



Grandstream Networks, Inc.

GVC3200/GVC3200 Conferencing System for Android™

Application Note:

Preliminary Interoperability Test between GVC3200/GVC3200 and
Other Video Conference Systems

Index

INTRODUCTION.....	2
INTEROPERABILITY AND TESTING NOTES.....	3
TESTING WITH CISCO® SX20	3
TESTING WITH POLYCOM® REALPRESENCE® GROUP 550.....	4
TESTING WITH POLYCOM® HDX8000®	5
TESTING WITH LifeSize® Express 220™	6
TESTING WITH telyHD™.....	7
TESTING WITH YEALINK VC400/VC120.....	8
GVC3200/GVC3202 SIP ACCOUNT SETTINGS	10
SETTINGS/SIP/GENERAL.....	10
SETTINGS/SIP/CODEC	11

INTRODUCTION

The GVC3200/GVC3200 is a SIP based video conference system that supports multi-stream H.264 High Profile/Main Profile/Base Profile video with up to 1080p 30fps and a number of standards/protocols such as FEC (Forward Error Correction), FECC (Far End Camera Control), BFCP (Binary Floor Control Protocol), etc. This report is a summary of some preliminary interoperability tests conducted between GVC3200/GVC3200 and some other video conference systems on the market. The content of this report is subject to update from time to time without notice.

INTEROPERABILITY AND TESTING NOTES

This section describes the features tested between GVC3200/GVC3200 and various video conference systems from different vendors. The information and result in this section are based on testing using Kamailio® platform.

TESTING WITH CISCO® SX20

Features	Compatibility	Note
Video Resolution	Compatible	Conference call with up to 1080P resolution, video frame rate 30fps and video bit rate 4Mbps can be established
Presentation Sharing	Compatible	
FEC	Not supported on Cisco® SX20	
FECC	Compatible	GVC3200/GVC3200 and Cisco® SX20 can control each other's camera remotely

TESTING WITH POLYCOM® REALPRESENCE® GROUP 550

Features	Compatibility	Note
Video Resolution	Compatible	Tested conference call with 720P resolution
Presentation Sharing	Compatible, when using BFCP via UDP	Currently GVC3200/GVC3200 doesn't support BFCP via TCP. Polycom® RealPresence® Group 550 has TCP configured for BFCP by default. Please change this setting to UDP so that it can be used with GVC3200/GVC3200
FECC	Compatible	GVC3200/GVC3200 and Polycom® RealPresence® Group 550 can control each other's camera remotely

TESTING WITH POLYCOM® HDX8000®

Features	Compatibility	Note
Video Resolution	Compatible	Tested conference call with 720P resolution
Max Bit Rate	Compatible	When GVC3200/GVC3200 and Polycom® HDX8000® have Max Bit Rate negotiated to 4Mbps, the GVC3200/GVC3200 is sending at about 2Mbps and Polycom® HDX8000® is sending at about 3.5Mbps
Max Frame Rate	Compatible	Up to 30fps
Presentation Sharing	Tested sharing presentation from GVC3200/GVC3200 and Polycom® HDX8000® supports receiving presentation at 4CIF resolution	
FECC	Compatible	GVC3200/GVC3200 and Polycom® HDX8000® can control each other's camera remotely

TESTING WITH LifeSize® Express 220™

Features	Compatibility	Note
Video Resolution	Compatible	The maximum negotiated video resolution is 720P when establishing video call between Lifesize® Express 220™ and GVC3200/GVC3200
Presentation Sharing	Not supported on Lifesize® Express 220™	
FEC	Not supported on Lifesize® Express 220™	
FECC	Compatible	GVC3200/GVC3200 and Lifesize® Express 220™ can control each other's camera remotely
SRTP	Not Compatible	When initiating SRTP call from GVC3200/GVC3200 to Lifesize® Express 220™, there is no response from Lifesize® Express 220™. There is no SRTP configuration found in Lifesize® Express 220™

TESTING WITH telyHD™

Features	Compatibility	Note
Video Resolution	Compatible	The negotiated video resolution is 720P but the actual sent resolution is 960x544 from telyHD™
Presentation Sharing	Compatible	To share presentation on telyHD™, it is required to install telyShare on PC.
FEC	Not supported on telyHD™	
FECC	Not supported on telyHD™	

TESTING WITH YEALINK VC400/VC120

Features	Compatibility	Note
Video Resolution	Compatible	Conference call with up to 1080P resolution can be established
		Note: On the Yealink VC400/VC120, the video resolution is automatically adjusted based on the bandwidth during the call. To achieve 1080P for the conference call, 2M bandwidth for upload and download is recommended
Presentation Sharing	Compatible	Presentation Sharing feature is disabled on Yealink VC400/VC120 by default. Users need to enable BFCP on the Yealink VC400/VC120.
FEC	Not compatible	Both GVC3200/GVC3200 and Yealink VC400/VC120 support FEC but not compatible with each other.
FECC	Compatible	GVC3200/GVC3200 and Yealink VC400/VC120 can control each other's camera remotely
		FECC feature is disabled on Yealink VC400/VC120 by default. Users need enable SIP FECC on Yealink VC400/VC120.

**Note:**

1. The above information is based on testing result on Kamailio® platform, except for Polycom® RealPresence® Group550.
 2. The information for Polycom® RealPresence® Group550 is based on testing result using Broadsoft platform.
 3. When using other SIP server or platform than the ones mentioned in 1 and 2, the testing result could be different and additional configuration on the video conference system might be required. For example, on the SIP server, H.264 payload type might be different from the value on the video conference system.
 4. The presentation sharing feature is implemented using BFCP.
 5. Please make sure the firmware on the GVC3200/GVC3200 is upgraded to the latest version. Please refer to the following link for firmware upgrading information:
<http://www.grandstream.com/support/firmware>
-

GVC3200/GVC3202 SIP ACCOUNT SETTINGS

This section describes options that may be necessary to configure on the GVC3200/GVC3202 when using the GVC3200/GVC3202 SIP account to establish conference call with other video conference systems.

The following configurations can be found under GVC3200/GVC3202 web UI->Settings->SIP. For other SIP account related options, please refer to the GVC3200/GVC3202 administration guide in www.grandstream.com.

SETTINGS/SIP/GENERAL

Parameters	Descriptions
Account Active	This field indicates whether the account is active. If disabled, the device will not send registration information to SIP server. The default setting is enabled.
Account Name	It is used to configure the name associated with each account to be displayed on the upper left corner of LCD.
SIP Server	It is used to configure the URL or IP address, and port of the SIP server. This should be provided by VoIP service providers (ITSP).
SIP User ID	It is used to configure the user account information, provided by your VoIP service provider (ITSP). It's usually in the form of digits similar to phone number or actually a phone number.
SIP Authentication ID	It is used to configure the SIP service subscriber's Authenticate ID used for authentication. It can be identical to or different from the SIP User ID.
SIP Authentication Password	It is used to configure the account password required for the device to authenticate with the ITSP (SIP) server before the account can be registered. After saving, it will appear as hidden for security purpose.
Voice Mail Access Number	This parameter allows you to access voice messages by entering voice mail or dialing access number. On the device, the users can select LCD menu->Applications->Voice Mail and click on the account to dial out to voicemail portal.
Display Name	It is used to configure the SIP server subscriber's name (optional) that will be used for Caller ID display. The configured content will be included in the From, Contact and P-Preferred-Identity headers of SIP INVITE message.

SETTINGS/SIP/CODEC

Parameters	Descriptions
DTMF	<p>It is used to set the parameter to specify the mechanism to transmit DTMF (Dual Tone Multi-Frequency) signals. There are 3 supported modes: in audio, RFC2833, or SIP INFO.</p> <ul style="list-style-type: none"> • In audio DTMF is combined in the audio signal (not very reliable with low-bit-rate codecs). • RFC2833 Specify DTMF with RTP packet. Users could know the packet is DTMF in the RTP header as well as the type of DTMF. • SIP INFO Use SIP INFO to carry DTMF. The disadvantage of this mode is that it's easy to cause desynchronized of DTMF and media packet if the SIP and RTP messages are required to transmitted respectively. <p>The default setting is "RFC2833".</p>
DTMF Payload Type	<p>It is used to configure the RTP payload type that indicates the transmitted packet contains DTMF digits. The valid range is from 96 to 127. The default setting is "101".</p>
Preferred Vocoder	<p>It lists the available and enabled audio codecs for this account. Users can enable the specific audio codecs by moving them to the Selected box and set them with a priority order from top to bottom. This configuration will be included with the same preference order in the SIP SDP message. The codec option includes "PCMU", "PCMA", "Opus", "G.722", and "G.722.1".</p>
Codec Negotiation Priority	<p>It is used to configure the codec negotiation sequence when the GVC3200/GVC3202 is acted as a callee. If set to "Caller", the device will negotiate codec by SDP codec sequence from the received SIP INVITE message. If set to "Callee", the device will negotiate codec according to the codec priority configuration on the device itself. The default setting is "Callee".</p>
Silence Suppression	<p>It is used to set the silence suppression/VAD feature. If it is set to "Yes", when silence is detected, a small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to "No", this feature is disabled. The default setting is "No".</p>

Voice Frames Per TX	It should be noted that the "ptime" value for the SDP will change with different configurations here. This value is related to the codec used and the actual frames transmitted during the in payload call. e.g.: if set to 2 and the first codec is G.729, G.711 or G.726, the "ptime" value is 20ms for the SDP. If the TX exceeds the maximum allowable value, the device will use and save the maximum allowed value according to what the first codec is. For end users, it is recommended to use the default setting, as incorrect settings may influence the audio quality. The default setting is "2".
G.722.1 Rate	It is used to select encoding rate for G.722.1 codec. It supports 24kbps or 32kbps. The default setting is "24kbps encoding rate".
G.722.1 Payload Type	It is used to configure payload type for G.722.1 codec. The valid range is 100-126. The default setting is "104".
Opus Payload Type	It is used to configure payload type for Opus codec. The valid range is 96-126. The default setting is "123".
Use First Matching Vocoder in 200OK SDP	If set to "Yes", the device will use the first matching vocoder in the sent 200OK SDP as the codec. The default setting is "No".
Enable RFC5168 Support	If set to "Yes", the RFC5168 mechanism will be enabled for video call. RFC5168 allows SIP party to request the sender to refresh its video frame in H.264, or refresh the full picture in VP8. The default setting is "Yes".
Enable Video FEC	If enabled, the video sender will temporarily allocate part of the bandwidth to one data channel to send FEC data to system, thus to improve the video quality the receiver gets. Enabling this function will take up part of bandwidth and reduce call rate. The default setting is "Yes".
Video FEC Mode	If set to 0, FEC is not sent by separate port. If set 1, FEC is sent by separate port. Default setting is 0.
FEC Payload Type	It is used to configure FEC payload type. The range is 96-127. The default setting is 120.
FEC RED Payload Type	It is used to configure FEC RED payload type. The valid range is from 96 to 127. The default value is 122.
Enable FECC	If set to "Yes", You can control the camera of the opposite side for video call, but the opposite site must support FECC, and allow remote control on its local camera. The default setting is "Yes".
FECC H.224 Payload Type	It is used to configure FECC H.224 payload type. The valid range is from 96 to 127. The default value is 125.
H.264 Payload Type	It is used to configure the H.264 codec message payload type format. The default setting is 99. The valid range is from 96 to 127.

<p>H.264 Image Size</p>	<p>It is used to set the H.264 image size. It can be selected from the dropdown list.</p> <ul style="list-style-type: none"> • 1080P • 720P • 4CIF • 4SIF • VGA <p>The default setting is 1080P.</p>
<p>H.264 Profile Type</p>	<p>It is used to set the H.264 profile type. It can be selected from the dropdown list.</p> <ul style="list-style-type: none"> • Baseline Profile • Main Profile • High Profile • BP&MP&HP <p>The default setting is "BP&MP&HP".</p> <p>Lower profile is easier to decode while higher profile provides higher compression ratio. For device with low CPU, select "Baseline Profile". Also, "Baseline Profile" is likely to be used in a video conference that has high demanding for the video quality.</p>
<p>Video Bit Rate</p>	<p>It is used to configure the bit rate of the video device. The default setting is 2048kbps. It is recommended to increase bit rate if bandwidth allows. If the bandwidth is not enough, the video quality will be reduced.</p> <ul style="list-style-type: none"> • H.264 Image Size = 1080p, the Video Bit Rate can be set to an integer value from 2048kbps to 4096kbps. • H.264 Image Size = 720p, the Video Bit Rate can be set to an integer value from 1024kbps to 2048kbps. • H.264 Image Size = 4SIF/4CIF/VGA, the Video Bit Rate can be set to an integer value from 384 to 1024 Kbps.

SDP Bandwidth Attribute	<p>Select the SDP bandwidth attribute from "Standard", "Media Level" or "None".</p> <ul style="list-style-type: none"> • Standard: Use AS at the session level and TIAS at the media level. • Media Level: Use AS at the media level. • None: Do not change the format. <p>The default setting is "Media Level".</p> <p>Note: Please do not change the format as it may cause decode failure if the user is unclear about what format the server supports.</p>
Video Frame Rate	Configures the frame rate for SIP video call. The default is 30 fps. Increase frame rate will take up bandwidth; the video quality will be reduced if not allocated enough bandwidth.
Video Jitter Buffer Maximum (ms)	Configures the video buffer size according to the network environment. The valid range is 0-1000, the default setting is 50.
Disable Presentation	If set to "Yes", the device will disable presentation and will not receive presentation during video call. The default setting is "No".
Presentation H.264 Image Size	It is used to select the H.264 image size from "720P" or "1080P". The default setting is "1080P".
Presentation H.264 Profile Type	Select the H.264 profile type from "Baseline Profile", "Main Profile", "High Profile" or "BP&MP&HP". The default setting is "BP&MP&HP". Lower profile type is easier to decode, while higher level has higher compression ratio. For device with low CPU, select "Baseline Profile". Also, "Baseline Profile" is likely to be used in a video conference that has high demand for the video quality.
Presentation Video Bit Rate	<p>Configures the bit rate of the video. The video bit rate can be adjusted based on the network environment. Increasing the video bit rate may improve video quality if the bandwidth is permitted. If the bandwidth is not permitted, the video quality will decrease due to packet loss. The default setting depends on presentation H.264 image size.</p> <ul style="list-style-type: none"> • Presentation H.264 Image Size = 1080p, Video Bit Rate can be set to an integer value from 1024kbps to 4096kbps • Presentation H.264 Image Size = 720p, Video Bit Rate can be set to an integer value from 512kbps to 2048kbps
Presentation Video Frame Rate	Configures the video frame rate for presentation. The options are 5fps, 10fps and 15fps. The default setting is 15fps.

BFCP Transport Protocol	Defines the transport protocol used for BFCP. Users can choose from Auto/UDP/TCP. The default setting is "UDP". If "Auto" is configured, the device will automatically switch between "UDP" and "TCP".
SRTP Mode	<p>It is used to set if the device will enable the SRTP (Secured RTP) mode. It can be selected from dropdown list:</p> <ul style="list-style-type: none"> • Disable • Enabled but not forced • Enabled and forced <p>SRTP uses encryption and authentication to minimize the risk of denial of service. (DoS). If the server allows to use both RTP and SRTP, it should be configured as "Enabled but not forced". The default setting is "Enabled But Not Force".</p>
SRTP Key Length	<p>It is to configure all the AES (Advanced Encryption Standard) key size within SRTP. It can be selected from dropdown list:</p> <ul style="list-style-type: none"> • AES 128&256 bit • AES 128 bit • AES 256 bit <p>If it is set to "AES 128&256 bit", the device will provides both AES 128 and 256 cipher suite for SRTP. If set to "AES 128 bit", it only provides 128 bit cipher suite; if set to "AES 256 bit", it only provides 256 bit cipher suite. The default setting is "AES 128&256 bit".</p>

-
- * ***Android is a Registered Trademark of Google, Inc.***
 - * ***Cisco® is a Registered Trademark of Cisco Systems, Inc.***
 - * ***Polycom® is a registered trademark of Polycom, Inc.***
 - * ***Lifesize is a Registered Trademark or Trademark of Logitech.***
 - * ***telyHD™ and the telyHD logo are trademarks or service marks of Tely Labs Inc.***
 - * ***Yealink is a Registered Trademark of Xiamen YeaLink Network Technology Co., Ltd.***
 - * ***Other trademarks and registered trademarks are the property of their respective owners.***
-