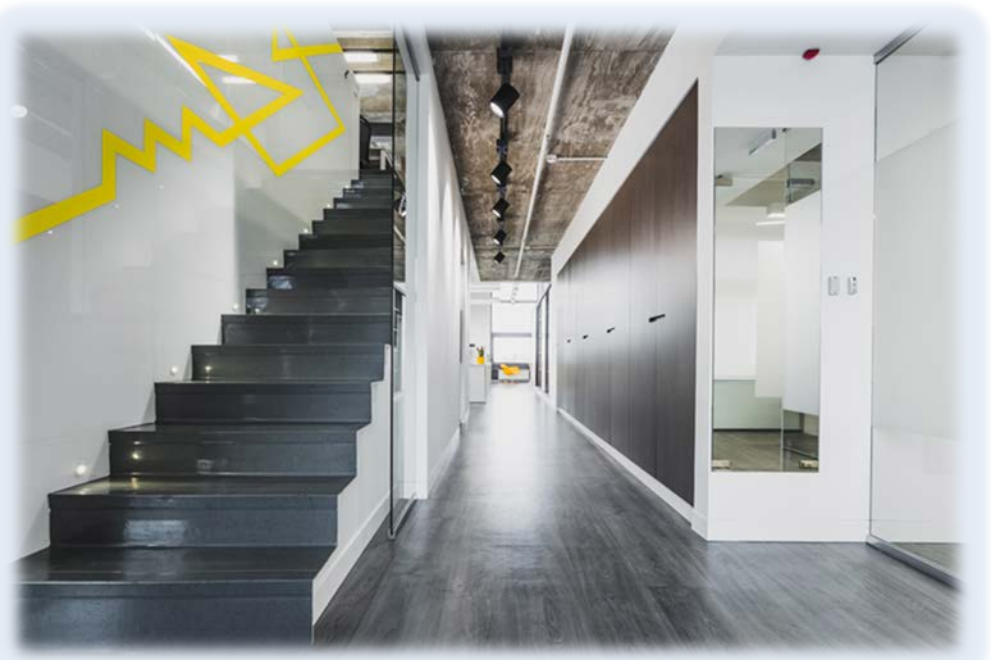


**HA9888TSD-IP(F623C)**  
**Standard Business IP Phone**



Intelligent Communication Device for Next Generation Network



# HA9888TSD-IP(F623C)

## Standard Business IP Phone

### A Cost-effective very popular business IP phone.

F623C is a cost-effective IP telephony, 2 SIP accounts, 128\*64 LCD, 10 programmable soft keys, RJ9 headset. 10 speed dial (Without BLF) buttons provide convenient communication experience for the busy call corporate office and management personnel. F623C excellent HD voice quality, rich and leading edge telephony system functions to provide users with personalized information customization, support automatic deployment, advanced security protection mechanism, compatible with a wide range of mainstream industry SIP/IMS platform.

F623C is standard office IP phones. It has two lines, four dynamic soft keys. high-quality speaker, 10 multi-function keys, and RJ9 headset port. Each line can be configured independently of the telephone number, or composite shared telephone number.

F623C support the latest configuration of various methods, such as: TR-069, HTTPS, HTTP and TFTP. Users can fully experience the high-definition voice, eight function keys, and intuitive menu options to bring convenient, fast.



### Highlights:

- Support SIP V2.0 (RFC 3261, 3262, 3263, 3264)
- 2\*10/100M RJ45 Port
- 128\*64 TFT
- 2 Lines supported
- 4 customizable buttons for LCD menu selection
- 10 multi-function keys, (Without BLF)
- Support the headset
- Auto Configuration- TFTP/HTTP/HTTPS/TR-069/SNMP

### Functions:

#### Codec and Voice Features

The F623C supports 5 codec: G.711 (A-Law, U-Law), G.729A/AB, G.723 with 5.3 kbps and 6.3 kbps, iLBC and G.722 (HD). The F623C supports VAD (Voice Active Detection), CNG (Comfort Noise Generation), AEC, ANC (Automatic Noisy Cancel), AGC (Automatic Gain Control) and adaptive Jitter buffer.

#### Call Features

2 lines, Call Waiting, Auto Answer, Music on hold, Caller ID and call waiting ID, 3-way Conference, Call Hold, Call Forwarding, Call Mute, Delayed Hotline, Redial, Call Transfer: blind transfer and attended transfer, Dial Plan, Phonebook, Black List, MWI, SMS, DND, Full-duplex Speakerphone Call log: redial list, answered calls and missed calls Volume Adjustment: Handset/Headset, Speaker and Ringer DTMF Relay: In-band, Out-band (RFC2833) and SIP INFO

#### Management

Menu Configuration, SNTP Time Synchronization, Daylight Saving Time, Alarm Clock, Password Reset, Web access management, Local and Remote Syslog (RFC3164), Factory Default, Firmware Upgradeable, Web Management Interface, Multi User Level, SNMPv2, TR069, Auto Provisioning: TFTP (including option 66), HTTP and HTTPS

#### Applications

MAC Address Cloning, VPN: PPTP and T2TP, DMZ, Direct IP to IP calling, IP conflict detection, Built-in NAT Router, DHCP Server and Client, Layer 2 QoS: 802.1Q/VLAN ID and 802.1p PRI Layer 3 QoS: SIP QoS, RTP QoS and Data QoS SIP proxy redundancy: dynamic via DNS SRV, A records NAT Traversal: Static NAT Route and Traversal by STUN

#### Web Configuration

The F623C supports web configuration with MD5 authentication, and two level configurations which are admin mode and user mode. In admin mode user can configure all parameters, while in user mode user can only configure partly parameters with partly can't configuration such as VoIP settings.

#### Cloud Platform Management

The F623C supports cloud platform management. By the device unique SN code, centralized Management will make the deployment, updating and remote management much more easier and fast, based on the standard TR069 protocol.



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### Specifications

CHARACTERISTICS	
VoIP SoC	<ul style="list-style-type: none"> <li>CPU Core: MIPS 580Mhz</li> </ul>
Flash	<ul style="list-style-type: none"> <li>16M Flash</li> </ul>
RAM	<ul style="list-style-type: none"> <li>64M RAM</li> </ul>
Display	<ul style="list-style-type: none"> <li>128×64 Graphic LCD without white back light</li> </ul>
Phone Keys	<ul style="list-style-type: none"> <li>2 Line keys and 4 Soft keys, MENU, UP,DOWN,MUTE/DEL, REDIAL •HOLD, MSG, Headset, Handsfree</li> <li>12 Dialing Buttons (0~9, *, #)</li> <li>10 Multi-functional key without BLF</li> </ul>
LED	<ul style="list-style-type: none"> <li>1 MSG LED</li> </ul>
Power Supply	<ul style="list-style-type: none"> <li>AC/DC Adapter</li> <li>AC Input: 100~240V, 50~60Hz</li> <li>DC Output: 5V, 1A</li> </ul>
I/O Interfaces	<ul style="list-style-type: none"> <li>2 RJ-45 port (10/100 Ethernet Ports)</li> <li>1 RJ9 for headset</li> <li>Headset jack and Wall-mount</li> </ul>
Environmental	<ul style="list-style-type: none"> <li>Operation Temperature: 0~50 Degree C</li> <li>Storage Temperature: -25~ 85 Degree C</li> <li>Relative Humidity: 10%~90% No Condensing</li> <li>Shock: Up to 75cm (30 inches) `Drop upon Package</li> </ul>
Audio Feature	<ul style="list-style-type: none"> <li>G.711 (A-Law, μ-Law)</li> <li>G.722</li> <li>G.723.1</li> <li>G.729AB</li> <li>G.726</li> <li>iLBC</li> <li>Adaptive Jitter Buffer Management</li> <li>Voice Activity Detection</li> <li>Comfort Noise Generation</li> <li>Echo Cancellation</li> </ul>
Protocols	<ul style="list-style-type: none"> <li>SIP V2 (RFC 3261,3262,3263,3264)</li> <li>SDP (RFC2327)</li> <li>RTP/RTCP (RFC1889 and RFC1890)</li> <li>NAPTR for SIP URI Lookup (RFC2915)</li> <li>STUN (RFC 3489)</li> <li>ARP/RARP (RFC 826/903)</li> <li>SNTP (RFC 2030)</li> <li>DHCP</li> <li>DNS/DNS SRV (RFC1706 and RFC 2782)</li> <li>IEEE802.1Q VLAN/802.1p and IP TOS</li> <li>802.11n 1T1R (Only for W series)</li> <li>SNMPv2</li> <li>TR069</li> </ul>
Management	<ul style="list-style-type: none"> <li>Firmware Upgradeable</li> <li>Web Management Interface</li> <li>Password Management</li> <li>Local and Remote Syslog (RFC3164)</li> <li>Auto Provisioning: TFTP, HTTP and HTTPS</li> <li>SNTP Time Synchronization</li> <li>Multi User Level</li> </ul>

Applications	<ul style="list-style-type: none"> <li>MAC address cloning</li> <li>SIP proxy redundancy: dynamic via DNS •• SRV,</li> <li>Direct IP to IP calling</li> <li>NAT Traversal: Traversal by STUN</li> <li>Built-in NAT Router</li> <li>QoS with Layer 2 and Layer 3</li> <li>DHCP Server and Client</li> <li>IP conflict detection</li> <li>Support PoE comply with • IEEE802.3af(optional)</li> </ul>
Call Features	<ul style="list-style-type: none"> <li>Call Waiting</li> <li>Auto Answer</li> <li>Music on hold</li> <li>Caller ID and call waiting ID</li> <li>3-way Conference</li> <li>Call Hold</li> <li>Call Forwarding</li> <li>Call Mute</li> <li>Call Transfer: blind transfer and attended •transfer</li> <li>Delayed Hotline</li> <li>SMS Functions</li> <li>Redial</li> <li>Speed Dial</li> <li>Pick up</li> <li>Call park</li> <li>Dial Plan</li> <li>Phonebook</li> <li>Black List</li> <li>MWI</li> <li>Call log: redial list, answered calls and missed •calls</li> <li>DND</li> <li>Full-duplex Speakerphone</li> <li>Volume Adjustment: Handset/Headset, •Speaker and Ringer</li> <li>DTMF Relay: In-band, Out-band(RFC2833) •and SIP INFO</li> </ul>
Phone feature	<ul style="list-style-type: none"> <li>Customized Ring Tone</li> <li>SMS (100 records)</li> <li>Call History (100 records )               <ul style="list-style-type: none"> <li>Most Recently Missed Calls</li> <li>Most Recently Received Calls</li> <li>Most Recently Dialed Numbers</li> </ul> </li> <li>Phone book ( 100 records)</li> <li>Speed Dial (10 records )</li> </ul>
Dimension	<ul style="list-style-type: none"> <li>(L) 191 x (W) 205 x (H) 75 mm</li> <li>720g (without package)</li> </ul>



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