# **VOPTech V60 Video Phone User Guide**



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Please read the following safety notices before installing or using this phone. They are crucial for the safe and reliable operation of the device.

- Please use the external power supply that is included in the package. Other power supplies may cause damage to the phone, affect the behavior or induce noise.
- Before using the external power supply in the package, please check with home power voltage. Inaccurate power voltage may cause fire and damage.
- Please do not damage the power cord. If power cord or plug is impaired, do not use it, it may cause fire or electric shock.
- The plug-socket combination must be accessible at all times because it serves as the main disconnecting device.
- Do not drop, knock or shake it. Rough handling can break internal circuit boards.
- Do not install the device in places where there is direct sunlight. Also do not put the device on carpets or cushions. It may cause fire or breakdown.
- Avoid exposure the phone to high temperature, below 0°C or high humidity. Avoid wetting the unit with any liquid.
- Do not attempt to open it. Non-expert handling of the device could damage it. Consult your authorized dealer for help, or else it may cause fire, electric shock and breakdown.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean it. Wipe it with a soft cloth that has been slightly dampened in a mild soap and water solution.
- When lightning, do not touch power plug or phone line, it may cause an electric shock.
- Do not install this phone in an ill-ventilated place.
- You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.

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# Introduction

Thank you for purchasing the V60 Android smart video phone. The V60 is a fully featured video phone that provides voice and video communication over the data network. This phone has all the features of a traditional telephone and all gives access to many data service features. This guide will help you easily use the various features and services available on your phone.

## **Smart Video Phone VOPTech V60 Overview**



## With Attachment

Item	Function
Power Adapter	Power supply for telephone.
Network Cable	Used to access network for the phone.
Hands <b>et</b>	Make phone calls with the phone's basic functions.
Handset Cable	Connected with the Handset and the phone.

## **Phone component descriptions**



## Phone elements icon

Key	Function
Q	Headset key. Click it enter the dial interface when the
	phone is in desktop, receive the call using headset mode
	or switch the call to headset mode during a call.

	Option key. You can browse and accomplish all functions of the phone through pressing this key. Also, you can press and hold this key to complete screenshot.
	The hands-free key.
5	Return key. Press this button in the detailed interface, it will return to the previous interface; If it is pressed in the application program interface, the current program will be closed.
<u> </u>	Home key. Press this key, the phone will return to the idle screen.
	The camera.

## **Interface introduction**



Interface illustration picture 1

	Name	Meaning
	SD Card interface	Connect SD Card for saving data.
<b>○</b> <b>○                                  </b>	DC Power Interface	Input: 220V AC Output: 12V DC
	PC interface	Specification RJ45, connect it computer.
	Internet interface	Specification RJ45, connect it to network.
	Handset interface	Specification RJ9, used to connect the Handset to the phone.



Interface illustration picture 2

 Name	Meaning
USB interface.	Connect it to USB disk.
Headset interface.	Specification RJ9, used to connect the Headset to the phone.
HDMI interface.	Interface for high-definition audio and video. A-A interface, A to A interface.

#### • Note:

1. Put the handset line into the handset interface according to the interface illustration.

2. Plug the power adapter into the DC port; poke the other side of the power adapter into an electric socket.

3. Insert one end of RJ45 network cable into the phone's WAN port (Please refer to the interface illustration picture 1) and put the other end into the network equipment. After

that, if the network connection status on the status bar is displayed as<sup>1</sup>, actions such as making some telephone calls and surfing the internet could be done. If the network

connection status on the status bar is displayed as  $\mathbf{r}$ , please verify whether the network was configured correctly and the network cable was plugged in correctly.

## **Touch Screen Description**

You can touch the screen manually to complete the corresponding operation. Here are three ways to use:

Click: Any icon or button can be clicked to realize its function.



#### **Touch screen:**

- Press and hold the standby interface will pop-up choice wallpaper desktop option box, you can according to the prompt to change the wallpaper.
- Support full multi-touch.

#### Slide:

Slide your finger on the screen upward or downward slowly to move the interface on the screen.

On some kinds of screen such as idle desktop, you can switch the desktop just by moving

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your fingers on the screen to left or right. Dragging your finger to slide will not select or activate anything on the screen.

You can just wait or gently press any place of the screen to stop the sliding after your finger rapidly sliding on the screen. Gently pressing or touching to stop the scrolling will not select or activate anything on the screen.

## **Status Bar**

All of the status information about V60 is displayed on the top of the screen. You can click on the notification icon in the left of the status bar or drag down the drop-down list with your fingers to enter the notification panel to further review and deal with all of the information.

Status bar		Meaning
6	Handset mode	Indicates that the phone is in the handset mode.
$\mathbf{O}$	Headset mode	Indicates that the phone is in the headset
	Silent mode	mode. Shows that the phone is in silent mode.
X		This mode can be canceled by directly
20		clicking the mute button when the screen is in the idle interface.
Ô	<b>O</b> Hands-free mode	Indicates that the call is in hands-free
		status.
Ó	Alarm clock	If you set an alarm clock, the alarm

Note: The right icon	of the status ba	r can't be drag.	just the left can.
- · · · · · ·			

		clock icon will display on the status bar.
	Network	Displays that the network connection is
		successful. Tap the icon directly into
	status (successful)	the network settings.
		Shows that the network connected
	Network status	failed. You need to check the network is
맞가	(failed)	properly connected and the parameters
		are configured correctly.
		Displays that the phone successfully
	CDl	identifies the USB device. Tap the icon
SD	SD card	directly to switch into the USB
		application wizard.
_		Shows that there are several missed
č	Missed calls	calls. You can click on the icon directly
_		to switch into the call log.
		Shows that the phone is on "Do Not
		Disturb" mode. Any of the call could be
4	Open the DND	directly rejected before it's ring
5		interface appears. There is only a
		missed call icon leaving in the status
		bar.
65	Call forward	Shows that the call forward function is
<u></u>	Call Iorward	opened.
۵.	Auto answer	Shows that the auto answer function is
<u></u>	<b>Solution</b> Auto answer	opened.

		If a phone number is
		added to the firewall, then open
Black list		the firewall function, all calls about this
	phone number could be rejected	
		directly.
		Indicates that there are several new
	New record	unread phone recordings.

# **Functional applications:**

<b>%</b> Phone	Click this icon, and it will switch to pre-dial interface. You can make some phone calls through the screen or keyboard.
<b>R</b> Email	Has the function of sending and receiving e-mail. When an account has been configured successfully, you can send and receive your e-mail on the phone and this account will automatically sync the contacts to the mailbox account.
<b>Settings</b>	It contains System, Network, Account, Call, Display and so on. You can configure some settings.
<b>EX</b> Contacts	Support functions such as search, add, remove and edit.
Android settings	It contains the Call settings, Basic settings, Advanced settings, VOIP and so on, you can configure some settings in the corresponding menu.(Android system settings)

Message	Like the phone, with messages to write, read and send function.
MWI	All calls will be transferred into voice mail when the MWI is enabled.
Calendar	Enter into the calendar and you can view the accurate date.
Æ	Enter the call log and you can view all call records. You can also view the "Incoming Calls", "Outgoing Calls" and "Missed
Call log	Calls" records by pressing the Option key.
Import and Export Managemen t	Have export contacts, add blacklists and outgoing call barring functions.
<b>PPPOE</b>	Connect PPPoE.

## Update of the phone

Please contact with service provider to get the new version to upgrade.

Strongly recommended:

**1**) Before the upgrade the new version, please send personal data backup to mobile devices,

to prevent accidental loss.

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**2** - Once the upgrade process, please do not power blackout, equipment may not be able to start.

**Auto provision:** Keep phone registered successfully and configure parameters about automatic updates such as server address, upgrade file name, time interval and so on. And then the phone will check for updates from the server at a fixed time intervals and upgrade itself. See the specific automatically upgrade instructions, please switch to Update.

Manual upgrade---- USB: Will upgrade package in U disk and inserted the U disk into the USB interface (the content of the need to be updated). Before the phone restart, at the same time hold down the "\_\_\_\_\_\_" and "\_\_\_\_\_", then insert the power supply and will enter the unit at this time to Recovery mode, you can see under the Recovery operation menu, shown below:



Through the hardkey " Q down", " T up", " OK", to operate on the Recovery menu.

Select "apply update from udisk" or "apply update from external storage" (SD card), enter the sub-directory and select the upgrade packages, click on "OK" key to start upgrade .After the success of the upgrade, there will be prompt on interface, the user can immediately return to the main menu select "reboot system now" to restart the system to start using the new version.

Notes: In some cases after upgrading, in order to avoid new and old version data of incompatible problems, needs to remove user data and system cache (the so-called double wiper) whether you need specific clear, please refer to the Release Note.

#### Web Update :

Input the phone's IP on the web;

Input the user name and password to phone's web;

Enter "MAINTENANCE"—"UPDATE";

Click the button "Browser" to select the image on your PC and click the button "Update"; Waiting, the system will be update success.

# Dss key

**Introduction :** This module contains five expansion module, 140 editable key. You can click "edit" or press and hold the key to edit it. Line1~6 and release are the default keys, but you can also edit them. Like the below picture:

	<b>(</b> ) 11:59
Line 1	🎢 Edit
Line 2	🧆 MWI
Line 3	Contacts
Line 4	Call log
Line 5	Forward
Line 6	Redial
Release	S Collapse

Туре
------

#### Meaning

Clicking on the button you can dial the mapped number directly.

Subscribe: BLF, Presence, Speed Dial and Intercom

Pickup number : the number which server set.

**Pickup:** That is, when A calls B, B ringing but no one answered the call, C could dial a number which is comprised of specified prefix and B's number, and then C can talk with A.

Memory Key BLF: It used to prompt you the state of the subscribe user, and that could pick up the subscribed number by the state. BLF help you monitor the state of subscribe user (idle, ringing, a call).

**Presence:** Compared to BLF, the Presence is also able to view whether the user is online.

Note: You cannot subscribe the same number for BLF and Presence at the same time

**Speed dial:** You can call the number directly which you set. This feature is convenient for you to dial the number

	Intercom: T	equently dialed. This feature allows the openet the phone quickly;	
Line	The button n	nap the sip line, it enter t ult sip line when you click t	1
	The button m	ap some Key Event。	
Key Event	Title: Consis	stent with the subscription r	name
	Subtype: So	me basic keys	
DTMF	-	keypad of the phone, allow mber or input numbers during	
URL	It maps the w	ebsite; click it to open the U	JRL directly.
Multicast	It maps the multicast.	Multicast; you can click	t it to make a
304 😴 📱 🗔		• • •	<b>(</b> )) 15:32
Line 1	BLF	UTMF456	End
Line 2	Presence	e http:// www.baidu.co	🔙 Dialpad
Line 3	MWI	Mcast	🔇 New call
Line 4	Intercom		Conference
Line 5	Call Park		Hold
Line 6	Call Forward		Transfer
Release	DND		S Collapse

The LED status of BLF

LED Status	Description
Steady green	The object is idle.
Slow blinking red	The object is ringing.
Steady red	The object is active.
Fast blinking red	The object has failed.
Steady orange	The subscribed number is in a call with current line.
Off	Not subscribed.

#### The LED status of Presence

LED Status	Description
Steady green	The object is online.
Slow blinking red	The object is ringing.
Steady red	The object is active.
Fast blinking red	The object has failed.
Off	Not subscribed.

Note: Please set Type first, then set subtype and value base on the introduction.

You can select value from cantacts by clicking the contact icon. Save the configuration and click complete in right side.

You can delete the dsskey by the same method, edit->clear->save->complete.

# Contact

## Add new local contact

- 1. Tap the contacts button.
- 2. Tap "+" in the upper-right corner.
- 3. Select "Keep Locally".

4. Add a picture for a contact. Tap into the picture selection interface to select

photos in your list of photos when editing contacts. It will automatically return to the editing interface after you save it. If you want to change the picture, tap the Contacts icon to select "use this photo", "delete the icon" or "change the icon".

5. You can click the little arrow to expand the name what you want to edit.

6. Follow the prompts to complete the edit contacts, you can also select save to Dsskey. Click "Done" to save it.

		<b>၍</b> ၈) 09:34
🗸 Done 🗌	Save To DSS Key	
Phone-only, uns	synced contact	<b>*</b>
Jasmine		
Fanvil		Take photo
JM		Choose photo from Gallery
PHONE	23 21	work
🕂 Add new		
EMAIL	jasmine.jiang@fanvil.com	номе

## Add account

- 1. Tap the contact button;
- 2. Tap "+" in the upper-right corner.
- 3. Select "Add account".
- 4. Edit exchange account and follow the prompts to complete the editing.

Add an Exchange account	
You can set up an Exchange account in just a fe	ew steps.
Shawns@voptech.com	
	Next

## **Creat group**

- 1. Tap the "Group" button;
- 2. Click "+" in the upper-right corner;
- 3. You can seclect "Create local group" or "Create account group";
- 4. Edit group name, then pick members from contact list;
- 5. Click "Done" to save the group.

🔜 📴 🥸 🖀			<ul><li>11:48</li></ul>
🗸 Done			
Support	[	Pick members from contact list	
	Frank		Î
	Jasmine		
	Tony		

## Favorite

- 1. Tap the "☆Favorite" button;
- 2. Select the contacts what you want to set, or click



## **Network Phonebook**

#### Set the remote phonebook

- 1. Click the edit icon in the upper-right;
- 2. Click URL;
- 3. Follow the prompts to fill in the information.
- 4. Click "Done" to save it.

🖬 🚅 🛠 🚆		<b>1</b> 1:58
🗸 Done		
Phonebook Name:	Phonebook Name	
Server Url:	Server Url	
User:	Please input user name,User name	
Password:	Please input Password,Pas	

to select all the contacts.

## Set the LDAP

- 1. Click the edit icon in the upper-right;
- 2. Click LDAP;
- 3. Edit the information following the prompts;
- 4. Click "Done" to save it.

🖬 🖻 🔇 🚆				15:28
🗸 Done				
Display Title:		Use SSL		
Server Address:	172.16.1.3	Server Port:	389	
Authentication:	DIGEST-MD5	<b>A</b>		
Username:	may	Password:	_ · · · · ·	
Search Base:	dc=winline,dc=com	A		
Telephone:	telephoneNumber	Mobile:	mobile	
Other:	other	Display Name:	<u>cn sn ou</u>	
Enable calling search:				
				_
C Find	contacts			
chrissie 姜瑞		4130		¢
xinyu 宋聚坡		4060		Ç
Test TestSn		4060		Ç
Test TestSn		4060		6
adelle.tony 王琦 tes	t	21971		6

## **Block List&Call Barring**

You can enable Blacklist, click the "+" to add number or contacts what you want. Using the same method to set the whitelist and Call Barring.

# **Phone Settings**

## System info

Tapping "Settings" can directly enter the default window "System info". It displays the current system information state.



System info interface

Configuration item	Meaning
	Displays the connection status of the PPPoE, there
PPPoE	are three display states : Disabled, Connecting and

	the IP address.	
Connect mode	Displays the selected network mode, DHCP or	
	Static IP.	
	Displays the IP address of the current network	
IP address	model.	
Subuct models	Displays the Subnet Mask of the current network	
Subnet mask	model.	
ID Cataway	Displays the default gateway of the current network	
IP Gateway	model.	
Primary DNS	Displays the primary DNS server address of the	
	current network model.	
	Displays the standby DNS server address of the	
Secondary DNS	current use of the network model.	
MAC address	Displays the current MAC address.	
Version	Displays the current version of the phone.	
Phone mode	Displays the current phone model.	
	Displays the current configuration of the Accounts	
Account	and more detailed information will be displayed on	
	the desktop widget.	

## Network

#### Network

Tap "Network"----> "WAN mode" interface, the default interface is "DHCP": click "Save"

button to save successfully after the configuration of each interface is finished. Then a tip box "Config saved" will be shown.



#### **Static IP interface**

Configuration	Meaning
DUCD	You can select to use DHCP, which means whether
DHCP	Use the DNS to connect network assigned by the

	DHCP server.
	IP address: Input your assigned IP address.
	Subnet mask: Input your assigned subnet mask.
	IP gateway: Input your assigned gateway.
Static IP	Primary DNS: Input your assigned DNS server
	address.
	Secondary DNS: Input your assigned Alert DNS
	server address.

#### PPPoE

Tap "Network" ----> "PPPoE". After configuring the parameters, you can click "Save" button to save.

WAN	QoS	Port	STUN	🗩 Status
Network mode		DHCP Static IP PPPoE		Network
User Password		user123		Call
Auto connect on pow	/er up			Display
Auto reconnect				Tone
	Cancel	Save		O Time&date

**PPPoE** interface

Configuration	Meaning
РРРоЕ	User: Input your assigned PPPoE Username.

Password: Input your assigned PPPoE Password.
Auto connect on power up: Whether to automatically connect PPPoE on power up.
Auto reconnect: When connecting failed whether to automatically connect PPPoE.

## QoS

Tap "Network"----> "QoS", after selecting or modifying the default settings click the "Save" button to save

٦			1	
WAN	QoS	Port	STUN	💷 Status
Enable DSCP				S Network
Audio RTP DSC	P	46		Accounts
Video RTP DSC	P	46		
SIP DSCP		46		Call
Enable WAN po	rt VLAN			🖳 Display
WAN port VLAN	ID	256		🔟 Tone
000 10		<b>^</b>		💽 Time&date
	Cancel	Save		

#### **QoS-DSCP** Settings

DSCP is one standard of QoS. It can set the priority of Voice, Video and Signal.

<b>Configuration item</b>	Meaning
QoS-DSCP	Enable DSCP: Enable/Disable DSCP.
	Voice DSCP: Set the number of Voice DSCP.

# Video DSCP: Set the number of Video DSCP.Signal DSCP: Set the number of Signal DSCP.

Slide down the screen to the QoS-WAN VLAN page.

VLAN is Virtual Local Area Network.

Enable VLAN, then you can set VLAN ID, the range of ID number is  $0\sim4095$ ; you can also set the priority of 802.1p, the range is  $0\sim7$ .

Note: You must enable DSCP if you want to set 802.1P priority.

Configuration item	Meaning
	Enable WAN port VLAN: Enable/Disable WAN
	port VLAN.
	WAN port VLAN ID: Set the number of VLAN
QoS- WAN VLAN	ID.
	SIP 802.1P priority: Set the priority number of
	SIP 802.1p priority.

Slide down the screen to QoS-Port VLAN interface

Port VLAN is based on port VLAN, in the same VLAN port to communicate with each other.

<b>Configuration item</b>	Meaning
	Port VLAN mode: Select the status of LAN Port
	VLAN.
QoS- Port VLAN	➢ Follow WAN: Follow the ID number of WAN.
	Disable: Disable Port VLAN.
	Enable: Enable Port VLAN and you can set the

# LAN Port VLAN ID different from WAN. LAN port VLAN ID: Set the number of LAN Port VLAN ID.

## Port

Tap "Network" ----> "Port", after selecting or modifying the default settings click "Save" button to save.

WAN	QoS	Port	STUN	💷 Status
RTP port rang	e start	10000		Network
RTP port quan	tity	200		Accounts
				Call
				🖳 Display
				🔟 Tone
	Cancel	Save		O Time&date
		Port Settings	,	
Configuration item	Meaning			
	RTP port	range start(10	000-60000):	Set the
	telephone'	's RTP start por	t. This port is	s distributed
Port	to dynami	c allocation.		
	<b>RTP</b> port	quantity: Set t	he maximum	n number of
	allocated I	RTP port. The d	efault is 200	

#### Stun

Tap "Network" ----> "Stun", after selecting or modifying the default settings click "Save" button to save.

WAN	QoS	Port	STUN	💷 Status
Server address		Server addres		S Network
Server port		3478		Accounts
				Call
				📃 Display
				🚺 Tone
				💽 Time&date
	Cancel	Save		

## **STUN interface**

<b>Configuration item</b>	Meaning		
STUN	Server address: Configure the SIP STUN server		
	address.		
	Server port: Configure the SIP STUN server		
	port .The default is 3478.		

## Account

Tap "Accounts"---->"Account1", the default password is 123456, you can slide down to set more parameters follow the prompts, after all the parameters are set, you can click "Save" button to save.

(Note: The Account1, 2, 3, 4, 5, 6 has the same settings.)

Account1 Account2 Account3	Account4 Account5 Account6	🔛 Status
Enable registration		🚳 Network
Server address	Server address	Accounts
Server port	5060	Call
Authentication user	Authentication user	
Authentication password	Authentication passw	🖳 Display
SIP User	SIP User	🔟 Tone
		💽 Time&date
Cancel	Save	

Account page 1

<b>Configuration item</b>	Meaning
	Enable registration: Enable/Disable registration.
	Disable register, Enable register.
	Server address: Set your SIP server address.
	It supports the address in the form of domain
Account	name.
	Authentication user: Set your SIP account.
	Authentication Password: Set your SIP
	password.
	SIP user: Input the phone number assigned by
	your VoIP service provider. Phone will not
	register if there is no phone number configured.

Account1 Account2 Account3	Account4 Account5 Account6	(
)		運 Status
SIP User	SIP User	S Network
Display name	_Display name	Accounts
Domain realm	Domain realm	Call
Proxy Server Address	Proxy Server Address	
Proxy Server Port	5060	🖳 Display
Proxy User	Proxy User	🔟 Tone
		🜀 Time&date
Cancel	Save	

#### Account

**Display name**: Configure the display name, it allows the English alphabet input (does not support Chinese).

**Domain realm**: Configure the SIP domain name (You do not need to configure it because the system will configure automatically).

**Proxy Server Address:** Configure the proxy server address.

**Proxy Server Port:** Configure the proxy server address.

**Backup Proxy Server Address**: Configure a backup server address. When the primary server is not connected, you can use the backup server for calling communications (The backup server can be connected only when the primary server cannot connect); When the primary server is connected, the phone will automatically switch back to the primary server to communicate. Backup server port: Configure the backup server port.

**Backup Proxy Server port**: Configure the SIP register proxy server port.

**Server name**: Configure the SIP register server name.

**Registration expire(s)**: Configure the server registration expire(s), default is 60 seconds. If the registration time of the server required is greater or less than the time of the phone to configure, telephone can automatically modify to the time limit that server recommended, and register again.

DTMF Type: Set DTMF mode, there are four types: In-band RFC2833 SIP\_INFO AUTO Different server vendors can provide different models. RFC protocol edition: Configure the protocol

version of the phone. When your phone needs to
communicate with gateway which uses SIP1.0, such as CISCO5300, you need to configure to RFC2543 to conduct normal communication. RFC3261 is used by default.

Anonymous call edition: Configure whether to use anonymous security call. It supports RFC3323 and RFC3325.

**Transport protocol**: Configure the transport protocol, TCP, UDP or TLS.

**Ban anonymous call**: Configure whether to ban anonymous call.

**Enable strict proxy**: Compatible with special server. (Using source address of opposite side when return message, no longer use the address in via field).

**Subscribe for MWI**: After successful registration, you can subscribe to information, such as someone else's status or voice mail and so on.

**Enable rport**: Configure whether to support RFC3581, rport mechanism is used in internal network, and requires a SIP server to support, it is used for keeping the NAT connect of the internal network and the external network.

Enable PRACK: Configure whether to support

SIP-PRACK function (used by "Color Ring Back
Tone") Recommend: Use the default
configuration.
Convert URI: When you send URI it will
convert # to %23.
Enable DNS SRV: Configure whether to enable
RFC2782 protocol edition.
Enable display name quote: Set to make
quotation mark to display name as the phone
sends out signal, in order to be compatible with
server.
Use Stun: Enable/Disable SIP STUN.
Enable BLF List: Enable/Disable BLF List.
BLF list number: Input the number of BLF List.
Enable session timer: Enable/Disable session
timeout.
Session timeout(s): Set the time of session
timeout.

# Call

Tap "Call" enter the interface, after configuring the parameters of each interface, you can click "Save" button to save.

#### General

Tap "Call" ——>"General" interface, after configuring the parameters, you can click "Save" button to save.

General Dial plan	Account1 Account2 Account3		💷 Status
DND (Do Not Disturb)	Disable	-4	S Network
Enable call waiting Enable Call Waiting Ton	ne 🗹		Accounts
Enable password dial			Call
Password dial prefix	Password dial prefix		🖳 Display
Password length	0		Tone
Hide DTMF	Disabled		Tone
	Cancel Save		C Time&date
	General interface		

Configuration	Meaning		
	<b>Do not disturb</b> : It will not allow any phone call while		
	enable the function, but there will have prompt in the missed calls.		
General	Enable call waiting: Whether enable call waiting.		
	Call waiting tone: If enable the function, there will have		
	call waiting tone if there has a call waiting.		
	<b>Enable password dial</b> : Whether enable the password dial.		
	Password dial prefix: Set the prefix of the number.		
	Password length: After setting the length successfully,		

the number dialed out will hide the appropriate length .For example, sets the prefix number to 138, the password length is 5, and dial the number 138142658941, then it will display as 138\*\*\*\*8941 on the dial interface.

Hide DTMF: It has four choices: Disable, All, Delay

and Last show.

Disable that is the content you input is clear text and it can be seen.

All that is the content you input is hidden immediately and displayed as "\*"

Delay that is the content you input will display in clear text first and then displayed as "\*"

Last show that is the last one inputted will displayed in clear text and the others will displayed as "\*".

Ban outgoing: Forbid call out.

Enable call transfer: Enable /disenable the function, if

disable the function, the phone can not do transfer.

Semi-attended transfer: During the talk, press Transfer

firstly, and then press Send after inputting the number that you want to transfer to. You are waiting for connection, now, press Transfer and the transfer will be done. (To use this feature, you need enable call waiting and call transfer first).

Enable Auto on Hook: Enable this function, and set the

auto on hook time, the phone will auto return to previous interface after the time when end a call.

**Enable 3-way Conference:** You can make a conference when enable this function.

**Enable call completion**: Enable Call Completion by selecting it, If the callee is busy, the sip server will inspect the callee status at intervals. If the callee is idle, the server will send notify message to inform the caller whether redial.

Enable auto redial: Enable this function, calling out

someone, if he is in busy, it will pop up a prompt box whether to auto redial, press OK, the phone will call out him according the Interval and Times that you set.

Auto redial interval: Specify the Auto Redial interval.

Auto redial times: Specify the Auto Redial times.

Enable intercom : Enable Intercom Mode by selecting

it. Then you can set up connection to the operator or the secretary quickly, and it widely used in office environments.

Enable intercom mute: Enable mute mode during the

intercom call.

**Enable intercom tone:** If the incoming call is intercom

call, the phone plays the intercom tone.

Enable intercom barge : Enable Intercom Barge by

selecting it, the phone auto answers the intercom call during a call. If the current call is intercom call, the phone will reject the second intercom call.

**Update dial call:** Enable this function, the phone will update the peer display name based on the 2000K packet.

Ring from headset : Enable Ring From Handset by

selecting it, the phone plays ring tone from handset.

**Enable Hide Local Area Code**: Enable /disenable the function, if enable the function it will hide the local number.

**Auto Hold:** when you call or receive the second way calls, the phone will hold the first way call.

**Country code:** Input the country number, such as the number of China is 86.

Area code: input the region number, that is the area number, for example, the number of Beijing is 10. Call Waiting Code: You can choose the Call waiting response, 180 or 182. Default Ext Line: You can select any sip to be the default line, the number will display in the upper-left. Default Dial Mode: Choose the default mode when you call other. Default Ans Mode: Choose the mode how you receive the call. Accept video mode: You can choose "Manual accept video", "Auto accept video" or "Auto reject video". Enable Record: You can record the conversation when enable this function. Use SIP numbering plan: Enable this function, you can register the sip number with letters and characters. Configure Emergency Number: You can configure one or more number here, and call them even if the screen lock. Enable 3rd IM App: Enable this function, you can install the third instant messaging app. **IM** App: If you select the third app, when you pick up handset or press Headset/Handsfree key, the phone will use the app to make a call.

#### **Dial plan**

Tap "Call"----> "Dial plan" interface. After configuring the parameters, you can click "Save" button to save.

General Dial plan Accou	nt1 Account2 Account3	🖼 Status
Enable E.164		S Network
Press # to Send		Accounts
Send after(3-30)		
	5	
Press # to Do Blind Transfer		🖳 Display
Blind Transfer on Onhook		
Attended Transfer on Onhook		🚺 Tone
Attanded Transfer on Oanfara		C Time&date
Canc	el Save	

## Dial plan interface

<b>Configuration item</b>	Meaning
	Use E164: Enable/Disable E164. After it is
	enabled, it will directly dial numbers according
	with E164 rules.
	<b>Press # to Send</b> : Enable/Disable. After enabling
	the function, input the number end with "#" in the
Dial plan	non-pre-dial mode, it will directly dial the phone
	number.
	Press # to Do Blind Transfer: Enable/Disable.
	After enable this function, when you do a blind
	transfer, you can input number end with "#" do
	send it.

**Blind Transfer on Onhook**: Enable/Disable. After enable this function, when you end input the transfer number, you can directly on hook the phone to do the blind transfer.

Attended Transfer on Onhook: Same as above, but do the attended transfer.

Attended Transfer on Conference: Enable this function, when you host a 3-way conference, you can on hook to do the attended transfer.

**Dial prefix**: Enable/Disable. After enabled the function, it will be automatically added the outside line prefix number before the outgoing number when you are going to call.

#### Method to use the prefix: Example

If you set the prefix is 135856, and you want to dial the number 13585679801. Then when you dial the number, you can press and hold 0, when the "+"appears, you can just input 79801and send the number, the "+79801" is the numb you input, but you can see the dialing number is 13585679801.

#### Account

Tap "Call" ——>"Account1 $2\3\4\5\6$ " interface, after configuring the parameters, you can click "Save" button to save.



Configuration	Meaning		
	Enable DND: When enable the DND line, you can		
	choose to enable DND here to control this account		
	reject the incoming call.		
Account 1	Enable Always Forward: Set the forward number		
	below and enable this function, when it has a coming		
	call, the phone will forward the call to the setting		
	number and display one missed call.		
	Enable Busy Forward: Same as the above function,		
	but when the phone has one call it can forward the		
	incoming call.		

**No Answer Forward Number**: Same as the above, when the phone does not answer the call after timeout, it will forward the incoming call.

**MWI number**: Fill in the MWI number.

**Enable Hotline**: Enable this function, configure the hotline number and time, the phone will auto call this number when enter the dialing interface and timeout.

**Enable auto answer:** Enable auto answer and set the timeout, when the phone has incoming call, it will auto answer when time out.

**Enable missed call log**: Enable this function, the status bar has prompt when missed incoming calls and save them to call log history .

**Caller ID Type:** It is default display long number, and support standard ISP number .If not enable, the phone will display last few numbers matched.

**Enable user=phone:** Enable this, the phone can make call directly, if not, you need to precede the phone number with country code and Area code, Depending on your environment.

**Dial without registered:** Set call out by proxy without registration.

#### **Display**

Configuring the phone screen displays a number of parameters, including screensavers and power indicator

After enable operator mode, if you press the home key, the main interface will enter dss key interface. If disable this function, the phone will return to desktop interface when press home key.

When enable shortcuts, you can press and hold the application to set shortcuts to desktop.

Tap "Display" ----> "Display" interface. After configuring the parameters, you can click "Save" button to save successfully.

#### Tone

Configuration of the telephone voice parameters, including selecting a ringtone, SMS notification tones, keypad tones, and other

Tap "Settings" ----> "Sound" interface. After configuring the parameters, you can click "Save" button to save successfully.



## Time&Date

Tap "Settings" ----> "Date time" interface. After configuring the parameters, you can click "Save" button to save successfully.

Enable SNTP				🗭 Status
Primary Server		209.81.9.7		
Timezone		GMT+08:00, China	a	S Network
a mice one		Standard Time		🙇 Accounts
Time format		🔘 Use 12-hour clock		
		💿 Use 24-hour clock		Call
Date format		2013/12/31		😨 Display
		0 31/12/2013		
		0 12/31/2013		🔟 Tone
				🕑 Time&date
	Cancel	Save		

## Maintain

Tap "Settings" ----> "Maintain" Enter the interface.

## Service port

Tap "Maintain"---->"Service port" interface. After configured the parameters, you can click "Save" button to save successfully.

	Y I		) )		-
Upgrade ice port	Password	CWMP set	Backup	Update	Call
Enable telnet					🖳 Display
Enable temet					
Web server type		НТТР			Tone
HTTP Server Port		80			🕒 Time&date
					🔀 Maintain
					関 Audio
					💽 Video
	Cancel				Coffkow

Service port interface

Configuration	Meaning
Service port	Enable telnet: Open telnet function.
	Telnet is a common method of remote control of
	the Web server, end users can enter the command
	telnet program, these commands will be run on the
	server, just like directly on the server console, enter
	the same. Can locally be able to control the server.

Web server type: Select "HTTP" or "HTTPS".After selecting one server, you can use the selected server to login the telephone's web page.Such as: https://192.168.1.20HTTP port: Input server port, default is 80.

#### **CWMP set**

Tap "Maintain"---->" CWMP set" interface. After configured the parameters, you can click "Save" button to save successfully.

Upgrade Service pc Pas	ssword CWMP se	t Backup	Update	Udil
				🖳 Display
Enable TR069				Diopidy
Enable TR069 Warning To	ne 🗹			🚺 Tone
ACS Server Type	China Tele	com	▲	🕒 Time&date
ACS Server URL	0.0.0.0			
ACS User	ACS Us			X Maintain
ACS Password	ACS Pa	ssword		🛐 Audio
CPF Serial Number	CPF Ser	ial Numher		💽 Video
C	ancel			Cofflor

**CWMP set Interface 1** 

Upgrade port Se Passw	vord CWMP set Backup Updat	e Call
בוומטופ דחטספ אימודווווע דטוופ		🖳 Display
ACS Server Type	China Telecom	🚺 Tone
ACS Server URL	0.0.0.0	🕒 Time&date
ACS User	ACS User	
ACS Password	ACS Password	🔀 Maintain
CPE Serial Number	CPE Serial Number	🛐 Audio
TR069 Auto Login		<b>o</b> Video
Canc	el Save	Cofflar

**CWMP set Interface 2** 

Configuration	Meaning			
	Enter the password to access the network menu landing configuration interface.			
	Enable TR069: Enable network mode 0 select China Telecom or Normal.			
CWMP set	ACS Server: Input the address of the ACS server provider.			
	Account: Input service provider assigns the account.			
	Password: Input the account corresponding password.			
	Serial Number: Input the Serial number.			
	Auto connect: When enabled, you can fill out the account and password.			

## Backup

Tap "Maintain"---->" Backup", the default password is 123456, enter the backup interface.

Upgrade ort Servi	Password CWMP set Backup	Update 🗾 너 🕬			
)	) )	Display			
	Reset Phone	🔟 Tone			
		O Time&date			
	Backup	🔀 Maintain			
	Recovery	🚼 Audio			
		💽 Video			
	Cancel	- Cottkow			
	Backup Interfac	e			
Configuration	Meaning				
	Reset Phone: This	will erase all data from yo			
	phone's internal stora	age.			
Backup	Backup: The contact data, phone settings data and				
Ł	calendar backup to a	calendar backup to a specified folder.			
	<b>Recovery:</b> Recovery has backup contacts, phone settings configuration.				

# Update

Tap "Maintain"---->" Update", enter the update interface.



#### Audio

Touch "Settings" ----> "Audio", enter the detailed view.

#### Audio

Touch "Audio"----> "Audio" interface. This is global setting, applies to all the sip lines. After configuring the parameters, you can click "Save" button to save successfully.

		8			
Audio		Αι	udio param	J	Call
					🖳 Display
First codec		G.711A			
Second codec		G.711U			Tone
Third codec		G.722			💽 Time&date
Fourth Codec		G.729AB			<b>N</b>
Enable VAD					🔀 Maintain
					😲 Audio
					🖸 Video
	Cancel				- Cofflor

#### Audio interface

Configuration item	Meaning
	Enable VAD: Whether to enable VAD. (Voice
Andia	Activity Detection); If you enabling VAD function,
Audio	then the G.729 payload length cannot be set greater
	than 20ms.

## Audio param

Touch "Audio"----> "Audio param" interface. After configuring the parameters, you can click "Save" button to save successfully.

Audio		Au	dio param		Gail
G.729AB Payload L	ength	20ms			厚 Display
DTMF payload type		101			🔟 Tone
ILBC payload type		97			🕒 Time&date
AMR payload type		108			🔀 Maintain
ILBC payload length G.723.1 bit rate		20ms 5.3kb/s		4	🛐 Audio
		● 6.3kb/s			💽 Video
	Cancel				Coffloy

Audio param interface

## Video

Touch "Settings" ----> "Video" and enter the settings interface.

#### Video param

Touch "Video"----> "Video param" interface. After configuring the parameters, you can click "Save" button to save successfully.

Note: If you want More clear and smooth video screen, please select H.264 within 2M bandwidth.

Video param			Bandwidth		Gui
Video codec		H.264		-	🖳 Display
H.264 payload type		_117			🔟 Tone
					🕒 Time&date
					🔀 Maintain
					臂 Audio
					O Video
	Cancel	Save			Cofflow

Video param interface

### Bandwidth

Touch "Video"----> "Bandwidth" interface. After configuring the parameters, you can click "Save" button to save successfully.



**Bandwidth interface** 

Configuration item	Meaning
	Video bit rate: Set receiving video bandwidth in
	video calls. You can choose 11 kinds of video
	bandwidth types ("64Kbps", "192Kbps",
	"256Kbps", "384Kbps", "512Kbps", "768Kbps",
	"1Mbps", "1.6Mbps",
Bandwidth	"2Mbps","3Mbps"or"4Mbps").
	Video Resolution: Set video encoding resolution
	in a video call. You can choose 4 types of video
	resolution ("QCIF (176*144)", " CIF (352*288)"
	"VGA(640*480)"or " 4CIF (704*576)").

## Softkey

Tap "Settings->Softkey" enter softkey interface. Select softkey-screen first, then add or delete the softkey by clicking the arrow, you can also click the arrow up or down to adjust the softkey's position.

SoftkeyScreen		Call Dialer	 O Time&date
UnSelected Softkey	s	Selected Softkeys	🔀 Maintain
None		MWI	🛐 Audio
Audio		Contact	💽 Video
Cancel	+	Call logs	📰 Softkey
Call Back		Video	K MCAST
Dial		Forward	Reboot
	Cancel		

Softkey interface



For example, the below picture shows the call dialer softkey. It can show up to six keys.

### MCAST

This feature allows you to make some kind of broadcast call to people who are in multicast group. You can configure a multicast Dss Key on the phone, which allows you to send a Real Time Transport Protocol (RTP) stream to the pre-configured multicast address(es) without involving SIP signaling. You can also configure the phone to receive an RTP stream from pre-configured multicast listening address(es) without involving SIP signaling. You can specify up to 10 multicast listening addresses.

Tap "Settings->MCAST" enter the multi-cast interface, you can follow the prompts to fill in the name and host, you can also set the priority for the call.

MCAST Setting	Meaning
Normal Call Priority	Define the priority of the active call, 1 is the highest
	priority, 10 is the lowest.
<b>Enable Page Priority</b>	The voice call in progress shall take precedence over
	all incoming paging calls.
Name	Listened multicast server name

**Call Dialer softkey** 

Host:port	Listened multicast server'	s multicast IP address and
	port.	

#### Reboot

Tap "Maintain"----> "Reboot" interface. You can click "Reboot" button to pop-up "reboot" dialog box, click "OK" to reboot the phone, click "Cancel" will have no operation.

# **Call Service**

#### Register

You can register the sip line on the web or by LCD, the below picture show the web register configuration. The status can show the if the line register successfully. When register success, the upper-left of status bar will show the display name and number

Account1	Account2	Account3	Account4	Account5	Account6		Status
Enable re	egistration					<b>S</b>	Network
Server ac	ldress		Server add	lress			Accounts
Server po	ort cation user		5060 Authentica	tion upor			Call
	cation user	vord		cation p	assw		Display
SIP User			SIP User				Tone
						0	Time&date
		Cancel	Save				

Register

# **Outgoing & Incoming call**

- 1. Pick up the handset, press the headset/handsfree key or click line key to the dial interface;
- 2. Select the line and enter the number ;
- 3. Click #send, it will call out follow the default dial mode you set;
- 4. You can also click video to invite a video call.



When the phone has an incoming call, you can click audio or video to answer the call, if you did not select, the phone will follow the default ans mode you set when you pick up the handset. You can also choose forward or reject the call.

## Video Call

- 1. Pick up the handset, press the headset/handsfree key or click line key to the dial interface;
- 2. Select the line and enter the number;
- 3. Click "Video" to make a call;



<image>

5. Click to adjust the video screen mode and video mode.

# **Blind Transfer**

- 1. During the call, click transfer enter the dial interface;
- 2. Input the number or select contact from contacts or call log;
- 3. Click transfer;

4. If transfer success, the phone will auto end the call, if transfer failed, the phone will return to the conversation interface and hold the call.

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## **Semi-attended Transfer**

- 1. During the conversation, click transfer enter the dial interface;
- 2.Input the number or select contact from contacts or call log, final, click #send;
- 3. When the other end ringing, click transfer to complete the semi attend transfer.

**NOTE:** If you want to use this function, you need to enable the call waiting and call transfer.

## **Attended Transfer**

1.During the conversation, press the transfer key;

2.Input the number you want to transfer to;

3.Tap "#send";

4. Press transfer key after the call answered, and transferred successfully.

#### NOTE:

Call waiting and call transfer must be enabled. The SIP server must support RFC3515.

## **Conference Call**

- 1. Press the CONF soft key during an active call.
- 2. The first call will be placed on hold and dial tone will be heard.
- 3. Dial the number to be added to the conference.
- 4. Press Send.
- 5. When the call is answered, the conference will be started.
- 6. To release the conference, press Split.



#### **Conference Interface**

### **Call Hold**



- 1. Click the "Hold" to put the active call on hold.
- 2. If there is only one call on hold, press the "Resume" to retrieve the call.
- 3. If there is more than one call on hold, click the dialog to change conversation, then click the Resume button to retrieve the call.

# **Android Settings**

# **Ethernet configuration**

Ethernet	
Furn off Ethernet	

Click "Settings", select " Ethernet configuration" switch to the Ethernet configuration interface. You can choose to turn on or turn off the Ethernet, if you turn on the Ethernet, it will auto connect to networking, otherwise, the phone can not connect to networking.

#### Security

Security	
SCREEN SECURITY	
Screen lock PIN	
Automatically lock 5 seconds after sleep	
Owner info	
PASSWORDS	
Make passwords visible	2

- 1. Set the Screen lock
- Click "Settings", enter the android settings interface;
- Click Security->Screen lock enter the configuration interface;
- Follow the prompts to set screen lock which you want to use;
- Set automatically lock time.

- 2. Set owner info, it will display on the lock screen.
- 3. Set if make passwords visible

## Language&Input



#### Language&input interface

- 1. Click "Settings" enter the android settings interface;
- 2. Click language&input enter the above picture interface;
- 3. Select language to set what you want to use;
- 4. Tick the input method which you want to use.

# Web Settings

#### Logon

There are two levels of access: root level and general level. A user with root level access can browse and set all configuration parameters, while a user with general level can set all

configuration parameters except server parameters for SIP and can not browse the

configuration file.

Default user with general level:

Username: guest

Password: guest

Default user with root level:

Username: admin

Password: admin

User:	
Password:	
Language:	English 🗸
	Logon

#### Network

About the network mode settings, please refer to the phone settings.

#### QoS & VLAN

The phone supports 802.1Q/P protocol and DiffServ configuration. Use of a Virtual LAN (VLAN) allows voice and data traffic to be separated.



Chart 1 shows a network switch with no VLAN. Any broadcast frames will be transmitted to all other ports. For example, and frames broadcast from Port 1 will be sent to Ports 2, 3, and 4.



Chart 2 shows an example with two VLANs indicated by red and blue. In this example, frames broadcast from Port 1 will only go to Port 2 since Ports 3 and 4 are in a different VLAN. VLANs can be used to divide a network by restricting the transmission of broadcast frames.

## Note: In practice, VLANs are distinguished by the use of VLAN IDs.

WAN	QoS&VLAN	SERVICE PORT	TIME&DATE	
Link Layer Discove	ry Protocol (LLDP	) Settings		
Enable LLDP 🚺 Enable Learning			Packet Interval(1~3600)	60 second(s)
Quality of Service ( Enable DSCP	(QoS) Settings		SIP DSCP	46 (0~63)
Audio RTP DSC		(0~63)	Video RTP DSCP	46 (0~63)
WAN Port VLAN Se	ttings			
Enable WAN Po	rt VLAN		WAN Port VLAN ID	256 (0~4095)
802.1P Priority	0	(0~7)		
LAN Port VLAN Set	tings			
LAN Port VLAN	Mode Fol	Iow WAN $ \sim$	LAN Port VLAN ID	254 (0~4095)
			Apply	
Service Port				
Service 1 off				_
WAN	QoS&VL	AN SERVIO	E PORT TIME&DATE	
Service Port Se	ettings 😧			
Web Serve	r Type	HTTP	$\checkmark$	
HTTP Serve	er Port	80		
HTTPS Serv	ver Port	443		
Telnet Port		23		
RTP Port Ra	ange Start	10000		
RTP Port Q	uantity	200		
			Apply	

Field Name	Explanation
Web Server Type	Specify Web Server Type - HTTP or HTTPS
HTTP Port	Port for web browser access. Default value is 80. To enhance security, change this from the default. Setting this port to 0 will disable HTTP access. Example: The IP address is 192.168.1.70 and the port value is 8090, the accessing address is http://192.168.1.70:8090.
HTTPS Port	Port for HTTPS access. Before using https, an https authentication certification must be downloaded into the phone. Default value is 443. To enhance security, change this from the default.
RTP Port Range Start	Set the beginning value for RTP Ports. Ports are dynamically allocated.
RTP Port Quantity	Set the maximum quantity of RTP Ports. The default is 200.
	n this page require a reboot to become active. anges to HTTP Port and Telnet ports be values greater than

1024. Values less than 1024 are reserved.

If the HTTP port is set to 0, HTTP service will be disabled.

# Registration

Status     Registered     Domain Realm       Server Address     172.16.1.2     Proxy Server Address       Server Port     5060     Proxy Server Port       Authentication User     2321     Proxy User       Authentication Password     ••••     Proxy Password       SIP User     2321     Backup Proxy Server Address       Display Name     Jasmine     Backup Proxy Server Port	sic Settings >>			2 <u>0</u>
Server Port     5060     Proxy Server Port       Authentication User     2321     Proxy User       Authentication Password     ••••     Proxy Password       SIP User     2321     Backup Proxy Server Address       Display Name     Jasmine     Backup Proxy Server Port	Status	Registered	Domain Realm	
Authentication User     2321     Proxy User       Authentication Password     ••••     Proxy Password       SIP User     2321     Backup Proxy Server Address       Display Name     Jasmine     Backup Proxy Server Port	Server Address	172.16.1.2	Proxy Server Address	
Authentication Password     Proxy Password       SIP User     2321       Display Name     Jasmine   Backup Proxy Server Port	Server Port	5060	Proxy Server Port	
SIP User     2321     Backup Proxy Server Address       Display Name     Jasmine     Backup Proxy Server Port	Authentication User	2321	Proxy User	
Display Name Jasmine Backup Proxy Server Port 5060	Authentication Password	••••	Proxy Password	
	SIP User	2321	Backup Proxy Server Address	
Enable Desistration	Display Name	Jasmine	Backup Proxy Server Port	5060
Enable Registration 🕑 Server Name	Enable Registration		Server Name	
dio Codecs >>	idio Codecs >>			

**Note**: If the proxy configuration is same as the register configuration, in the web interface, the proxy fields are empty. While it is different with register configuration, the config info will display in the web interface. The backup proxy server will be used if the primary server is unavailable.

Set the audio codecs for current sip line.

sic Settings >>				
dio Codecs >>				
Disable codecs			Enable codecs	
G.711U G.722 G.723.1 G.726-32 G.729AB AMR	*	→ ←	G.711A	

## **Dial Plan**

#### **Basic Settings**

1	Press "#" to Send
	Send after 5 second(s)(3~30)
	Press # to Do Blind Transfer
-	Blind Transfer on Onhook
	Attended Transfer on Onhook
	Attended Transfer on Conference Onhook
	Dial Prefix
	Enable E.164

This phone supports 7 dialing modes:

1. Press # to Send - Dial the desired number, and press # to send it to the server.

2. Send after seconds – Number will be sent to the server after the specified time.

3. Press # to Do Blind Transfer - Press # after entering the target number for the transfer.

The phone will transfer the current call to the third party.

4. Blind Transfer on Onhook - Hang up after entering the target number for the transfer. The phone will transfer the current call to the third party.

5. Attended Transfer on Onhook - Hang up after the third party answers. The phone will transfer the current call to the third party.

6. Attended Transfer on Conference Onhook- Hang up during a 3-way conference call, the other two ways will make a call.

7. Dial Prefix- If you set the prefix is 135856, and you want to dial the number

```
13585679801. Then when you dial the number, you can press and hold 0, when the "+" appears, you can just input 79801and send the number, the "+79801" is the numb you input, but you can see the dialing number is 13585679801.
8. Enable E.164- You can refer to the E.164 standard.
```

#### **Dial Plan Add**

This feature allows the user to create rules to make dialing easier. There are several different options for dial rules. The examples below will show how this can be used. Example 1: Substitution -- Assume that it is desired to place a direct IP call to IP address 172.168.2.208. Using this feature, 123 can be substituted for 172.168.2.208.

User-defined Dial Plan Table

Digit Map	Call	Match To Send	Line	Alias Type:Number(length)	Suffix	Media
"123"	Out	No	SIP(172.16.2.208:5060)			Default

Example 2: Substitution -- To dial a long distance call to Beijing requires dialing area code 010 before the local phone number. Using this feature 1 can be substituted for 010. For example, to call 62213123 would only require dialing 162213123 instead of 01062213123.

User	User-defined Dial Plan Table								
	Digit Map	Call	Match To Send	Line	Alias Type:Number(length)	Suffix	Media		
	"1T"	Out	No	Default	rep:010(1)		Default		

Example 3: Addition -- Two examples are shown. In the first case, it is assumed that 0 must be dialed before any 11 digit number beginning with 13. In the second case, it is assumed that 0 must be dialed before any 11 digit number beginning with 135, 136, 137, 138, or 139. Two different special characters are used.

x -- Matches any single digit that is dialed.

[] -- Specifies a range of numbers to be matched. It may be a range, a list of ranges separated by commas, or a list of digits.

Digit Map	Call	Match To Se	end Line	Alias Type:N	lumber(length)	Suffix	Media
"13[2-9]xxxxxxxx"	Out	Yes	Default	: add:0			Defaul
"131xxxxxxx"	Out	Yes	Default	: add:0			Audio
ial Plan Add							
S							
Digit Map							
	itgoing Ca		Match to Send	No 🔻	Media	Default 🔻	
Apply to Call Ou	itgoing Ca efault 🔻		Match to Send Destination	No 🔻	Media Port	Default 🔻	
Apply to Call Ou Line De	-	<u>   </u> ▼		No V		Default 🔻	

Field Name	Explanation						
Phone number	There are two types of matching: Full Matching or Prefix						
	Matching.						
	In Full matching, the entire phone number is entered and						
	then mapped per the Dial Peer rules.						
	In prefix matching, only part of the number is entered						
	followed by T. The mapping with then take place						
	whenever these digits are dialed. Prefix mode supports a						
	maximum of 30 digits.						
Destination	Set Destination address. This is for IP direct.						
Port	Set the Signaling port, the default is 5060.						
Alias	Set the Alias. This is the text to be added, replaced, or						
	deleted. It is optional.						
Note: There are four 1) All: xxx - xxx w	types of aliases. vill replace the phone number.						
2) Add: xxx - xxx	will be dialed before any phone number.						
3) Delete: The charac	ters will be deleted from the phone number.						
4) Replace: xxx - x	xxx will be substituted for the specified characters.						
Suffix	Characters to be added at the end of the phone number.						
	This is optional.						
Delete Length	Sets the number of characters to be deleted. For example,						
	if this is set to 3, the phone will delete the first 3 digits of						
	the phone number. This is optional.						

# Security

WEB FILTER	SECURITY				
Update Security File					
	Select Sec	curity File:		Browse	Update
Delete Security File					
	Select	Security File:	~	✓ Delete	
SIP TLS File					
HTTPS File					

Browse to the security file to be updated. Click the Update button to update.

Note: The sip TLS file and https file both support "xx.pem" format.