Hotel-tech Antique-Style IP Phone
User Manual

Note: You can query the IP address of the phone by pressing *1*1*2 in standby mode.
The default web UI login name=admin, password=admin

project	Hotel-tech Antique-Style IP Phone	
Display language	The webpage supports multiple Languages such as Chinese, English, Russian, French, etc	
Number of lines	1	
Function keys	Flash-off keys, DSS keys, numeric keys	
VoIP protocol	SIP 2.0	
Network protocols	HTTP、BOOTP、FTP、TFTP、IEEE 802.1Q	
Speech coding	G.723.1 (5.3K 6.4K b/S)、G.729 A (8Kb/s)、G.711 A/U (64Kb/S)、G.722(64Kb/S)	
QoS	Speech-First Marking (TOS); Jiffer Buffer; Voice Detection (VAD) and Comfort Background Noise Generation (CNG); Echo Cancellation G.168 (32ms)	
Network interfaces	1 x 100M Ethernet interface (LAN). Static IP addresses and DHCP automatically obtain IP addresses	
Call Handling Function	Call Waiting, Transfer, Hold, Suspension, Redial	
Safe	Password to log in to the web Signaling encryption Voice encryption	
Maintenance	Carrier TR069 support Phone keypad settings The web maintains all the data Telnet maintenance HTTP/TFTP/FTP Auto provision/TR069	
Power supply	External AC Power Adapter (5V DC) PoE IEEE 802.af supported	
Physical Specifications	Operating temperature: 0°C ~ 60°C Relative humidity: 10% ~ 90%	
Product Certification	CE、FCC、RoHS	
Implement functionality	**Here's how**	
Make a call from the receiver	Pick up the receiver	Dial the number first and then press the # button to make a call

Press the intermittent button during a call	1. Press once during the call, the original call is held, enter the transfer state, and the phone listens to the dial tone 2. If the transfer status is not dialed and pressed once, the transfer function will be canceled, the original call will continue to be held, and the phone will listen to the hold tone 3. Press once in the hold state to resume the call on hold	
Implement functionality	**Here's how**	
Answer a call	Once, you hear the phone ringing (the ringtone can be set by yourself);	--Answer after off-hooking
Autoresponder	--1) Set up the automatic response in the phone web; --2) When there is an incoming call, the phone will automatically go off-hook and onto speakerphone.	
Implement functionality	**Here's how**	
Turn on Do Not Disturb	--1) Web > SIP Account; --2) Select Select Line >Do Not Disturb> Turn on	
Cancel Do Not Disturb	--1) Web > SIP Account; --2) Select Select Line >Do Not Disturb> Turn off	
LAN port configuration	Set the network parameters related to the LAN port	
Select an IP type	The default mode is DHCP, and there are three options: DHCP, static IP, and PPPOE	
DNS configuration	There are two modes: automatic acquisition and manual input	
VPN configuration	Set VPN network parameters	
VLAN configuration	Set VLAN network parameters	
Port management	You can set the WEB management port and Telnet management port intercom for maintenance and management, and the factory default is 80 and 23	
QoS	Set QoS-related network parameters	
Network port mirroring	Set parameters related to network port mirroring	
802.1x	Set 802.1x parameters	
LLDP	Set LLDP parameters	
Paging	Set the parameters related to Paging	

Directory

SIP-related	
Enabled*	Check this box to enable this account, otherwise to disable this account
Display name	Set the name of the account to be displayed on the phone screen (if you don't set it, it will be displayed as the account you set)
Account*	The account that matches the SIP server
Authentication username	The authentication username that matches the SIP server
Password*	The password of the account that matches the SIP server
description	A description of the account
SIP Server*	By default, all calls go through the server
Standby server	It is the standby server of the primary server, and the configuration is the same as that of the primary server. When the primary server is unavailable, the standby server is used for calling by default.
Outgoing proxy servers	Both the primary server and the standby server can set up an outgoing proxy server.
Alternate outgoing proxy server	After the outgoing proxy server fails, switch to a standby proxy server
Poll registration interval	Set the polling setup time
NAT Traversal	Specify whether to enable NAT traversal
STUN server configuration	Configure the STUN server on your network
BLA	If you want to enable the BLA feature, the server must support it
Subscribe period	Set up a subscription period
Registration refresh time	Register the refresh interval, the minimum is 20s, the default is 3600s
Autoresponder	Turn on auto-answer for your account
Connection type	The options are UDP, TCP and TLS, and the default is UDP;
Ringtones	Select the ringtone settings of different accounts and set their own ringtones for different accounts
RPport	A parameter used in the SIP protocol suite for NAT traversal registration
Message function code	Go to the voicemail and check the message message
Do not disturb	Do Not Disturb (DND) is enabled to help you block incoming calls
Anonymous calls	to turn on anonymous calling
Anonymous rejection	You can turn on anonymous rejection
Session time	Turn session time on and off
Session time settings	Sets the value of the session time
Refresh the session party	The UAS session is refreshed by the server side; The UAC is refreshed by the client

Calling mode	There are two registration methods: SIP and TEL, and the default is SIP
DNS srv	When the phone is enabled, it can register by resolving the address corresponding to the SRV record of the DNS server domain name, and when it is disabled, it will be a normal address and will not be resolved
Allow-events	Allows the sending of various events in the SIP protocol
Sign up for network traversal	Enable or disable registration for network traversal
Keep-alive Type	Register the heartbeat type
Keep-alive interval	Register for heartbeat intervals
Enable user=phone	If this is enabled, this field will be displayed in the invite data
Meeting mode	It can be implemented by the local mixing of the phone or by the mixing of the server
Meeting URI	Set the specified meeting URI
BLA	If you want to enable the BLA feature, the server must support it
Subscribe period	Set up a subscription period
Signaling encryption	One of the encryption methods
Encryption	RC4 encryption is supported
Encryption keys	Set the encryption key
Voice Encryption (SRTP)	Optional encryption and mandatory encryption
Autoresponder	It is off by default and can be enabled or enabled for groups as needed
Auto-answer mode	Hands-free by default, auto-answer mode can be adjusted as needed
call waiting	The switch is busy when a new call comes in
The call wait tone plays	Set up a tone to alert you to a new call during a call
Automatic redialing	Automatic redial switch
Auto Redial Interval (1~300 seconds)	Automatic redial intervals
Number of auto replays (1~300)	The number of times auto-replay
Hotline function	Hotline function switch
Hotline number	Set up a hotline number
Answering function	Answering function switch
Answering function code	The number that implements the answering function
Keep the tone on track	Whether there is a tone when the person is on hold during a call
DTMF transmission mode	A way to send DTMF by pressing the number key during a call
Hide DTMF	Hide DTMF during a call

100rel reliable retransmission	Support PRACK transmission
Dial lookup	Look for the phone book when dialing and display the matching number
Phone book lookup	Select the phonebook lookup type
Call record keeping	Call record saving switch
Enable BLF transfer during a call	Whether BLF transfer can be enabled during a call
BLF transfer mode	BLF transfer mode: blind transfer or consultation transfer
Rounded corner lights	The default is the call indicator and the general information light, and you can choose to only use it as the call indicator or turn it off
Whether or not to subscribe to voicemail	Whether or not to subscribe to voicemail
Unread voicemail tones	Unread voicemail tones are turned off by default
Missed caller ID	Missed caller ID switch, which is turned on by default
Do Not Disturb soft keys	Do Not Disturb soft key switch, which is turned on by default
Hang up busy tone playback	The idle busy tone playback switch is enabled by default
Transfer function code	Turn on the function code switch, which is disabled by default
Meeting function code	The meeting function code is disabled by default
Keep the feature code	Keep the function code and turn it off by default
The host of the meeting hangs up	When the host of the meeting hangs up, they can end all calls or keep the other two parties on call
Reject the return code	The incoming call refuses to return the message
DND return code	Messages that are returned when you set DND
Strip springs are enabled	The way the strip spring is enabled is turned off by default
Strip spring time (<800ms)	Strip spring time
Called no answer time	Called no answer time
The protagonist has no response time	The protagonist has no response time
Pound Send Method	# OR %23
RFC 2833 Payload	RFC 2833 Payload value settings

Caller ID field	Caller ID field FROM orPAI settings
SIP Session Timer (sec) T1	Timer, when entering the completed state, set a timer H=64*T1 second for all transmissions, and the timer determines when the server-side transaction is resent and answered
SIP Session Timer (sec) T2	T2 is used in conjunction with T1
SIP Session Timer (sec) T4	T4 represents the possible time for the network to transmit information in server-side and client-side transactions, and the default T4 = 5 seconds
Register the carry Port	Register the carry port switch
Headset working mode	This model is not supported
Ringing in agent mode	This model is not supported
Whether the BLF is idle or not	This model is not supported
serial number	The sequence number of the new dial plan
description	A description of the new rules
IP	IP address of the peer device
port	The default value is 5060, which can be changed according to the actual situation
Dial plan	Rules for issuing dial-ups
Eat code/insert code	According to the actual needs, the position and number of digits of the code/insertion code can be set
Select the country	The choice of dial tone allows the country dial tone to be adjusted according to the standards used by the country
Volume settings	The volume can be set for each mode
Speech codecs	Supports five main encoding types, including G722, G711A, G711U, G729A, G723 Set voice function codecs in the list of options, in order of precedence from top to bottom. --Adjust the priority of voice encoding: Click to select the voice code in the "Encoded Enabled" field on the left, and then click 'Up' or 'Down' to adjust it. --Disable voice coding: Click to select the voice code in the "Enabled Encoding" field on the left, select an enabled voice code, and then click on it  , the selected voice code moves to the "No Encoding Enabled" bar on the right. --Enable voice coding: Click to select the voice code in the "No encoding enabled" bar on the right, select a disabled voice code, and then click on it  , the selected voice code moves to the Enabled encoding bar on the left.
other	VAD, Echo Suppression Mode Options
Jitter Buffer	It is a data sharing area where voice packets are collected, stored and sent evenly, with a settable delay time
other	Set parameters such as RTP packet size, VAD, and sidetone
Server Address	This parameter can be written as the LDAP server IP address or DNS

	domain name. For example: 192.168.1.100 Ldap.company.com
Port	This parameter is the port number used by the LDAPserver. Default port: 389
Base	This setting specifies the root node of the query database set on the LDAP server, and specifies the root node of the specific query database to help reduce the query time.
user name	This parameter is used by the endpoint to authenticate the user to the LDAP server.
password	This parameter is used by the endpoint to authenticate the user identity password used to connect to the LDAP server.
Max.Hits(1~32000)	This parameter is used to set the number of query results returned by the LDAP server, write 0 or blank will return all results. Default value: 50
LDAP Name Attributes	This parameter is set in the hope that the result will be output with "Name" as the condition. Default value: Null
LDAP Number Attributes	This parameter is set to the desired output of the result conditional on "Number Attributes". Default value: Null
Protocol	There is version 2 and version 3. Default value: 3
Search Delay(ms)(0~2000)	This parameter is used to set the interval between each query to the server. Default value: 0
LDAP Lookup for Incoming Call	If this option is enabled, the endpoint will initiate a query request to the LDAP server every time a call is made, and the name of the queried contact will be displayed on the endpoint. Default value: Not enabled
LDAP Lookup for PreDial/Dial	If this option is enabled, the endpoint will send a query request to the LDAP server every time a call is made, and the name of the queried contact will be displayed on the endpoint. Default value: Not enabled
Blacklist	
serial number	The sequence number of the blacklist
description	A description of the blacklist
surname	The last name of the contact who was added to the blacklist
name	The name of the contact who was added to the blacklist
Mobile phone number	The mobile phone number of the contact that was added to the blacklist
Family number	The home number of the contact who was added to the blacklist
Office number	The office number of the contact who was added to the blacklist
Account	Accounts that apply this blacklist will default to automatic mode
Server IP	The IP address of the FTP server
filename	Enter the name of the upgrade file correctly
PNP activation	Request the server to get the URL that supports the update
PNP Update Cycle (min)	PNP update cycle
Automatic upgrade enabled	This is enabled by default, if you select "Off", the phone will not be automatically upgraded, and the following options will not take effect.
agreement	At present, four protocols are supported, namely TFTP, HTTP, HTTPS and FTP.

Firmware server address	Download the server address of the firmware, configuration, etc., and use it with the protocol" option, and fill in the address of the server with which protocol you choose.
Username and password	Some firmware servers require a username and password to download, please fill in both fields.
Firmware files are automatically downloaded	This is checked by default. How do I know the software version on my server? Before downloading the software, you need to download a file that records the version of the server software, which is a .cfg file, so we need to configure these in advance. To elaborate, the cfg file for U1/U1S is in the form of FD000U1.cfg. Write the name of the firmware to be upgraded in this cfg file, and then find the firmware to be upgraded through the file name in this file. Keep in mind that as long as the server is turned on and the automatic upgrade function is not turned off, the phone will go to the server to find it, and if there is a firmware in the server that is different from the current version of the phone, then the phone will download the upgrade.
Automatically download configuration files	This is checked by default. The file name of this configuration file must be the same as the MAC of the phone from which this file is downloaded, and the file name is MAC.xml.
Automatic download of the extension station	This is checked by default. The file name of the extension station must be in the form of account _ Extern.xml", otherwise it will not be downloaded.
Automatically download your corporate address book	This is checked by default. The file name is uniformly Enterprise_Phonebook.xml, otherwise it will not be downloaded.
Automatically download your personal phone book	This is checked by default. The file name is in the form of account number _ Phonebook.xml", otherwise it will not be downloaded.
Boot detection	This is checked by default. If this option is checked, the detection will be upgraded immediately after the phone is turned up, and if it cannot be detected, the detection will be repeated up to 2 times in a row. If you can't detect it twice, wait for the next testing cycle.
One-click update enabled	Enable/disable the configuration on the download server after the phone is powered on
Wait time	One-click update of the waiting time
Disable the phone during power-on detection	It is off by default. If this option is enabled, the phone will be detected immediately after the power is turned off, and the phone cannot do any operation at this time. If it is detected, it will be upgraded, and if it is not detected, it will be returned to the standby interface.
Automatic upgrade frequency	The default is 7 days, and you can manually set how long it will take to download. Note: The download period is calculated from the time of the last test, for example, if the boot check is set, the time is calculated from the time of booting; If the "Auto Provision" button is clicked, the time is calculated from the time of the click; If boot check is turned off, the time is calculated from the last time the server was checked, regardless of

	whether it was restarted or not.
Automatic upgrade time	By default, there is no time set. You can set the time for automatic upgrades.
The time of the next automatic upgrade	
AES enabled	The AES encryption algorithm is used to decrypt the downloaded file, and the key is configured on the WEB. This is not done yet
AES key	AES key
The name of the download file	The default is lowercase, or you can choose to uppercase if you want
Upgrade now	You can click Upgrade Now, and the phone will be detected and downloaded immediately, and then upgraded.

The HS Retro01P contains a digital dial with a programmable speed dial in the middle, as well as a joystick and a fork spring key

Key name		Corresponding symbol (this model does not have a screen to ignore this column)
Numeric keys	1	1
	2	2
	3	3
	4	4
	5	5
	6	6
	7	7
	8	8
	9	9
	*	*
	0	0
	#	#
Function keys	navigation	^
	Navigation	~