

## Grandstream GXV3140 – IP DoorPhone

**Symmetric RTP** = communication uses the same number of port for incoming and outgoing direction of talk (acoustic). If there is any port blocked on the PC or in the Network, if you hear one direction of talk only, then you need to use the „Symmetric RTP“ parameter in the web tool interface of the IP doorphone.

### Settings of Grandstream GXV3140 for H.263 / H.264 video on the IP DoorPhone



The screenshot displays the web administration interface for a Grandstream GXV3140 IP DoorPhone. The interface is titled "GXV3140 -- Multimedia Phone Administration Interface". At the top, there are navigation tabs: "Status", "Account 1", "Account 2", "Account 3", "Advanced Setting", "Maintenance", and "Application Setting". The "Account 1" tab is selected. On the left side, there is a sidebar menu with "General Settings" highlighted, and other options include "Network Settings", "SIP Settings", "Codec Settings", and "Call Settings". The main content area is titled "General Settings" and contains the following configuration fields:

* Account Active :	<input checked="" type="checkbox"/> Yes
* Account Name :	220
* SIP Server :	192.168.1.80
* SIP User ID :	220
* Authenticate ID :	220
* Authenticate Password :	***
Voice Mail UserID :	*26
* Name :	220
* User ID is phone number :	<input type="checkbox"/> Yes

At the bottom of the settings area, there are two buttons: "Save" and "Cancel".

Status

Account 1

Account 2

Account 3

Advanced Setting

Maintenance

Application Setting

General Settings

Network Settings

SIP Settings

Codec Settings

Call Settings

### SIP Settings



- \* SIP Registration :  Yes
- \* Unregister On Reboot :  Yes
- Register Expiration (m) :
- Wait Time Retry Registration (s) :
- \* Local SIP Port :
- \* SUBSCRIBE for MWI :  Yes
- Session Expiration (s) :
- Min-SE (s) :
- UAC Specify Refresher :
- UAS Specify Refresher :
- Force INVITE :  Yes
- Caller Request Timer :  Yes
- Callee Request Timer :  Yes
- Force Timer :  Yes
- \* Enable 100rel :  Yes
- \* SIP Transport :
- Symmetric RTP :  Yes
- \* Support SIP Instance ID :  Yes
- \* Validate Incoming Messages :  Yes
- \* SIP T1 Timeout :
- \* SIP T2 Interval :
- \* Remove OBP from route :  Yes

Save

Cancel

Status

Account 1

Account 2

Account 3

Advanced Setting

Maintenance

Application Setting

General Settings

Network Settings

SIP Settings

Codec Settings

Call Settings

### Codec Settings



Enable Video :

DTMF :  In audio  RFC2833  SIP INFO

DTMF Payload Type :

Preferred Vocoder :

Available		Selected
G723.1	↑	PCMU
G726-32	←	PCMA
GSM	→	G729A/B
L16-256	↓	G722

Preferred Video Codec :

Available		Selected
	↑	H264
	←	H263
	→	H263+
	↓	

H.264 Payload Type :

H.263+ Payload Type :

L16-256 Payload Type :

H.263 Encoder Resolution :  CIF  QCIF

SRTP Mode :

Silence Suppression :  Yes

Voice Frames Per TX :

G723 Rate :

Jitter Buffer Type :

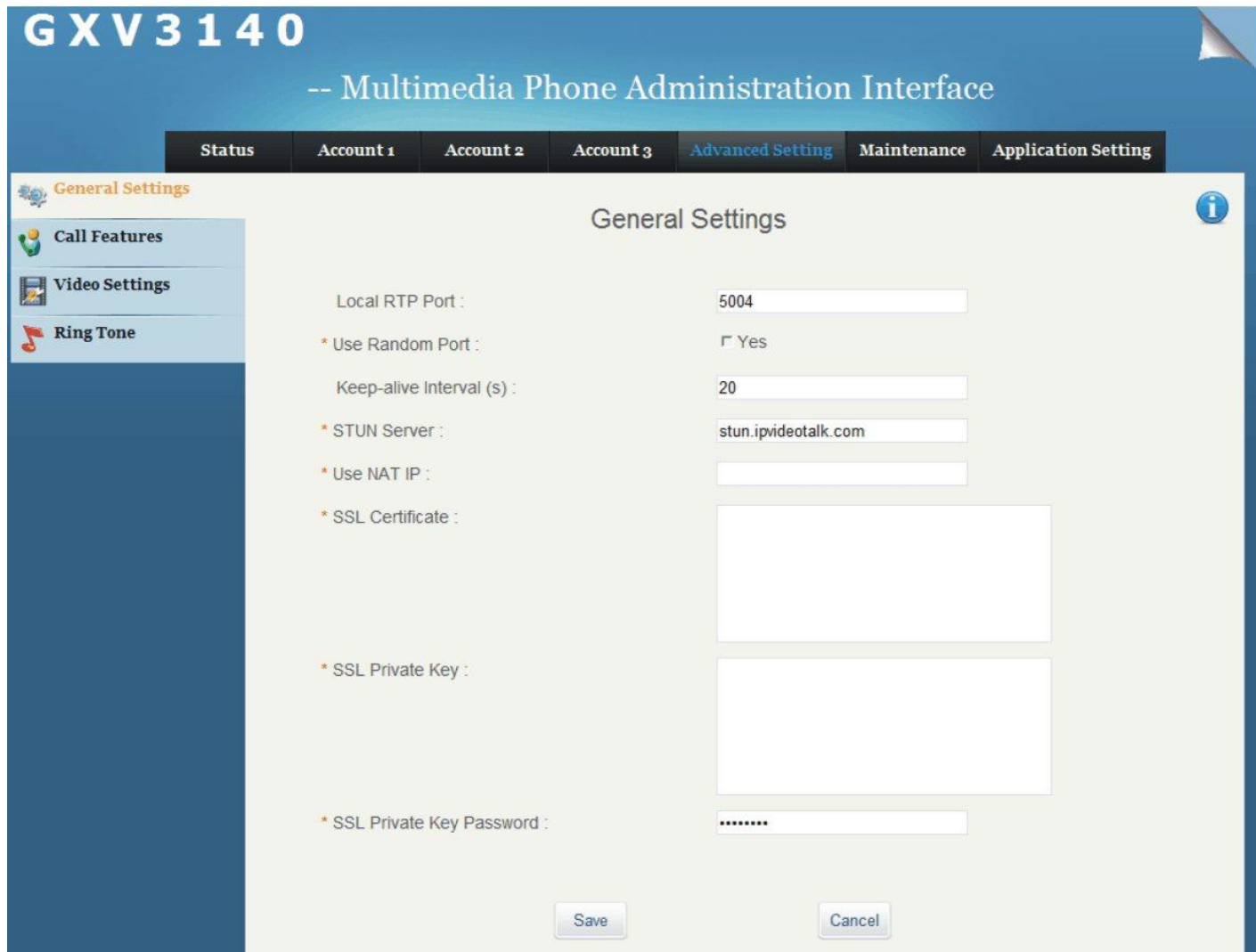
Jitter Buffer Length :

Save

Cancel

## P2P in Grandstream GXV3140

For P2P connection on GXV3140 you need to connect the GXV3140 IP phone to an active account and disable "Use random port". Without registering the GXV3140 IP phone an active account it is impossible to connect it in P2P mode.



The screenshot displays the GXV3140 Multimedia Phone Administration Interface. The interface has a blue header with the title "GXV3140 -- Multimedia Phone Administration Interface". Below the header is a navigation bar with tabs: "Status", "Account 1", "Account 2", "Account 3", "Advanced Setting" (selected), "Maintenance", and "Application Setting".

The main content area is titled "General Settings" and contains the following configuration options:

- Local RTP Port : 5004
- \* Use Random Port :  Yes
- Keep-alive Interval (s) : 20
- \* STUN Server : stun.ipvideotalk.com
- \* Use NAT IP :
- \* SSL Certificate :
- \* SSL Private Key :
- \* SSL Private Key Password :

At the bottom of the settings area, there are two buttons: "Save" and "Cancel".