration					
- Fax Sending						
- Wakeup Service	Account Name:	1000				
- CRM User Settings	(i) Extension*:	1000				
	Public Identity*:	sip:1000@192.168.40.214				
	(i) Password*:	•••••				
	WebSocket Server URL*:	ws://192.168.40.214:8088/ws				
		Unregister Register				

Figure 6: User Portal->Value-added Features: Register

3. Click on "Register" button on the above picture. The connected status will show in a few seconds.





User Portal				
	User Portal >> Value-added F	Features >> WebRTC 🕥		
Basic Information	WebRTC			
My files	Advanced Options			
Value-added Features	Need Registration			
 WebRTC Fax Sending Wakeup Service CRM User Settings 	 Account Name: Extension[*]: Public Identity[*]: Password[*]: WebSocket Server URL[*]: 	Connected 1000 1000 sip:1000@192.168.40.214 wss://192.168.40.214:8445/ws Unregister Register Register		
	Call Control			

Figure 7: User Portal->Value-added Features: Connected

4. Now the users can make call from the user portal. Enter the number to call and select "Audio" or "Video" call. Call control options are also available once the call is established.

User Portal				
	User Portal >> Value-added F	eatures >> WebRTC 🗘		
Basic Information		Connected		
My files	Account Name:	1000		
Value-added Features	(i) Extension*:	1000		
- WebRTC	Public Identity*:	sip:1000@192.168.40.214		
- Fax Sending	Password [*] :	•••••		
- Wakeup Service	(i) WebSocket Server URL*:	wss://192.168.40.214:8445/ws		
United Settings		Unregister		
	Call Control			
		Video enabled		
	(i) Dial Number:	Hang-up Call -		

Figure 8: Dial Number





User Portal				
User Portal >> Value-added Features >> WebRTC 🙃				
Basic Information	(i) Password*:			
My files	(i) WebSocket Server URL*:	wss://192.168.40.214:8445/ws		
Value-added Features		Unregister		
- WebRTC	Call Control			
- Fax Sending		In Call		
- Wakeup Service	 Dial Number: 	1001		
- CRM User Settings		Hang-up Call -		
		Mute Hold Transfer KeyPad		

Figure 9: Call Control

Note:

• It is recommended to enable TLS for HTTP & WebSocket because web browser might have security restrictions to use HTTP for webRTC.

REFERENCES

Users could find reference in below website, which includes live demo as well as source code:

http://www.doubango.org/sipml5/

1. Open the above link and click on "Enjoy our live demo".

World's first HTML5 SIP client

This is the world's first open source (BSD license) HTML5 SIP client entirely written in javascript for integration in social networks (FaceBook, Twitter, Google+), online games, e-commerce websites, email signatures... No extension, plugin or gateway is needed. The media stack rely on WebRTC.

The client can be used to connect to any SIP or IMS network from your preferred browser to make and receive audio/video calls and instant messages.

Enjoy our live demo »

Figure 10: Click on "Enjoy our live demo"

2. Click on "Expert mode".





Registration				
Display Name:	e.g. John Doe			
Private Identity*:	e.g. +3360000000			
Public Identity*:	e.g. sip:+3360000000@doubango.org			
Password:				
Realm*:	e.g. doubango.org			
* Mandatory Field	LogIn LogOut			
Need SIP account?				
Expert mode?				

Figure 11: Click on "Expert mode"

3. Enter the WebSocket Server URL to the same one as shown in UCM configuration *Figure 1: HTTP & WebSocket*. Click on "Save".

Expert settings

Disable Video:	
Enable RTCWeb Breaker ^[1] :	
WebSocket Server URL ^[2] :	ws://192.168.40.214:8088/ws
SIP outbound Proxy URL ^[3] :	e.g. udp://sipml5.org:5060
ICE Servers ^[4] :	e.g. [{ url: 'stun:stun.l.google.com:19302'}, { url:'turn:user@
Max bandwidth (kbps) ^[5] :	{ audio:64, video:512 }
Video size ^[6] :	{ minWidth: 640, minHeight:480, maxWidth: 640, maxHeig
Disable 3GPP Early IMS ^[7] :	
Disable debug messages ^[8] :	
Cache the media stream ^[9] :	
Disable Call button options ^[10] :	
	Save





Figure 12: WebSocket Server URL

4. In "Registration" page, fill up "Display Name", "Private Identity", "Public Identity", "Password" and "Realm".

"Display Name": The name you would like to display when calling the remote party.

"Private Identity": The SIP User ID on UCM.

"Public Identity": The SIP URI for this extension, in the format of sip:extension@ucm_ip.

"Password": This is the password used to register the SIP extension

"Realm": Enter "Grandstream" here.

- 5. Click on "Login".
- 6. Once the extension is successfully logged in, enter the number to call to establish call.
- 7. Call control options are also available once the call is established.

Registratio	'n	Video enabled	
Display Name:	1000	1001	
Private Identity*:	1000		Lingdin
Public Identity*:	sip:1000@192.168.40.214		HangUp
Password:			
Realm*:	Grandstream		
* Mandatory Field Need SIP account? Expert mode?	LogIn LogOut		

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Figure 13: Registration and Call Control

