Portable IP Phone FIP16

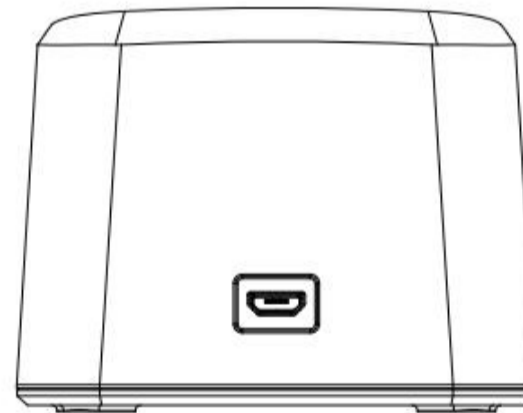## Feature Keys

- Battery lift: (4000mA Battery)

    Standby: 24 Hours

    Talk: 6 Hours

- 1.8 TFT Color Screen

- 2.4G/5G Wireless

- 1 USB 2.0 Port

- 1 Base

USB Provide Power

Overview

FIP16 is a new generation of color IP phone, providing convenient communications experience to call a busy corporate office and management personnel. The built-in internal antenna make user connect the network much easier and more quickly, user can not only connect FIP16 to the network by using a ethernet cable, but also he/ she can connect FIP16 to the wireless network, the performance is the same as mobile phone. User can connect the wireless network on FIP11WP's LCD or on phone's webpage. FIP16 excellent HD 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 / IM S platform.

Technical Parameters

VoIP SoC	• CPU: MT7628(580Mhz)
Flash	• 16MB Flash
RAM	• 64MB DDR2 RAM
Power	• AC/DC Adapter 　　AC Input: 100~240V, 50~60Hz 　　DC Output: 5V, 1A
Operating System	• Linux 2.6.36
Battery	• 4000mA Battery
Environmental	• Operation Temperature: 0~50 Degree C • Storage Temperature: -25~ 85 Degree C • Relative Humidity: 10%~90% No Condensing
Audio Feature	• G.711(A/u) • G.722(HD) • G.729AB • G.723 • G.726 • iLBC • Adaptive Jitter Buffer • Voice Activity Detection • Comfort Noise Generation • Echo Cancellation
Phone Keys	• 6 Function Keys • 12 Dialing Buttons (0~9, *, #) • 2 Soft Keys
Applications	• SMS Function • MAC Address Cloning • SIP proxy redundancy: dynamic via DNS SRV, A records • Direct IP to IP calling • DNS SRV • NAT Traversal: Traversal by STUN • Built-in 100M NAT Router • VPN: PPTP,l2tp,OpenVPN • QoS with Layer 2 and Layer 3 • DHCP Server and Client • IP conflict detection • Support Mutil-Vlan • Support 2.4GHZ
Phone Features	• Customized Ring Tone • SMS (100 Records) • Call History (100 Records) 　　Most Recently Missed Calls 　　Most Recently Received Calls 　　Most Recently Dialed Numbers • Phone Book (500 Records)

Management	• Firmware Upgradeable
	• Web Management Interface
	• Password Management
	• Local and Remote Syslog (RFC 3164)
	• Auto Provisioning: TFTP, HTTP And HTTPS
	• SNTP Time Synchronization
	• Multi User Level
	• Support TR069 Management Mode
	• Support Flying FCMS Cloud Management
Call Features	• DTMF Relay: In-band, RFC 2833 and SIP info
	• Volume Adjustment: Handset, Handfree and Ring
	• Call Hold, Call Transfer And Call Waiting
	• CID And CWCID
	• Music ON Hold
	• MWI
	• 3 Way Conference
	• Auto Answer
	• Call Pickup
	• Full-duplex Speakerphone
	• Dial Rule
	• Hotline
	• Redial
	• Mute
	• Call Forward
Protocol	• SIP V2 (RFC 3261, 3262, 3263, 3264)
	• SDP (RFC2327)
	• RTP/RTCP (RFC1889 and RFC1890)
	• NAPTR for SIP URI Lookup (RFC2915)
	• STUN (RFC 3489)
	• ARP/RARP (RFC 826/903)
	• SNTP (RFC 2030)
	• IEEE802.11 b/g/n
	• IEEE802.11 ac
	• HTTP Server for Web Management
	• TFTP/HTTP/HTTPS for Auto Provisioning
	• Message Waiting Indicator (RFC3842)
	• DHCP Option Codes for SIP (RFC3361)
	• DNS/DNS SRV (RFC1706 and RFC 2782)
	• IEEE802.1Q VLAN/802.1p and IP TOS
	• SNMPV2
	• TR069 (TR104/TR098/TR111)