

# Flying Voice 2011 products catalogue



Flying Voice Technology Limited

# PRODUCT CATALOG

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## IP ATA Series

Features/Model	GIX101	G501	G501C	G502	G102
Network Ports	2* RJ45 10/100Mbps (WAN/LAN)	2* RJ45 10/100Mbps (WAN/LAN)	2* RJ45 10/100Mbps (WAN/LAN)	2* RJ45 10/100Mbps (WAN/LAN)	1* RJ45 10/100Mbps (WAN)
FXS Ports	1	1	1	2	2
SIP Accounts	1	1	1	2	2
Lifeline Ports (PSTN)	Yes	No	Yes	No	No
Wire-Speed NAT	No	Yes	Yes	No	No
DHCP	Client/Server	Client/Server	Client/Server	Client/Server	Client
Voice Codec	G.711(A-law, U-law),G.729A/B				
Management	Integrated IVR, Web browser, Auto-Provision with TFTP/HTTP/HTTPS, TR069, SNMP				
Fax	T.30, T.38 Fax				

- SIP V2.0 ( RFC 3261, 3262, 3263, 3264)
- 2 10/100M RJ45 ports (1 WAN, 1LAN), 1 FXS, 1 PSTN Lifeline
- Real time FAX over IP via T.38 and T.30 FAX with G.711
- PAMS is above 4.0 with G.729 Codec
- Ultra-compact and lightweight
- VPN Pass through (PPTP, L2TP and IPSec)
- Field proven compatibility and stability



Size: (L) 110 x (W) 75 x (H) 28 mm

## PLACE ORDER

MODEL	DESC.
GIX101	2 10/100M RJ 45 ports, 1 FXS, 1 PSTN Lifeline

**VoIP SoC** • CPU Core: ARM946E@150MHZ

**Flash** • 4MBytes Flash

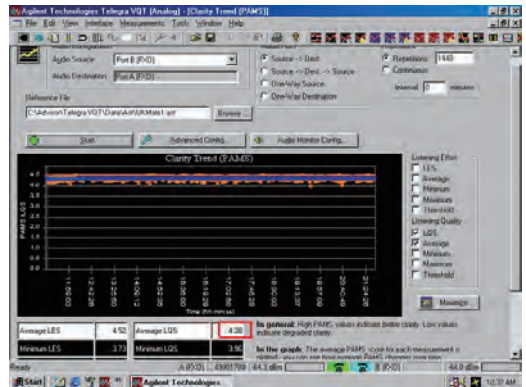
**RAM** • 8MBytes SDRAM

**Power Supply** • AC/DC Adapter  
• AC Input: 100~240V, 50~60Hz  
• DC Output: 5V, 2A

**I/O Interfaces** • 2 RJ-45 for 10/100 Ethernet Ports that can be configured as ether switched or routed ports  
• 1 RJ-11 for FXS Port  
• 1 RJ11 for PSTN Port (PSTN life line)

**Environmental** • Operation Temperature: 5~45 Degree C  
• Storage Temperature: -25~ 85 Degree C  
• Relative Humidity: 10%~90% No Condensing  
• Shock: Up to 75cm ( 30 inches) Drop upon Package

**Audio Codec** • G.711 (A-Law,  $\mu$ -Law) with PAMS above 4.3  
• G.723.1  
• G.729A/AB with PAMS above 4.0  
• Adaptive Jitter Buffer Management  
• Voice Activity Detection  
• Comfort Noise Generation  
• Echo Cancellation



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- Protocols**
- SIP V2 (RFC 3261,3262,3263,3264)
  - Backward Compatible with RFC2543
  - Session Timer (RFC4028)
  - SDP (RFC2327)
  - RTP/RTCP (RFC1889 and RFC1890)
  - NAPTR for SIP URI Lookup (RFC2915)
  - STUN (RFC 3489)
  - ARP/RARP (RFC 826/903)
  - SNTP (RFC 2030)
  - DHCP/PPPoE
  - PPTP/LzTP VPN
  - HTTP Server for Web Management
  - TFTP/HTTP for Auto Provisioning
  - Message Waiting Indicator (RFC3842)
  - DHCP Option Codes for SIP (RFC3361)
  - DNS/DNS SRV (RFC1706 and RFC 2782)

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- Applications**
- MAC Address Cloning
  - Support VPN to prevent service blocking issue
  - SIP proxy redundancy
    - dynamic via DNS SRV, A records
  - Direct IP to IP calling
  - Support PSTN backup
  - NAT Traversal
    - Traversal by STUN
  - Support lifeline function
  - Built-in NAT Router
  - QoS with Layer 2 and Layer 3
  - DHCP Server and Client
  - IP conflict detection
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- Management**
- Firmware Upgradeable
  - Web Management Interface
  - IVR-driven Management Interface
  - Local and Remote Syslog(RFC3164)
  - Auto Provisioning
  - SNTP Time Synchronization
  - Multi User Level
  - SNMPv2
  - TR069
- 

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- Call Features**
- 3-way Conference
  - Music on hold
  - Call Hold
  - Call Forwarding
  - Call Mute
  - Call Transfer
  - Call Waiting
  - Caller ID and CWCID
  - DTMF Relay
    - In-band
    - Out-band(RFC2833)
    - SIP Info
  - Hotline
  - Real time FAX over IP via T.38
  - T.30 FAX with G.711
  - Dial Plan
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**Dimension** • (L) 110 x (W) 75 x (H) 28 mm

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**Unit Weight** • 110 g ( without package)

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- SIP V2.0 ( RFC 3261, 3262, 3263, 3264 )
- 2 10/100M RJ45 (1 WAN, 1LAN) , 1 FXS
- 1 PSTN Pass through port (lifeline)- for G501C only
- Built-in Hardware Wired Speed NAT Router with 100Mbps wire-speed NAT
- Real time FAX over IP via T.38 and T.30 FAX with G.711
- PAMS is above 4.0 with G.729 Codec
- Field proven compatibility and stability
- Auto-Provisioning TFTP/HTTP/HTTps,TR-069, SNMP



Size: (L) 120 x (W) 80 x (H) 28 mm

### PLACE ORDER

MODEL	DESC.
G501	2 10/100M RJ45 ports, 1 FXS
G501C	2 10/100M RJ 45 ports, 1 FXS, 1 PSTN lifeline

**VoIP SoC** • 32-bit RISC CPU + Hardware NAT Engine

**Flash** • 4MBytes Flash

**RAM** • 16MBytes SDRAM

**Power** • AC/DC Adapter

**Supply** • AC Input: 100~240V, 50~60Hz  
• DC Output: 5V, 2A

**I/O Interfaces** • 1 RJ-45 (Ethernet 10/100 Base-T)  
• 1 RJ11 for FXS Port  
• 1 RJ11 for PSTN Port (PSTN life line, G501C only)

**Key Feature** • Wire-speed NAT/NAPT route function

**Environmental** • Operation Temperature: 0~50 Degree C  
• Storage Temperature: -25~ 85 Degree C  
• Relative Humidity: 10%~90% No Condensing  
• Shock: Up to 75cm ( 30 inches)

**Audio Codec** • G.711 (A-Law, u-Law) with PAMS above 4.3  
• G.729A/AB with PAMS above 4.0  
• Adaptive Jitter Buffer Management  
• Voice Activity Detection  
• Comfort Noise Generation  
• Echo Cancellation  
• T.30 FAX with G.711  
• Real time FAX over IP via T.38

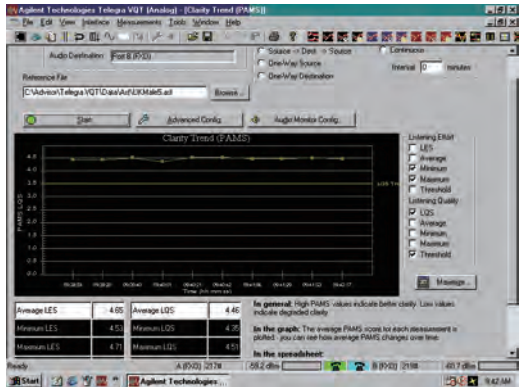
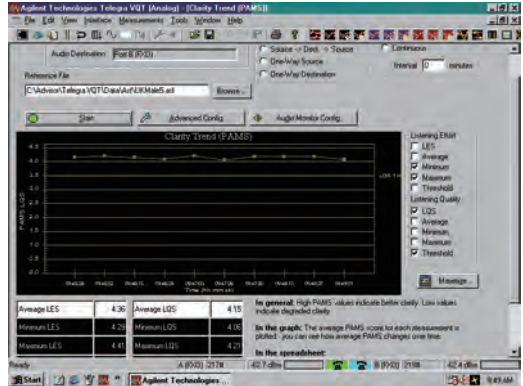
**Protocols** • SIP V2 (RFC 3261, 3262, 3263, 3264)  
• Backward compatible with RFC2543  
• Session timer (RFC4028)  
• SDP (RFC2327)  
• RTP/RTCP (RFC823/903)  
• NAPTR for SIP URI lookup (RFC2915)  
• STUN (RFC2030)  
• ARP/RARP (RFC 826/903)  
• SNTP (RFC 2030)  
• DHCP/PPPoE  
• HTTP Server for Web Management  
• TFTP/HTTP 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 DSCP

- Management**
- Firmware Upgradeable
  - Web Management Interface
  - IVR-driven Management Interface
  - Local and Remote Syslog (RFC3164)
  - Auto Provisioning
  - SNTP Time Synchronization
  - Multi User Level
  - SNMP v2
  - TR069

- Applications**
- NAT/NAPT router with wire-speed
  - MAC address cloning
  - SIP proxy redundancy:
    - dynamic via DNS SRV
    - A records
  - Direct IP to IP calling
  - NAT Traversal by STUN
  - Support DMZ and Super DMZ
  - Built-in Wired speed NAT Router
  - DHCP Server and Client
  - IP conflict detection
  - VPN Passthrough (PPTP, L2TP and IPsec)
  - Virtual Server for Port Mapping
  - Speed Limit
  - QoS with Layer 2 and Layer 3
  - ARP spoofing prevention

- Call Features**
- 3-way conference
  - Music on hold
  - Call hold
  - Call forwarding
  - Call mute
  - Call Transfer
  - Call waiting
  - Caller ID and CWCID
  - DTMF Relay:
    - In-band
    - Out-band
    - SIP info
  - Hotline
  - Short Dial
  - Call Block
  - Dial Plan

**Dimension** • (L) 120 x (W) 80 x (H) 28 mm  
**Unit Weight** • 140 g ( without package)



# IP ATA 2 FXS Analog Telephone Adapter

## G502/G102

- SIP V2.0 ( RFC 3261, 3262, 3263, 3264 )
- 2 10/100M RJ45 port (1 WAN, 1LAN), 2 FXS- for G502
- 1 10/100M RJ45 port (1 WAN), 2FXS for G102
- Real time FAX over IP via T.38 and T.30FAX with G.711
- PAMS is above 4.0 with G.729 Codec
- Field proven compatibility and stability
- Auto-Provisioning-TFTP/HTTP/HTTps,TR-069,SNMP



Size: (L) 120 x (W) 80 x (H) 28 mm

### PLACE ORDER

MODEL	DESC.
G502	2 10/100M RJ45 ports, 2 FXS
G102	1 10/100M RJ 45 ports, 2 FXS

**VoIP SoC** • CPU Core: MIPS24KEc@360MHZ

**Flash** • 8MBytes Flash

**RAM** • 16MBytes SDRAM

**Power** • AC/DC Adapter

**Supply** • AC Input: 100~240V, 50~60Hz  
• DC Output: 5V, 2A

**I/O Interfaces** • 2 RJ-45 for 10/100 Ethernet Ports  
• 2 RJ-11 for FXS Port

**Environmental** • Operation Temperature: 5~45 Degree C  
• Storage Temperature: -25~ 85 Degree C  
• Relative Humidity: 10%~90% No Condensing  
• Shock: Up to 75cm ( 30 inches) Drop upon Package

**Audio Codec** • G.711 (A-Law,  $\mu$ -Law) with PAMS above 4.3  
• G.729A/AB with PAMS above 4.0  
• Adaptive Jitter Buffer Management  
• Voice Activity Detection  
• Comfort Noise Generation  
• Echo Cancellation  
• T.30 FAX with G.711  
• Real time FAX over IP via T.38(Phone 2 port only)

**Protocols** • SIP V2 (RFC 3261,3262,3263,3264)

• Backward Compatible with RFC2543

• Session Timer (RFC4028)

• SDP (RFC2327)

• RTP/RTCP (RFC1889 and RFC1890)

• NAPTR for SIP URI Lookup (RFC2915)

• STUN (RFC 3489)

• ARP/RARP (RFC 826/903)

• SNTP (RFC 2030)

• DHCP/PPPoE

• PPTP/L2TP VPN

• TFTP Server for Web Management

• TFTP/HTTP/HTTps for Auto Provisioning

• DNS/DNS SRV (RFC1706 and RFC 2782)

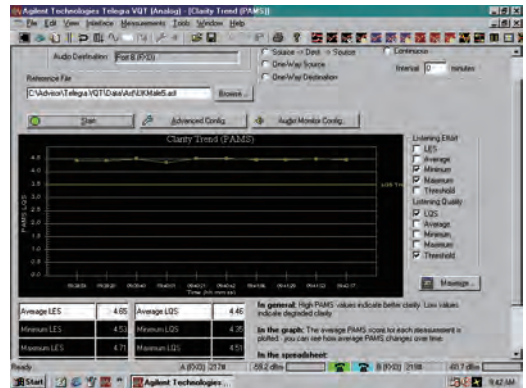
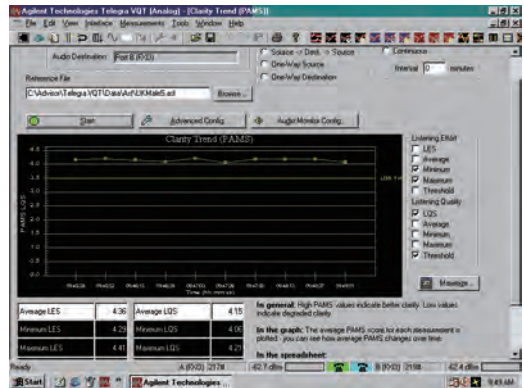
- Management**
- Firmware Upgradeable
  - Web Management Interface
  - IVR-driven Management Interface
  - Local and Remote Syslog (RFC3164)
  - Auto Provisioning
  - SNTP Time Synchronization
  - Multi User Level
  - SNMPv2
  - TR069

- Applications**
- NAT/NAPT Router function
  - MAC Address Cloning
  - DHCP Server
  - PPTP/L2TP VPN
  - PPPoE
  - SIP proxy redundancy
    - dynamic via DNS SRV, A records
  - NAT Traversal by STUN
  - DMZ
  - QoS with Layer 3
  - DHCP Client and DHCP Server
  - IP conflict detection

- Call Features**
- 3-way Conference
  - Music on hold
  - DTMF Relay: In-band, Out-band and SIP INFO
  - Call Hold
  - Call Forwarding
  - Call Mute
  - Call Transfer
  - Call Waiting
  - Speed Dial
  - Caller ID and CWCID
  - Hotline
  - Real time fax over IP via T.38
  - T.30 FAX with G.711
  - Dial Plan
  - Black List
  - Call Log

**Dimension** • (L) 120 x (W) 80 x (H) 28 mm

**Unit Weight** • 140 g ( without package)





## Wireless Router+ VoIP Series

Features/Model	G800	G801	G900
Network Ports	5* RJ45 10/100Mbps (1WAN/4LAN)	5* RJ45 10/100Mbps (1WAN/4LAN)	5* RJ45 1000Mbps (1WAN/4LAN)
FXS Ports	0	1	0
USB Port	No	No	Yes
Ethernet Ports	100M	100M	1000M(Giga)
Wire-Speed NAT	Yes (100M)	Yes(100M)	Yes(1000M)
Wireless	2.4G 2T2R (300Mbps)	2.4G 2T2R (300Mbps)	2.4G/5G Dual-band (300Mbps + 300Mbps)
Voice Codec	N/A	G.711(A-law, U-law),G.729A/B	N/A
Management	Integrated IVR, Web browser, Auto-Provision with TFTP/HTTP/HTTPS, TR069, SNMP		

# AP 802.11n 300M AP Router with HNAT

G800

- 1 100M port (WAN) , 4 100M RJ45(4LAN)
- IEEE802.11n 2T2R 300Mbps wireless
- 5dBi external antenna
- Hardware NAT(HNAT) engine with 100Mbps NAT/NAPT speed
- Wireless throughput is above 80Mbps in average
- Rich QoS capability in wired and wireless
- Auto-Provisioning-TFTP/HTTP/HTTps,TR-069,SNMP



Size: (L) 180 x (W) 110 x (H) 30 mm

## PLACE ORDER

MODEL	DESC.
G800	1 100M, 4 10/100M RJ45 ports, 2T2R

- VoIP SoC** • CPU Core: MIPS24KEc(400MHz)
- Flash** • 8MBytes Flash
- RAM** • 32Mbytes DDR2
- Power Supply** • AC/DC Adapter
- AC Input: 100~240V, 50~60Hz
- DC Output: 5V, 2A

- I/O Interfaces** • 1 RJ-45 for WAN port (Ethernet 10/100 Base-T)
- 4 RJ-45 for LAN port (Ethernet 10/100 Base-T)
- Two 5dB Antenna for IEEE802.11n 2T2R Wireless

- Environmental** • Operation Temperature: 0~45 Degree C
- Storage Temperature: -25~ 85 Degree C
- Relative Humidity: 10%~90% No Condensing
- Shock: Up to 75cm ( ` 30 inches)

- Key Feature** • Hardware NAT(HNAT) engine with 100Mbps NAT/NAPT speed
- Wireless throughput is above 80Mbps in average

- Protocols** • ARP/RARP (RFC 826/903)
- SNTP (RFC 2030)
- DHCP/PPPoE
- IGMPv1/v2/v3
- HTTP Server for Web Management
- TFTP/HTTP for Auto Provisioning
- DHCP Option Codes for SIP (RFC3361)
- DNS/DNS SRV (RFC1706 and RFC 2782)
- IEEE802.1Q VLAN/802.1p and IP DSCP
- IEEE802.11e – WMM
- IEEE802.1x certification

- 
- Wireless**
- 802.11 b/g/n support
  - 4 SSID
  - Static WEP(64/128bit)
  - Dynamic WEP(64/128bit)
  - AES(CCMP/TKIP)
  - MAC certification
  - WEP certification
  - WPA-PAK/WPA-PSK(2.0)
  - WPA/WPA2
  - AP Isolation and MBSSID AP Isolation
  - WMM
  - WDS
  - WPS
  - Station list
  - Radius certification and billing
  - MAC Filter(different from wired MAC Filter)

- Applications**
- MAC address cloning
  - DDNS
  - Virtual Server for Port Mapping
  - SupportDMZ
  - Built-in Wired speed NAT Router
  - DHCP Server and Client
  - IGMP snooping and IGMP proxy
  - IP conflict detection
  - VPN Pass through (PPTP,L2TP and IPSec)
  - Speed Limit
  - QoS with Layer 2 and Layer 3
  - 4 Priority Queue with WRR and SPQ
  - Classification with the combination of physical port, 802.1p, DSCP, IP address and TCP/UDP port.
  - ARP spoofing prevention
  - Anti-DDOS attack
  - Support ACL for specific address, port, domain and network
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- Management**
- Firmware Upgrade
  - Web Management Interface
  - IVR-driven Management Interface
  - Local and Remote Syslog (RFC3164)
  - Auto Provisioning
  - SNTP Time Synchronization
  - Multi User Level
  - SNMP v2
  - Telnet
  - TR069
- 
- Dimension** • (L) 180 x (W) 110 x (H) 30 mm
- 
- Unit Weight** • 338 g ( without package)
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# AP+VOIP 802.11n 300M AP with VoIP and HNAT

G801

- 1 100M port (WAN) , 4 100M RJ45(4LAN)
- IEEE802.11n 2T2R 300Mbps wireless
- 5dBi external antenna
- Hardware NAT(HNAT) engine with 100Mbps NAT/NAPT speed
- Wireless throughput is above 80Mbps in average
- Rich QoS capability in wired and wireless
- Real time FAX over IP via T.38 and T.30 FAX with G.711
- PAMS is above 4.0 with G.729 Codec
- Auto-Provisioning-TFTP/HTTP/HTTPS,TR-069,SNMP



Size: (L) 180 x (W) 110 x (H) 30 mm

## PLACE ORDER

MODEL	DESC.
G801	1 100M port, 4 10/100M RJ45 ports, 1 FXS, 2T2R

- VoIP SoC** • CPU Core: MIPS24KEc(400MHz)
- Flash** • 8MBytes Flash
- RAM** • 32MBytes DDR2
- Power Supply** • AC/DC Adapter
- AC Input: 100~240V, 50~60Hz
- DC Output: 5V, 2A

- I/O Interfaces** • 1 RJ-45 for WAN port (Ethernet 10/100 Base-T)
- 4 RJ-45 for LAN port (Ethernet 10/100 Base-T)
- 1 RJ-11 for FXS port
- Two 5dB Antenna for IEEE802.11n 2T2R Wireless

- Environmental** • Operation Temperature: 0~45 Degree C
- Storage Temperature: -25~ 85 Degree C
- Relative Humidity: 10%~90% No Condensing
- Shock: Up to 75cm ( 30 inches)

- Key Feature** • Hardware NAT(HNAT) engine with 100Mbps NAT/NAPT speed
- Wireless throughput is above 80Mbps in average
- Support VoIP FXS port with SIP and T.38 FAX

- Audio Codec** • G.711 (A-Law, u-Law) with MOS above 4.0
- G.729 with MOS above 3.6
- Adaptive Jitter Buffer Management
- Voice Activity Detection
- Comfort Noise Generation
- Echo Cancellation
- T.30 FAX with G.711
- Real time FAX over IP via T.38

- Protocols** • SIP V2 (RFC 3261, 3262, 3263, 3264)
- Backward compatible with RFC2543
- Session timer (RFC4028)
- SDP (RFC2327)
- RTP/RTCP (RFC823/903)
- NAPTR for SIP URI lookup (RFC2915)
- STUN (RFC2030)
- ARP/RARP (RFC 826/903)
- SNTP (RFC 2030)
- DHCP/PPPoE
- HTTP Server for Web Management
- TFTP/HTTP 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 DSCP

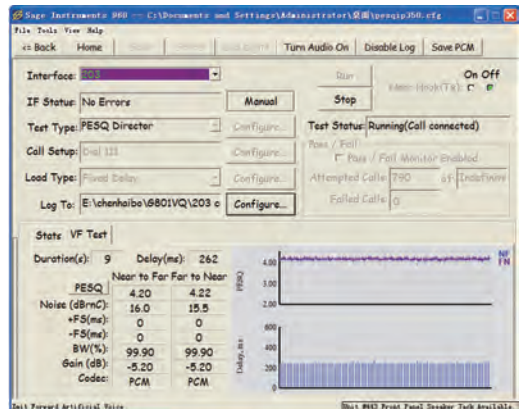
- Wireless**
- 802.11 b/g/n support
  - 4 SSID
  - Static WEP(64/128bit)
  - Dynamic WEP(64/128bit)
  - AES(CCMP/TKIP)
  - MAC certification
  - WEP certification
  - WPA-PAK/WPA-PSK(2.o)
  - WPA/WPA2
  - AP Isolation and MBSSID AP Isolation
  - WMM
  - WDS
  - WPS
  - Station list
  - Radius certification and billing
  - MAC Filter(different from wired MAC Filter)

- Applications**
- Support NAT mode and Bridge mode
  - Support DDNS
  - MAC address cloning
  - SIP proxy redundancy
  - NAT Traversal by STUN
  - Built-in Wired speed NAT Router
  - DHCP Server and Client
  - IP conflict detection
  - VLAN
  - Flow Control
  - VPN Pass through (PPTP,L2TP and IPSec)
  - Port forwarding and DMZ host
  - Rate Limit per port
  - Broadcast Storm Control
  - Classification by physical port, IP port, 802.1p,DSCP and TCP/UDP port
  - Marking by 802.1p and DSCP
  - MAC filter per port
  - Packer filter for port, IP address, protocol and physical port
  - Firewall defend attack and UDP flooding

- Management**
- Firmware Upgrade
  - Web Management Interface
  - IVR-driven Management Interface
  - Local and Remote Syslog (RFC3164)
  - Auto Provisioning
  - SNTP Time Synchronization
  - Multi User Level
  - SNMP v2
  - Telnet
  - TR069
  - System Log: local log and remote log

- Call Features**
- 3-way conference
  - Call hold
  - Call forwarding
  - Call Transfer
  - Call waiting
  - Caller ID and CWCID
  - DTMF Relay: In-band, Out-band and SIP INFO
  - QoS Layer 3
  - Call Log
  - CID/CWCID
  - Dial Plan
  - DTMF Mode: Inband, outband and SIP Info

- Dimension** • (L) 180 x (W) 110 x (H) 30 mm
- Unit Weight** • 338 g ( without package)



# AP 2.4G/5G Dual-band AP with Giga and HNAT

G900

- 5 1000M RJ45 ports(1 WAN / 4 LAN)
- 2.4GHz/5GHz Dual-band AP router
- Up to 340Mbps wireless bi-direction speed when 2.4G + 5G con-current
- Hardware NAT(HNAT) engine with 1000Mbps NAT/NAPT speed
- Auto-Provisioning-TFTP/HTTP/HTTPS,TR-069,SNMP



Size: (L) 180 x (W) 110 x (H) 30 mm

## PLACE ORDER

MODEL	DESC.
G900	5 1000M RJ45 port, 2T2R with 2.4GHz and 5GHz

**VoIP SoC** • CPU Core: MIPS 74KEc 500MHz with 64KB I-cache / 32KB D-cache

**Flash** • 16MB SPI Flash

**RAM** • 64MB DDR2

**Power** • AC/DC Adapter

**Supply** • AC Input: 100~240V, 50~60Hz  
• DC Output: 5V, 2A

**I/O Interfaces** • 1 RJ-45 for WAN port (Ethernet 10/100/1000 Base-T)  
• 4 RJ-45 for LAN port (Ethernet 10/100/1000 Base-T)  
• Two 5dB Antenna for IEEE802.11n 2T2R Wireless  
• 2.4G and 5G Dual band  
• 1 USB port

**Environmental** • Operation Temperature: 0~45 Degree C  
• Storage Temperature: -25~ 85 Degree C  
• Relative Humidity: 10%~90% No Condensing  
• Shock: Up to 75cm ( 30 inches)

**Key Feature** • Hardware NAT(HNAT) engine with 1000Mbps NAT/NAPT speed  
• Wireless throughput is up to 340Mbps in bi-direction and 2.4G/5G con-current mode

**Protocols** • IPv4 and IPv6  
• ARP/RARP (RFC 826/903)  
• SNTP (RFC 2030)  
• DHCP/PPPoE  
• IGMPv1/v2/v3  
• HTTP Server for Web Management  
• TFTP/HTTP for Auto Provisioning  
• DHCP Option Codes for SIP (RFC3361)  
• DNS/DNS SRV (RFC1706 and RFC 2782)  
• IEEE802.1Q VLAN/802.1p and IP DSCP  
• IEEE802.11e – WMM  
• IEEE802.1x certification

- 
- Wireless**
- 802.11 a/b/g/n support
  - 4 SSID for 2.4G and 4 SSID for 5G
  - Static WEP(64/128bit)
  - Dynamic WEP(64/128bit)
  - AES(CCMP/TKIP)
  - MAC certification
  - WEP certification
  - WPA-PAK/WPA-PSK(2.0)
  - WPA/WPA2
  - AP Isolation and MBSSID AP Isolation
  - WMM
  - WDS
  - WPS
  - Station list
  - Radius certification and billing
  - MAC Filter(different from wired MAC Filter)

- Applications**
- 
- MAC address cloning
  - DDNS
  - Virtual Server for Port Mapping
  - Support DMZ
  - Built-in Wired speed NAT Router
  - DHCP Server and Client
  - IGMP snooping and IGMP proxy
  - IP conflict detection
  - VPN Pass through (PPTP,L2TP and IPSec)
  - Speed Limit
  - QoS with Layer 2 and Layer 3
  - 4 Priority Queue with WRR and SPQ
  - Classification with the combination of physical port, 802.1p, DSCP, IP address and TCP/UDP port.
  - ARP spoofing prevention
  - Anti-DDOS attack
  - Support ACL for specific address, port, domain and network
- 

- 
- Management**
- Firmware Upgrade
  - Web Management Interface
  - IVR-driven Management Interface
  - Local and Remote Syslog (RFC3164)
  - Auto Provisioning
  - SNTP Time Synchronization
  - Multi User Level
  - SNMP v2
  - Telnet
  - TR069
- 
- Dimension**
- (L) 180 x (W) 110 x (H) 30 mm
- 
- Unit Weight**
- 338 g ( without package)
-



## IP Phone Series

Features/ Model	IP542N	IP652	IP622	IP601	IP350	WF400
LCD	128*64 Graphic LCD	128*64 Graphic LCD	128*32 Graphic LCD	No LCD	128*64 Graphic LCD	128*160 1.8" 65K color TFT LCD
Headset Port	3.5mm	RJ9	RJ9	No	No	3.5mm
Hand free	Yes					
802.3af PoE	Option					N/A
Battery	No					Li-ion 1300mAh
Wall-Mount Support	Yes	Yes	Yes	Yes	No	No
Extension Module	No	Yes, Up to 6 Modules ( EXP40)	No	No	No	No
SIP Accounts	4	5	2	1	2	1
Line Keys	4	5	2	No	No	No
Soft keys	4	3	4	No	No	2
Multi-function keys	No	20 ( Support 260 at Max with 6 EXP40 )	10	20 ( No BLF function )	No	N/A
Wireless	Yes, 802.11n 1T1R	No	No	No	No	Yes, 802.11b/g
Voice Codec	G.711(A-law, U-law), G.723, G.729A/B, G.722 ( Wide band )					
Integrated IVR	N/A	N/A	N/A	Yes	N/A	N/A
Management	Web browser, Auto-Provision with TFTP/HTTP/HTTPS, TR069, SNMP					

# IP Phone 4 lines IP SIP Phone with wireless and PoE

## IP542N/IP542NP

- SIP V2.0 ( RFC 3261, 3262, 3263, 3264 )
- 2 10/100M RJ45 ports
- 128\*64 graphic LCD with white backlight
- 4 Line-Keys & 4 SIP accounts
- 4 software defined keys for LCD menu
- Compliant with IEEE 802.3af Power over Ethernet (PoE)- Optional
- Headset jack and Wall-mount supported
- Built-in internal antenna, 1T1R, 802.11N
- Auto-Provisioning- TFTP/HTTP/HTTPS,TR-069, SNMP
- HD Voice and HD Codec G.722 supported



Size: (L) 220 x (W) 187 x (H) 97 mm

### PLACE ORDER

MODEL	DESC.
IP542N	128*64 Blank-White graphic LCD, 2 10/100M RJ45 ports, 802.11n 1T1R, w/o PoE
IP542NP	128*64 Blank-White graphic LCD, 2 10/100M RJ45 ports, 802.11n 1T1R, with PoE

**VoIP SoC** • CPU Core:MIPS24KEc@360MHZ

**Flash** • 8MBytes Flash

**RAM** • 32MBytes SDRAM

**Display** • 128\*64 Graphic LCD with white back light

**Phone Keys** • 4 soft keys

- 4 Line keys with red-green dual color LEDs for indication
- MENU, SMS, HOLD
- Volume+, Volume- and navigate key MUTE
- Headphone, Handsfree
- 12 Dialing Buttons (0~9, \*, #),

**Power Supply** • AC/DC Adapter  
 • AC Input: 100~240V, 50~60Hz  
 • DC Output: 5V, 1A  
 • IEEE802.3af PoE in WAN port ( IP542P only )

**I/O Interfaces** • 2 RJ-45 ports (Ethernet 10/100 Base-T)  
 • 3.5mm earphone jack  
 • 3dB internal antenna for 802.11n 1T1R  
 • Full duplex hands-free speakerphone

**Environmental** • Operation Temperature: 0~50 Degree C  
 • Storage Temperature: -25~ 85 Degree C  
 • Relative Humidity: 10%~90% No Condensing  
 • Shock: Up to 75cm (30 inches) `Drop upon Package

**Audio Codec** • G.711 (A-Law,  $\mu$ -Law) with PAMS above 4.3  
 • G.723.1  
 • G.729A/AB with PAMS above 4.0  
 • Adaptive Jitter Buffer Management  
 • Voice Activity Detection  
 • Comfort Noise Generation  
 • Echo Cancellation

**Protocols** • SIP V2 (RFC 3261,3262,3263,3264)  
 • Backward Compatible with RFC2543  
 • Session Timer (RFC4028)  
 • SDP (RFC2327)  
 • RTP/RTCP (RFC1889 and RFC1890)  
 • NAPTR for SIP URI Lookup (RFC2915)  
 • STUN (RFC 3489)  
 • ARP/RARP (RFC 826/903)  
 • SNTP (RFC 2030)  
 • DHCP server and client  
 • 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

# IP Phone 4 lines IP SIP Phone with wireless and PoE

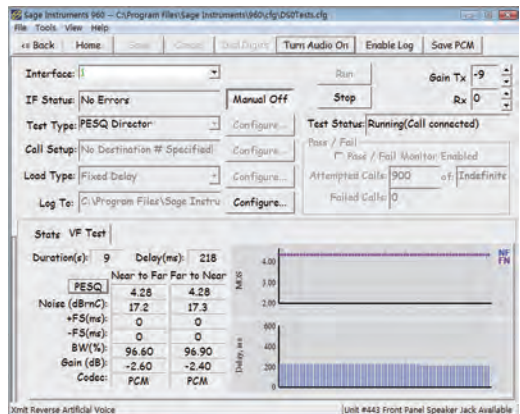
## IP542N/IP542NP

- 
- Applications**
- SMS Function
  - MAC Address Cloning
  - SIP proxy redundancy: dynamic via DNS SRV, A records
  - Direct IP to IP calling
  - Alarm Clock
  - Day Light Saving
  - NAT Traversal: Traversal by STUN
  - Built-in NAT Router
  - VPN
  - QoS with Layer 2 and Layer 3
  - DHCP Server and Client
  - IP conflict detection
  - Support PoE comply with IEEE802.3af(optional)
- 

- Management**
- Firmware Upgradeable
  - Web Management Interface
  - Password Management
  - Local and Remote Syslog (RFC3164)
  - Auto Provisioning: TFTP, HTTP and HTTPS
  - SNTP Time Synchronization
  - Multi User Level
  - SNMPv2
  - TR069
- 

- Call Features**
- 4 lines
  - Auto Answer
  - 3-way Conference
  - Music on hold
  - Call Hold
  - Call Forwarding
  - Call Mute
  - Call Transfer
  - Call Waiting
  - Call Pickup
  - MWI
  - CID and CWCID
  - DTMF Relay: In-band, Out-band(RFC2833) and SIP Info
  - Full-duplex Speakerphone
  - Delayed Hotline
  - Redial
  - Dial Plan
  - Volume Adjustment
    - Handset/Headset, Speaker and Ringer
- 

- 
- Phone feature**
- 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)
- 
- Dimension**
- (L) 220 x (W) 187 x (H) 97 mm
- 
- Unit Weight**
- 602 g ( without package)
- 



- SIP V2.0 ( RFC 3261, 3262, 3263, 3264 )
- 2 10/100M RJ45 ports
- 28\*64 graphic LCD with white backlight
- 5 Line-Keys & 5 SIP accounts
- 3 software defined keys for LCD menu
- 20 multi-function keys with green/red dual-color LEDs
- Up to 6 expansion modules with 40 Multi-function keys ( EXP40 )
- Compliant with IEEE 802.3af Power over Ethernet (PoE)- Optional
- Auto-Provisioning-TFTP/HTTP/HTTPS,TR-069, SNMP
- HD Voice and HD Codec G.722 supported



Size: (L) 230 x (W) 168 x (H) 81 mm

## PLACE ORDER

MODEL	DESC.
IP652	128*64 Black-White graphic LCD, 5 lines, 2 10/100M RJ45 ports, RJ9, EXT port, w/o PoE
IP652P	128*64 Black-White graphic LCD, 5 lines, 2 10/100M RJ45 ports, RJ9, EXT port, with PoE

**VoIP SoC** • CPU Core: MIPS24Kec(360M)

**Flash** • 8MBytes Flash

**RAM** • 32MBytes SDRAM

**Display** • 128\*64 Graphic LCD with white back light

**Phone Keys** • 5 Line-Keys & softkey button  
 • XFER, HOLD, CONF, MUTE/DEL  
 • MSG, MENU, REDIAL, Handsfree/headset  
 • 20 multi-functional key with dual-color LED

**LED** • 5 LEDs used to indicate line status  
 • 1 MSG LED  
 • 1 Handsfree/Headset LED  
 • 20 LEDs used to indicate multi-functional key status

**Power Supply** • AC/DC Adapter  
 • AC Input: 100~240V, 50~60Hz  
 • DC Output: 5V, 1A  
 • IEEE802.3af PoE in WAN port ( IP652P only )

**I/O Interfaces** • 1 RJ-45 for Internet port (Ethernet 10/100 Base-T)  
 • 1 RJ-45 for PC port (Ethernet 10/100 Base-T)  
 • 1 EXT port  
 • 1 headset port  
 • 1 DND switch

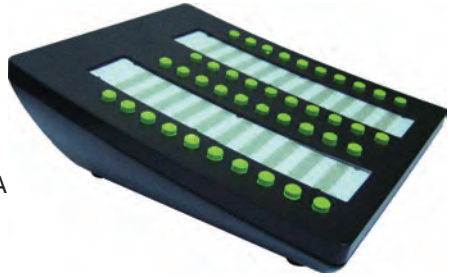
**Key Features** • Support 20 multi-functional keys  
 • Support up to 6 expansion modules (EXP40) with 260 multi-functional keys

**Environmental** • Operation Temperature: 0~50 Degree C  
 • Storage Temperature: -25~ 85 Degree C  
 • Relative Humidity:10%~90% No Condensing  
 • Shock:Up to 75cm (30 inches) `Drop upon Package

**Audio Codec** • G.711 (A-Law,  $\mu$ -Law) with PAMS above 4.3  
 • G.729A/AB with PAMS above 4.0  
 • G.723: 5.3kbps and 6.3 kbps  
 • G.722 (high quality)  
 • Adaptive Jitter Buffer Management  
 • Voice Activity Detection  
 • Comfort Noise Generation  
 • Echo Cancellation

<b>Protocols</b>	<ul style="list-style-type: none"> <li>• SIP V2 (RFC 3261,3262,3263,3264)</li> <li>• Backward Compatible with RFC2543</li> <li>• Session Timer (RFC4028)</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>• HTTP Server for Web Management</li> <li>• TFTP/HTTP/HTTPS for Auto Provisioning</li> <li>• Message Waiting Indicator (RFC3842)</li> <li>• DNS/DNS SRV (RFC1706 and RFC 2782)</li> </ul>	<b>Call Features</b>	<ul style="list-style-type: none"> <li>• 5 lines</li> <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>• Redial</li> <li>• Speed Dial</li> <li>• BLF</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>
<b>Applications</b>	<ul style="list-style-type: none"> <li>• MAC: Address Cloning</li> <li>• VPN: PPTP and T2TP</li> <li>• DMZ</li> <li>• Layer 2 QoS: 802.1Q/VLAN ID and 802.1p PRI</li> <li>• Layer 3 QoS: SIP QoS, RTP QoS and Data QoS</li> <li>• SIP proxy redundancy: dynamic via DNS SRV, A records</li> <li>• Direct IP to IP calling</li> <li>• NAT Traversal: Static NAT Route and Traversal by STUN</li> <li>• Built-in NAT Router</li> <li>• DHCP Server and Client</li> <li>• IP conflict detection</li> <li>• Support PoE comply with IEEE802.3af(optional)</li> </ul>	<b>Phone feature</b>	<ul style="list-style-type: none"> <li>• Customization 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 (20 records )</li> </ul>
<b>Management</b>	<ul style="list-style-type: none"> <li>• Menu Configuration</li> <li>• SNTP Time Synchronization</li> <li>• Daylight Saving Time</li> <li>• Alarm Clock</li> <li>• Password Reset</li> <li>• Web access management</li> <li>• Local and Remote Syslog (RFC3164)</li> <li>• Factory Default</li> <li>• Firmware Upgradeable</li> <li>• Web Management Interface</li> <li>• Auto Provisioning: TFTP, HTTP and HTTPS</li> <li>• Multi User Level</li> <li>• SNMPv2</li> <li>• TR069</li> </ul>	<b>Dimension</b>	<ul style="list-style-type: none"> <li>• (L) 230 x (W) 168 x (H) 81 mm</li> </ul>
		<b>Unit Weight</b>	<ul style="list-style-type: none"> <li>• 730g (without package)</li> </ul>

- 40 multi-function keys with green/red dual-color LEDs
- 2 RJ11 Jack for extension module connection
- 1 DC Jack for standby 5V/1A
- Power source from the host IP652, or the standby 5V/1A power adapter



Size: (L)167mm X (W)132mm X (H)51mm

### PLACE ORDER

MODEL	DESC.
EXP40	40 multi-function keys extension module with dual-color LEDs

**Keys** • 40 multi-functional key with dual-color LED

**Power Supply** • 5V/1A Power Adapter or IP652

**Key Features** • 40 multi-functional key can be used as speed dial, BLF, shortcut Key, pick up, call park  
 • Daisy chain for 6 EXP40 at Max.  
 • 260 multi-functional keys at Max.

**Protocols** • SIP V2 (RFC 3261, 3262, 3263, 3264)  
 • Backward Compatible with RFC2543

**LED** • 40 Red-Green dual-color LEDs used to indicate multi-functional key status

**I/O Interfaces** • 1 RJ11 Jack for extension module connection

**Environmental** • Operation Temperature: 0~50 Degree C  
 • Storage Temperature: -25~ 85 Degree C  
 • Relative Humidity: 10%~90% No Condensing  
 • Shock: Up to 75cm (30 inches) Drop upon Package

**Dimension** • (L)167mm X (W)132mm X (H)51mm

**Unit Weight** • 240g

- SIP V2.0 ( RFC 3261, 3262, 3263, 3264 )
- 2 10/100M RJ45 ports
- 128\*32 graphic LCD with white backlight
- 2 Line-Keys & 2 SIP accounts
- 4 software defined keys for LCD menu
- 10 multi-function keys with green/red dual-color LEDs
- Compliant with IEEE 802.3af Power over Ethernet (PoE)- Optional
- Headset jack and Wall-mount supported
- Auto-Provisioning-TFTP/HTTP/HTTPS,TR-069, SNMP
- HD Voice and HD Codec G.722 supported



Size: (L) 191 x (W) 205 x (H) 75 mm

## PLACE ORDER

MODEL	DESC.
IP622	128*32 Black-White graphic LCD, 2 lines, 2 10/100M RJ45 ports, RJ9, w/o PoE
IP622P	128*32 Black-White graphic LCD, 2 lines, 2 10/100M RJ45 ports, RJ9, with PoE

**VoIP SoC** • CPU Core: MIPS24KEc(360M)

**Flash** • 8MBytes Flash

**RAM** • 16MBytes SDRAM

**Display** • 128x32 Graphic LCD with white back light

**Phone Keys** • 2 Line keys and 4 soft keys, MENU,UP,DOWN,MUTE/DEL, REDIAL  
 • HOLD, MSG, Headset, Handsfree  
 • 12 Dialing Buttons (0~9, \*, #)  
 • 10 Multi-functional key with dual-color LED

**LED** • 2 Line LEDs used to indicate line status  
 • 1 Handsfree, 1 Headset LED  
 • 10 LEDs used to indicate multi-functional key status

**Power Supply** • AC/DC Adapter  
 • AC Input: 100~240V, 50~60Hz  
 • DC Output: 5V, 1A  
 • IEEE802.3af PoE in WAN port ( IP622P only )

**I/O Interfaces** • 2 RJ-45 port (10/100 Ethernet Ports)  
 • 1 RJ9 for headset  
 • 1 DND switch  
 • Full duplex hands-free speaker phone  
 • Headset jack and Wall-mount

**Environmental** • Operation Temperature: 0~50 Degree C  
 • Storage Temperature: -25~ 85 Degree C  
 • Relative Humidity:10%~90% No Condensing  
 • Shock:Up to 75cm (30 inches) `Drop upon Package

**Audio Codec** • G.711 (A-Law,  $\mu$ -Law) with PAMS above 4.3  
 • G.729A/AB with PAMS above 4.0  
 • G.723: 5.3kbps and 6.3kbps  
 • G.722  
 • Adaptive Jitter Buffer Management  
 • Voice Activity Detection  
 • Comfort Noise Generation  
 • Echo Cancellation

<b>Protocols</b>	<ul style="list-style-type: none"> <li>• SIP V2 (RFC 3261,3262,3263,3264)</li> <li>• Backward Compatible with RFC2543</li> <li>• Session Timer (RFC4028)</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>• HTTP Server for Web Management</li> <li>• TFTP/HTTP/HTTPS for Auto Provisioning</li> <li>• Message Waiting Indicator (RFC3842)</li> <li>• DNS/DNS SRV (RFC1706 and RFC 2782)</li> <li>• IEEE802.1Q VLAN/802.1p and IP TOS</li> </ul>	<b>Call Features</b>	<ul style="list-style-type: none"> <li>• 4 lines</li> <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>• BLF</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>
<b>Management</b>	<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> <li>• SNMPv2</li> <li>• TR069</li> </ul>	<b>Phone feature</b>	<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>
<b>Applications</b>	<ul style="list-style-type: none"> <li>• MAC address cloning</li> <li>• SIP proxy redundancy: dynamic via DNS SRV, A records</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>	<b>Dimension</b>	<ul style="list-style-type: none"> <li>• (L) 191 x (W) 205 x (H) 75 mm</li> </ul>
		<b>Unit Weight</b>	<ul style="list-style-type: none"> <li>• 720g (without package)</li> </ul>

- SIP V2.0 ( RFC 3261, 3262, 3263, 3264 )
- 1 10/100M RJ45 ports
- 20 multi-function keys without LEDs and BLF function
- Compliant with IEEE 802.3af Power over Ethernet (PoE)- Optional
- Wall-mount supported
- Auto-Provisioning-TFTP/HTTP/HTTPS,TR-069, SNMP
- HD Voice and HD Codec G.722 supported



Size: (L)185mm X (W)146mm X (H)60mm

## PLACE ORDER

MODEL	DESC.
IP601	1 10/100M RJ45 ports, No LCD, 1 DND switch, 20 multi-function keys, w/o PoE
IP601P	1 10/100M RJ45 ports, No LCD, 1 DND switch, 20 multi-function keys, with PoE
IP601A	1 10/100M RJ45 ports, No LCD, No DND switch, No multi-function keys, w/o PoE

**VoIP SoC** • CPU Core: MIPS24KEc(360M)

**Flash** • 8MBytes Flash

**RAM** • 16MBytes SDRAM

**Phone Keys** • XFER, HOLD, CONF, Speaker

• 12 Dialing Buttons (0~9, \*, #)

• 20 speed dial keys

**Power Supply** • AC/DC Adapter

• AC Input: 100~240V, 50~60Hz

• DC Output: 5V, 1A

• IEEE802.3af PoE in WAN port ( IP601P only )

**I/O Interfaces** • 1 RJ-45 port (10/100 Ethernet Ports)

• 1 DND switch

• Full duplex hands-free speaker phone

• Wall-mount

**Environmental** • Operation Temperature: 0~50 Degree C

• Storage Temperature: -25~ 85 Degree C

• Relative Humidity:10%~90% No Condensing

• Shock:Up to 75cm (30 inches) `Drop upon Package

**Audio Codec** • G.711 (A-Law,  $\mu$ -Law) with PAMS above 4.3

• G.729A/AB with PAMS above 4.0

• G.723: 5.3kbps and 6.3 kbps

• G.722 (high quality)

• Adaptive Jitter Buffer Management

• Voice Activity Detection

• Comfort Noise Generation

• Echo Cancellation

- Protocols**
- SIP V2 (RFC 3261,3262,3263,3264)
  - Backward Compatible with RFC2543
  - Session Timer (RFC4028)
  - SDP (RFC2327)
  - RTP/RTCP (RFC1889 and RFC1890)
  - NAPTR for SIP URI Lookup (RFC2915)
  - STUN (RFC 3489)
  - ARP/RARP (RFC 826/903)
  - SNTP (RFC 2030)
  - DHCP
  - HTTP Server for Web Management
  - TFTP/HTTP/HTTPS for Auto Provisioning
  - Message Waiting Indicator (RFC3842)
  - DNS/DNS SRV (RFC1706 and RFC 2782)
  - IEEE802.1Q VLAN/802.1p and IP TOS

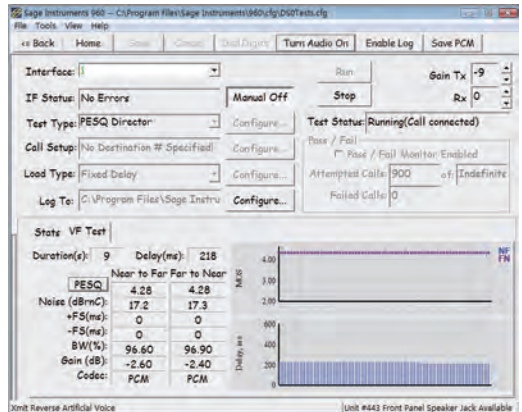
- Management**
- Firmware Upgradable
  - Web Management Interface
  - Local and Remote Syslog (RFC3164)
  - Auto Provisioning: TFTP, HTTP and HTTPS
  - SNTP Time Synchronization
  - Two User Level Configuration
  - SNMPv2
  - TR069

- Applications**
- MAC Address Cloning
  - SIP proxy redundancy: dynamic via DNS SRV, A records
  - Direct IP to IP calling
  - NAT Traversal: STUN
  - QoS with Layer 2 and Layer 3
  - DHCP Server and Client
  - IP conflict detection
  - Support PoE comply with IEEE802.3af(optional)

- Call Features**
- 3-way Conference
  - Music on hold
  - Call Hold
  - Call Forwarding
  - Call Transfer
  - Call Waiting
  - DTMF Relay: In-band, Out-band(RFC2833) and SIP INFO
  - Full-duplex Speakerphone
  - Delayed Hotline
  - Speed Dial
  - Dial Plan
  - MWI
  - DND
  - Volume Adjustment: Handset, Speaker and Ringer Phone Features
  - Customized Ring Tone
  - Call History ( 100 records )
    - Most Recently Missed Calls
    - Most Recently Received Calls
    - Most Recently Dialed Numbers
  - Blank list
  - Speed Dial (20 records)

**Dimension** • (W)185mm X (H)146mm X (D)60mm

**Unit Weight** • 510 g ( without package)



- SIP V2.0 ( RFC 3261, 3262, 3263, 3264 )
- 2 10/100M RJ45 ports
- 128\*64 graphic LCD with white backlight
- 2 SIP accounts with one switch key
- Compliant with IEEE 802.3af Power over Ethernet (PoE)- Optional
- Auto-Provisioning-TFTP/HTTP/HTTps,TR-069,SNMP



Size: (L) 220 x (W) 180 x (H) 90 mm

### PLACE ORDER

MODEL	DESC.
IP350	128*64 Black-White graphic LCD, 2 lines, 2 10/100M RJ45 ports, w/o PoE
IP350P	128*64 Black-White graphic LCD, 2 lines, 2 10/100M RJ45 ports, with PoE

- VoIP SoC** • CPU Core: MIPS24KEc(360M)
- Flash** • 8MBytes Flash
- RAM** • 16MBytes SDRAM
- Display** • 128\*64 Graphic LCD with white back light
- Phone Keys** • Phone book, Call log, MSG, Pick Up, Delete  
• Mute,3-way Call, Transfer, DND  
• Hold, Redial, Speaker  
• Menu, Enter, Cancel, Line Switcher, Vol+/Up, Vol-/Down  
• 12 Dialing Buttons (0~9, \*, #),

- Power Supply** • AC/DC Adapter  
• AC Input: 100~240V, 50~60Hz  
• DC Output: 5V, 1A  
• IEEE802.3af PoE in WAN port (for IP350P only)

- I/O Interfaces** • 2 RJ-45 for 10/100 Ethernet Ports that can be configured as either switched or routed ports  
• Full duplex hands-free speaker phone

- Environmental** • Operation Temperature: 0~50 Degree C  
• Storage Temperature: -25~ 85 Degree C  
• Relative Humidity:10%~90% No Condensing  
• Shock:Up to 75cm (30 inches) ` Drop upon Package

- Audio Codec** • G.711 (A-Law,  $\mu$ -Law)  
• G.729A/AB  
• Adaptive Jitter Buffer Management  
• Voice Activity Detection  
• Comfort Noise Generation  
• Echo Cancellation

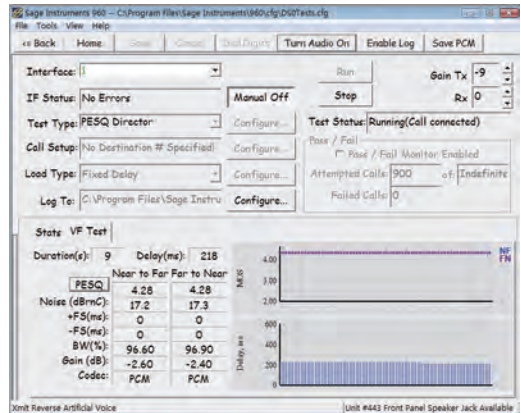
- Protocols** • SIP V2 (RFC 3261,3262,3263,3264)  
• Backward Compatible with RFC2543  
• Session Timer (RFC4028)  
• SDP (RFC2327)  
• RTP/RTCP (RFC1889 and RFC1890)  
• NAPTR for SIP URI Lookup (RFC2915)  
• STUN (RFC 3489)  
• ARP/RARP (RFC 826/903)  
• SNTP (RFC 2030)  
• DHCP  
• HTTP Server for Web Management  
• TFTP/HTTP/HTTPS for Auto Provisioning  
• Message Waiting Indicator (RFC3842)  
• DNS/DNS SRV (RFC1706 and RFC 2782)  
• IEEE802.1Q VLAN/802.1p and IP TOS

- Applications**
- SMS Functions
    - MAC Address cloning
    - SIP proxy redundancy
      - dynamic via DNS SRV, A records
    - Direct IP to IP calling
    - NAT Traversal by STUN
    - Built-in NAT Router
    - QoS with Layer 2 and Layer 3
    - DHCP Server and Client
    - IP conflict detection
    - Support PoE comply with IEEE802.3af(optional)

- Management**
- Firmware Upgradeable
  - Web Management Interface
  - Local and Remote Syslog (RFC3164)
  - Auto Provisioning
  - SNTP Time Synchronization
  - Multi User Level
  - SNMPv2

- Call Features**
- 3-way Conference
  - Music on hold
  - Call Hold
  - Call Forwarding
  - Call Mute
  - Call Transfer
  - Call Waiting
  - Caller ID
  - DTMF Relay:
    - -In-band, Out-band(RFC2833)
    - -SIP INFO
  - Full-duplex Speakerphone
  - Hotline
  - Redial
  - Speed Dial
  - Dial Plan
  - Phonebook
  - 2 lines
  - MWI
  - Call log
  - DND
  - Volume Adjustment
    - Handset/Headset, Speaker and Ringer

- Phone feature**
- 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 )
- Dimension** • (L) 220 x (W) 180 x (H) 90 mm
- Unit Weight** • 765 g ( without package )



- 1.77" 65K color TFT LCD (128\*64 Pixel)
- Compliant with SIPV2 ( RFC 3261, 3262, 3263, 3264 )
- IEEE802.11b/g with 2.4G Derivation Model
- WEP, WPA/WPA2 encryption supported
- Support WMM(WiFi Multimedia) Qos
- Embedded web browser
- Auto-Provisioning-TFTP/HTTP/HTTps,TR-069,SNMP



Size: (L)140 x(W)50 x (H)72 mm

### PLACE ORDER

MODEL	DESC.
WF400	1.8" 65K color TFT LCD, 802.11b/g WiFi phone

<b>VoIP SoC</b>	• CPU Core: ARM926E@300MHz
<b>Flash</b>	• 64Mbytes NAND Flash
<b>RAM</b>	• 32MBytes SDRAM
<b>Display</b>	• 65K TFT Color LCD • 128x160 Pixel • Progressive back light
<b>Power Supply</b>	• Charge Input: 100~240Vac, 7~63Hz • Charge Output: 4.2Vdc, 720mA • Battery: Li-ion 1300mAh • Charge Time: 2 Hour • Standard by: up to 72 hours (approx.) • Talk time: up to 4 hours(approx.)
<b>Handset Keys</b>	• Send,End,5-way navigation key, 4-soft keys • Volume up/down side button • Keypad 0~9,*,#
<b>Wireless</b>	• IEEE802.11b/g supported • Frequency band: 2.4G • Output power: 15dBm(802.11b), 15dBm(802.11g at6~24Mbps) • RX Sensitivity: -87dBm@11Mbit/sec, -74dBm@54Mbit/sec • WEP, WPA/WPA2 supported
<b>Environmental</b>	• Operation Temperature: 0~50 Degree C • Storage Temperature: -25~ 85 Degree C • Relative Humidity:10%~90% No Condensing • Shock: Up to 75cm (30 inches) ` Drop upon Package

<b>Audio Codec</b>	• G.711 (A-Law, $\mu$ -Law) with PAMS above 4.3 • G.729A/AB with PAMS above 4.0 • Adaptive Jitter Buffer Management • Voice Activity Detection • Comfort Noise Generation • Echo Cancellation
<b>Protocols</b>	• SIP V2 (RFC 3261,3262,3263,3264) • Backward Compatible with RFC2543 • Session Timer (RFC4028) • SDP (RFC2327) • RTP/RTCP (RFC1889 and RFC1890) • NAPTR for SIP URI Lookup (RFC2915) • STUN (RFC 3489) • ARP/RARP (RFC 826/903) • SNTP (RFC 2030) • DHCP server and client • HTTP Server for Web Management • TFTP/HTTP 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

- 
- Applications**
- WMM™ for QoS
  - SMS Functions
  - SIP proxy redundancy: dynamic via DNS SRV, A records
  - NAT Traversal: traversal by STUN
  - Downloadable music ringing tones
- 

- Management**
- Firmware Upgradeable
  - Web Management Interface
  - Local and Remote Syslog (RFC3164)
  - Auto Provisioning
  - SNTP Time Synchronization
  - Multi User Level
  - SNMPv2
  - TR069
- 

- Call Features**
- 3-way Conference
  - Music on hold
  - Call Hold
  - Call Forwarding
  - Call Mute
  - Call Transfer
  - Call Waiting
  - Caller ID
  - DTMF Relay: In-band, Out-band(RFC2833) and SIP INFO
  - Full-duplex Speakerphone
  - Hotline
  - Redial
  - Speed Dial
  - Dial Rule
  - Volume Adjustment
  - Handset/Headset, Speaker and Ringer
- 

- Phone feature**
- 40 Polyphonic MIDI Ring
  - Call History ( 100 records )
  - Most Recently Missed Calls
  - Most Recently Received Calls
  - Most Recently Dialed Numbers
  - Phone book ( 500 records )
  - Speed Dial ( 99 records )
- 

**Dimension** (L)140 x(W)50 x (H)27 mm

---

**Unit Weight** 113g (without package)

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- Internet Radio with hand free VoIP talk
- 3 channel for internet radio
- 3 hotline for hand free VoIP talk
- Compliant with SIPV2 ( RFC 3261, 3262, 3263, 3264 )
- Support G.729, G.711A/u, WMA, AAC, MP3, OGG and WAV
- Support both uni-cast and multicast
- Auto-provision with TFTP, SNMP
- High volume mono speaker with 48K/44.1K Sample Rate digital audio
- Table-top or Wall-mount
- Built-in internal antenna, 1T1R, 802.11n



Size: (L) 200 x (W) 108 x (H) 36 mm

## PLACE ORDER

MODEL	DESC.
FIV378	1 10/100M RJ45 port, 3 Internet radio channel, 3 hotline button for VoIP hand free talk

<b>VoIP SoC</b>	• CPU Core: MIPS24KEc(360M)
<b>Flash</b>	• 4MBytes Flash
<b>RAM</b>	• 16MBytes SDRAM
<b>Buttons</b>	• 3 Radio Button • 3 SIP Speed dial Button for VoIP talk • Power Button • Volume Control knob • Reset Button
<b>LEDS</b>	• Power LED • Trouble Shooting LED • Status LED
<b>Ethernet port</b>	• 1 RJ-45 Internet port (Ethernet 10/100 Mbps) • Support Auto Negotiation • Support Auto MDI/MDIX
<b>Power Supply</b>	• AC/DC Adapter • AC Input: 100~240V, 50~60Hz • DC Output: 5V, 2A
<b>Environmental</b>	• Operation Temperature: 0~50 Degree C • Storage Temperature: -25~ 85 Degree C • Relative Humidity:10%~90% No Condensing • Shock:Up to 75cm (30 inches) ` Drop upon Package

<b>Radio Features</b>	• 3 Radio • Volume level adjustment • UDP multicast • Formats: MP3, WAV, AAC, OGG, WMA
<b>Protocols</b>	• SIP V2 (RFC 3261,3262,3263,3264) • Backward Compatible with RFC2543 • Session Timer (RFC4028) • SDP (RFC2327) • RTP/RTCP (RFC1889 and RFC1890) • NAPTR for SIP URI Lookup (RFC2915) • STUN (RFC 3489) • ARP/RARP (RFC 826/903) • SNTP (RFC 2030) • DHCP Client • HTTP Server for Web Management • Auto Provisioning: TFTP/HTTP/HTTPS • DHCP Option Codes for SIP (RFC3361) • DNS/DNS SRV (RFC1706 and RFC 2782) • QoS: Layer 3 and Layer 2

<b>Applications</b>	<hr/> <ul style="list-style-type: none"><li>• NAT Traversal: Traversal by STUN</li><li>• DHCP Client</li><li>• DNS</li><li>• Radio</li><li>• UDP Multicast</li><li>• Daylight Saving Time</li><li>• Supported table-top</li><li>• Mono speaker</li><li>• Microphone</li><li>• IVR function</li><li>• Prompt on radio connection status</li><li>• Idle Timeout</li></ul> <hr/>
<b>Management</b>	<ul style="list-style-type: none"><li>• Firmware Upgradeable</li><li>• Web Management Interface</li><li>• Local and Remote Syslog (RFC3164)</li><li>• Auto Provisioning: TFTP</li><li>• SNTP Time Synchronization</li><li>• SNMPv2</li></ul> <hr/>
<b>Dimension</b>	<ul style="list-style-type: none"><li>• (L) 200 x (W) 108 x (H) 36 mm</li></ul> <hr/>
<b>Unit Weight</b>	<ul style="list-style-type: none"><li>• 270 g ( without package)</li></ul> <hr/>

- 5 10/100/1000M RJ45 ports
- 2 FXS ports, 4/8 FXO ports
- 1 SD card slot , 1 USB ports
- 100 registered extensions
- 30 synchronous calls
- 75 hours voicemail, 5000 fax pages, 2 hours video mail.



### PLACE ORDER

Size:19 inches1U

MODEL	DESC.
APX5024	5 10/100/1000M RJ45 ports 2 FXS, 4 FXO
APX5028	5 10/100/1000M RJ45 ports 2 FXS, 8 FXO

**VoIP SoC** • CPU Core:MIPS 74KEc 500MHz with 64KB I-cache / 32KB D-cache

**RAM** • 256MB DDR2

**Flash** • 8MBytes Flash for BIOS, SD Memory Card for Asterisk

**Power Supply LED** • AC Input: 100~240V, 50~60Hz  
• 1 power led for indicating power on/off.  
• 1 Internet led for indicating Internet connecting

• 4 Ethernetled for indicating Ethernet connecting  
• 8 lines led for indicating line status

**Environmental** • Operation Temperature: 0~45 Degree C  
• Storage Temperature: -25~ 85 Degree C  
• Relative Humidity: 10%~90% No Condensing  
• Shock: Up to 75cm ( 30 inches)

**Key Features** • Running Asterisk 1.8 with Free PBX edition  
• Source code opened in [www.asteriskmips.org](http://www.asteriskmips.org)  
• 100 registered extensions  
• 10 SIP lines  
• 8FXO lines  
• 30 synchronous calls  
• Conference room (G.711):  
• two for 10 persons  
• one for 20 persons  
• Unified Message storage:  
• 75 hours of voicemail  
• 5000 fax pages  
• 2 hours of video mail

**I/O Interfaces** • 1 RJ45 for WAN port( 10/100/1000M )  
• 4 RJ45 for LAN ports ( 10/100/1000M )  
• 1 RS232 Serial Port  
• 1 SD Slot for Flash Memory Card  
• 1 USB port for 3G card and printer  
• 2 RJ11 ports for FXS  
• 8 RJ11 ports for FXO  
• 1 Power ON/OFF switch  
• 1 button for reset default

**Dimension** • 19 inches1U

# VoIP Products with Excellent Voice Quality

Our products passed the strict carrier-grade benchmark test, The benchmark test is a professional and comprehensive test including environment, function and capability parts and it is the essential condition to enter the VOIP telecom carrier's market. Our product performance is excellent especially in long run test and call completion rate test. The description of Voice Quality measured by Agilent Voice Quality Tester (VQT) under long run test and call completion rate test incorporated with environmental Test were listed below.

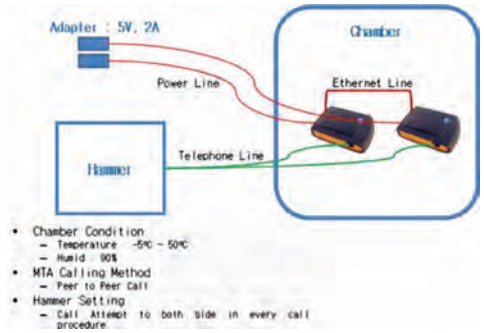


Fig 1 Chamber Test Environment

## 1.Call Completion Rate incorporated with Environmental Test

The call completion rate test was done incorporated with environmental test in which the temperature and humidity changed sharply from -5°C~ 50°C and 0~ 90%. The call completion rate of GIX100 is above 98% at the rate of 180 calls per hour for continuing 72 hours incorporated such protean environment.

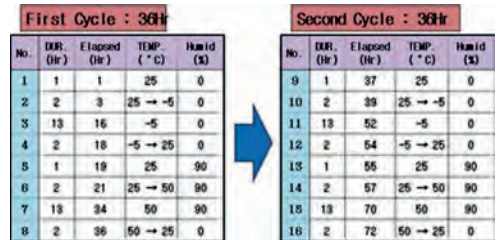


Fig 2 chamber test temperature and humidity change graph

## 2.Long Run Test

In 8 hours continuing long run test, The voice quality of GIX100 is stable and as good as the beginning of getting through not matter using G.711, G.711U or G.729 codec after a long run call lasted for 8 hours. And in the telecom carrier requirement the standard voice quality for G.729 and G.711 PAMS score should keep above 3.6 and 4.0. Below is the test environment graph and G.711A, G.711U and G.729 PAMS score of GIX100 for 8 hours long run which were tested by Agilent Voice Quality Tester (VQT).

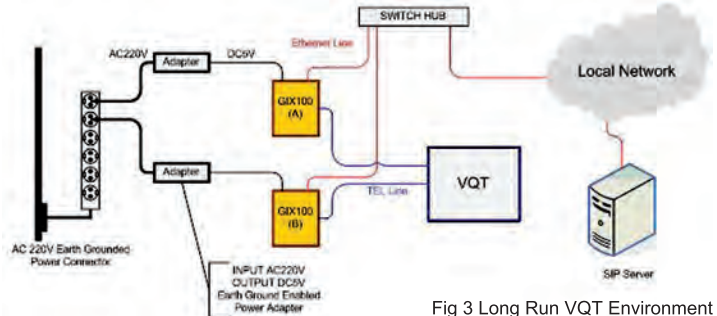


Fig 3 Long Run VQT Environment

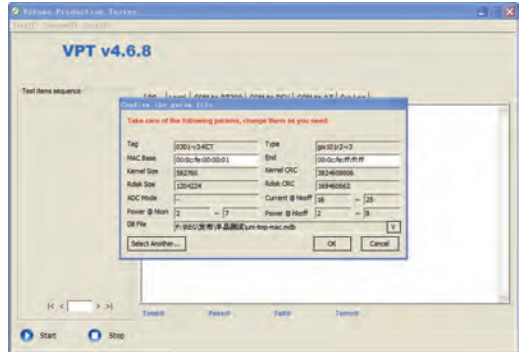
Through strict quality control and tests in production procedure, EASY3CALL delivery quality products to all customers.

Production procedure:

SMT→DIP→Semi-board test→Assembly→Aging (4 hours)→Products test (final test) →packing

Semi-board test items:

- \_\_\_\_\_ system test
- \_\_\_\_\_ minimum system test
- \_\_\_\_\_ SDRAM test
- \_\_\_\_\_ FLASH test
- \_\_\_\_\_ WAN-LAN test
- \_\_\_\_\_ FXS hook on test
- \_\_\_\_\_ FXS hook on power test
- \_\_\_\_\_ Magic Key test
- \_\_\_\_\_ FXS ring test
- \_\_\_\_\_ FXS hook off test
- \_\_\_\_\_ FXS hook off power test
- \_\_\_\_\_ VQ test
- \_\_\_\_\_ FXO ring test
- \_\_\_\_\_ FXO(AB/BA) hook on test
- \_\_\_\_\_ FXO(AB/BA) hook off test
- \_\_\_\_\_ FXO(AB/BA) soft hook on test
- \_\_\_\_\_ FXO(AB/BA) soft hook off test



# About Auto-Provision

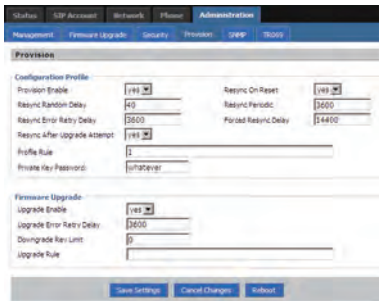
This article briefly introduced the auto-provision about Flying Voice products. To get more information, please refer to **Provision User Manual\_en\_v1.1.doc**. Flying Voice products support secure remote provisioning and firmware upgrades. There are three auto-provision methods, which are TFTP (concluding DHCP Option 66), HTTP and HTTPS with client certificates. All Flying Voice products except FV378 (which is an internet radio) can support the three methods, while the FV378 supports TFTP (concluding DHCP Option 66) and HTTP without HTTPS.

## Macro Expansion

The Flying Voice products support macro expansion; the variables available for macro substitution are **MA, MAU, MAC, PN, SN, IP, SWVER and HWVER**. The macro variables are invoked by prefixing the name with a '\$' character (e.g. \$MAC). The substitution works even within a quoted string, without requiring additional escapes. If the variable name is immediately followed by an alphanumeric character, enclose the variable name in parentheses (e.g. "\$(MAC).conf").

## Webpage

Below is the provision webpage of IP350, the provision webpage of other devices are similar as IP350:

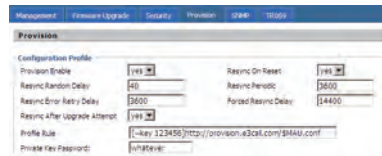


Picture 1 Provision Webpage

The complete set of configuration parameters for a Flying Voice products corresponding to an individual subscriber is referred to as a configuration profile or simply a Profile. The Profile can be a simple plain text file with a list of tag/value pairs or encoded as an XML file.

## Encrypting Configuration Files

Flying Voice products supports encrypted configuration profiles. This can be used for subsequent configuration files stored on or generated by either TFTP or HTTP servers. When used in concert with HTTPS for initial Config, this provides complete security, but only uses the HTTPS server for initial enrollment. Below is an example about download encrypted configuration file via HTTP looks like this:



Picture 2 Encrypting

## DHCP Option 66

Flying Voice products will automatically attempt to resync to a specific file on a TFTP server, whose IP address is offered as one of the DHCP provided parameters. Thus, if desired, a service provider can arrange to connect each new product to a LAN environment configured to pre-provision products. Any new product connected to this LAN would automatically resync to the local TFTP server, initializing its internal state in preparation for deployment. Among other parameters, this pre-provisioning step will configure the URL of the unit's provisioning server. Subsequently, on power-up the product would already know which server to contact for its periodic resync update.

Flying voice products will check option 66 by default, if the products fail to get any information from DHCP option 66; it will go to detect the TFTP/HTTP/HTTPS resync.

## TFTP:

In a small deployment within a single LAN environment, it is often common to rely on a simple TFTP server for provisioning of network devices. Flying Voice devices support TFTP for both provisioning resync and firmware upgrade operations.

Following picture is the provision webpage:

The screenshot shows the 'Provision' configuration page. It is divided into two main sections: 'Configuration Profile' and 'Firmware Upgrade'.  
**Configuration Profile:**  
- Provision Enable: [Yes] (dropdown)  
- Resync On Reset: [Yes] (dropdown)  
- Resync Random Delay: 40 (input field)  
- Resync Periodic: 3600 (input field)  
- Resync Error Retry Delay: 3600 (input field)  
- Periodic Resync Delay: 14400 (input field)  
- Resync After Upgrade Attempt: [Yes] (dropdown)  
- Profile Rule: http://provision.e3call.com/MAU.conf (input field)  
- Private Key Password: cHh4ZGVlcw== (input field)  
**Firmware Upgrade:**  
- Upgrade Enable: [Yes] (dropdown)  
- Upgrade Error Retry Delay: 3600 (input field)  
- Downgrade Rev Limit: 0 (input field)  
- Upgrade Rule: http://provision.e3call.com/P350\_0252\_5127\_S\_110321 (input field)  
At the bottom, there are three buttons: 'Save Settings', 'Cancel Changes', and 'Submit'.

Picture 3 TFTP

## HTTP/HTTPS:

As an alternative to HTTPS, the product can resync to a configuration profile using HTTP. In this case, a separate explicit profile encryption can be used instead of SSL. The product supports 256-bit AES in CBC mode to pre-encrypt individual profiles. These can then be downloaded by the product “in the clear” using HTTP. This resync mode may be useful to reduce the computational load on the provisioning server, with respect to using HTTPS for every resync request.

This screenshot is identical to the one for TFTP, showing the 'Provision' configuration page with the same settings for 'Configuration Profile' and 'Firmware Upgrade'.

Picture 4 HTTP

This screenshot is identical to the one for TFTP, showing the 'Provision' configuration page with the same settings for 'Configuration Profile' and 'Firmware Upgrade'.

Picture 5 HTTPS

## Uploading Certificates

If you want to use HTTPS, you should uploading the certificates which get from you administrator or provider. The following webpage is the place where you can upload certificates.

The screenshot shows the 'Administration' page, specifically the 'Security' section. It has a navigation bar with 'Management', 'Provision', 'Log Job', 'Security', 'Provision', 'SIP', and 'TR069'.  
**Certificate Management:**  
- A table with columns: Issuer ID, Issued By, Expiration.  
- Rows: CA Certificate (none, none, none), User Certificate (none, none, none), Private Key (none).  
**Provision:**  
- A table with columns: Issuer ID, Issued By, Expiration.  
- Rows: CA Certificate (Example Certificate Authority, May 7 02:59:00 2011 GMT), User Certificate (user@example.com, Example Certificate Authority, May 7 02:59:00 2011 GMT), Private Key (none).  
**Certificate Upload:**  
- Update Type: TR069 CA Certificate (dropdown)  
- Local Upload: [Yes] (checkbox)  
- Update button at the bottom.

Picture 6 Administration→Security webpage

# Dial Plan Overview

## 1. Introduction

This overview briefly introduced the dial plan of Flying Voice products. To get more information about dial plan, please refer to the user manual of the product which you get.

The dial plan specifies how to interpret digit sequences dialed by the user, and how to convert those sequences into an outbound dial string. The Flying Voice products allow each line to be configured with a distinct dial plan.

## 2. Webpage

Assuming that you own an Flying Voice product, you should access to the 'Admin mode' and open 'Phone → Dial Plan' webpage to configuration dial plan.

Below picture 1 and picture 2 are the Dial Plan webpage of IP652, the webpage of other Flying Voice products is same as the IP652.





Picture 1 IP652 Phone/Dial Plan webpage



Picture 2 Dial Plan Configuration Page

The dial plan functionality is regulated by the following configurable parameters:

- **Dial Plan Enable:** to enable or disable dial plan
- **No.:** the series number of dial plan.
- **Line:** the line you want to configuration, you can create dial plans to different lines; choose the line which you'd like to modify the dial plan from Line parameter.
- **Digit Map:** the digit map string of dial plan
- **Action:** choose the action from deny or dial out
- **Move Up/Down:** replace the highlighted dial plan, press the  button to move it up, press the  button to move it down

The dial plans contain a series of digit sequences, when a user dials a series of digits; each sequence in the dial plan is tested as a possible match from the first one to the last one. The matching sequences form a set of candidate digit sequences. As more digits are entered by the user, the set of candidates diminishes until only one or none are valid.

Any one of a set of terminating events triggers the Flying Voice products to accept the user-dialed sequence and continue the next step (action: deny the call or dial out the call), or else transmit the input phone number as dialed.

The terminating events are:

- No candidate sequences remain: transmit the input phone number as dialed
- Only one candidate sequence remains, and it has been matched completely: the number is accepted and transmitted the number if the value of the action parameter is dial out, or else the number will be rejected.
- More than one candidates sequence remains, a timeout occurs: the digit sequence is accepted and transmitted as dialed if incomplete, or transformed as per the dial plan if complete.
- An explicit 'send' (user presses the '#' key): the digit sequence is accepted and transmitted as dialed if incomplete, or transformed as per the dial plan if complete.

The timeout duration depends on the matching state.

If a candidate sequence is complete, the **Timer Overrides T** applies.

If no candidate sequences are as yet complete (as dialed), the **Dial Time Out** applies. You can change the value of dial time out in **Phone/Preference** webpage, **miscellaneous** column like below:



Picture 3 Miscellaneous Column in Phone/Preference webpage

### 3. Digit Map Syntax

Each digit sequence within the dial plan consists of a series of elements, which are individually matched to the keys pressed by the user.

**Elements can be one of the following:**

- Individual keys '0', '1', '2' ... '9', '\*', '#'.
- The lower English letter 'x' matches any one numeric digit ('0'.....'9')
- A subset of keys within brackets (allows ranges): '[ \* set ]' (e.g. [389] means '3' or '8' or '9')
- Numeric ranges are allowed within the brackets: **digit '-' digit** (e.g. [2-9] means '2' or '3' or... or '9')
- Ranges can be combined with other keys: e.g. [235-8\*] means '2' or '3' or '5' or '6' or '7' or '8' or '\*'.

#### Element repetition:

Any element can be repeated zero or more times by appending a period ( '.' character) to the element.

For example, "01." matches "0", "01", "011", "0111" ... etc.

#### Subsequence Substitution:

A subsequence of keys (possibly empty) can be automatically replaced with a different subsequence using an angle bracket notation:

'<' **dialed-subsequence** ':' **transmitted-subsequence** '>'.

So, for example, "<8:1650>xxxxxxx" would match "85551212" and transmit "16505551212".

#### Inter-sequence Tones:

An "outside line" dial tone can be generated within a sequence by appending a ',' character between digits. Thus, the sequence "9,1xxxxxxxx" sounds an "outside line" dial tone after the user presses '9', until the '1' is pressed.

For an office environment, the following plan requires a user to dial 8 as a prefix for local calls and 9 as a prefix for long distance. In either case, an "outside line" tone is played after the initial 8 or 9, and neither prefix is transmitted when initiating the call.

<9,:> 1 xxx xxxxxxx

<8,:111> xxxxxxx

#### Timer Overrides T:

The syntax for timer override is: 'T **delay-value** ', the specified delay-value is measured in seconds.

For example, you configuration a dial plan digit sequence '01133xxxxxxxxxT3'; so when this sequence matched completely, it overrides the Dial Time Out and causing an immediate call after 3 seconds.

The timer overrides T applies only the sequence is matched completely and in default the delay-value of timer overrides T is 0.

# IP652 Overview

## 1. Introduction

Flying Voice IP652 which has 5 lines is an advanced business VoIP (Voice over Internet Protocol) phone that provides voice communication over an IP network in the everyday business environment. It offers high voice quality HD handset, HD speaker and HD codec (G.722)

The built-in 20 multi-functional keys for programmed as the IP-PBX features like speed dial, BLF (Busy Lamp Field), shortcut key, Call Pickup, and Call Park, and the five dual-function keys below the LCD will help you to use the phone easily.

The IP652 can support up to 6 expansion modules (EXP40, the expansion modules of Flying Voice, which has 40 multi-functional keys).

Moreover, The IP652 has the rich external interfaces including PoE (Power over Ethernet), headset and expansion module ports for connecting to EXP40.

## 2. Overview

### 2.1. Appearance



Picture 1 IP652



Picture 2 IP652 with two EXP40

### 2.2. Key Feature

Five dual-function keys which can be used as line keys and softkey buttons.

Support 20 multi-functional keys can be used as speed dial button, BLF (Busy Lamp Field), shortcut key (Phonebook, Call History, Text Message, Volume+, Volume- and Login/Logout), Call Pickup, and Call Park.

Support up to 6 expansion modules with 40 multi-functional keys, the function of the multi-functional is the same as the multi-functional keys of IP652.

### 2.3. Interfaces

Following picture 1 displays the interfaces in rear panel.



Picture 3 Rear Panel

### 2.4. Applied Environment

Operation Temperature: 0~50 Degree C  
Storage Temperature: -25~ 85 Degree C  
Relative Humidity: 10%~90% No Condensing  
Shock: Up to 75cm (30 inches) Drop upon Package

### 2.5. Codec and Voice Features

The IP652 supports 4 codec: G.711 (A-Law, U-Law), G.729A/AB, G.723 with 5.3 kbps and 6.3 kbps, and G.722 (HD).

The IP652 supports VAD (Voice Active Detection), CNG (Comfort Noise Generation), AEC, ANC (Automatic Noisy Cancel), AGC (Automatic Gain Control) and adaptive Jitter buffer.

## 2.6. Call Features

5 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, Speed Dial, BLF, Pick up, Call Park, 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

## 2.7. 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

## 2.8. 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  
Support PoE complies with IEEE802.3af (optional)

## 2.9. With Expansion Module (EXP40)

Following picture 4 display the IP652 connect to one EXP40.



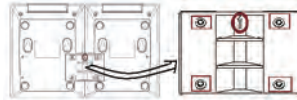
Picture 4 IP652 connect to one EXP40

Following picture 52 displays the IP652 connect to two EXP40s.



Picture 5 IP652 connect to two EXP40s

Following picture 6 displays the IP652 fix to one EXP40 with four screws, the iron is used to indicator up.



Picture 6 IP652 fix to one EXP40

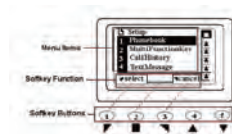
## 2.10. LCD Screen

The IP652 has a special 128\*64 pixel customized LCD. The LCD transmits the main information to customer, such as date, time, Internet IP address, line status and so on. Below picture shows the LCD which is in standby:



Picture 7 LCD standby

Below picture shows the LCD which is in menu configuration:



Picture 8 LCD Menu

## 2.11. Web Configuration

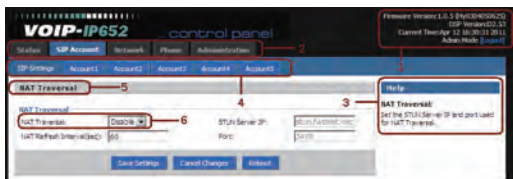
The IP652 supports web configuration with MD5 authentication, and two level configurations which are admin mode and user mode. In admin mode user can configure all parameters, while in user mode user can only configure partly parameters with partly can't configuration such as VoIP settings.

Below picture is the login web: user enters the user name and password to login and start to configuration



Picture 9 Login webpage

Below picture is the IP652 web:



Picture 10 Configuration webpage

### Explanation:

Place 1: Display the main information, such as firmware version, DSP version, Current Time, and configuration mode. Press Logout button to logout.

Place 2: Click navigation bar, and then many sub-navigation bars will appear in the place 4.

Place 3: Display the briefly explanation of the corresponding web; user can get help from it directly.

Place 4: Choose sub-navigation bars to open one configuration webpage.

Place 5: The configuration column.

Place 6: Parameters and settings value.

# TR069 overview

## 1. Background

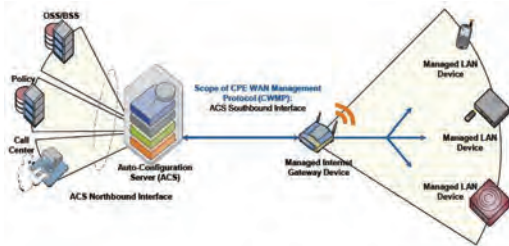
In the course of the boom of the broadband market, the number of different Internet access possibilities grew as well (e.g. modems, routers, gateways, set-top box, VoIP-phones). At the same time the configuration of this equipment became more complicated -- too complicated for the end-users. For these reasons the TR-069 standard was developed. It provides the possibility of auto configuration of these access types and reduces the cost of management.

## 2. What is TR-069

TR-069 (short for Technical Report 069) is a DSL Forum technical specification entitled CPE WAN Management Protocol (CWMP). It defines an application layer protocol for remote management of end-user devices.

Using TR-069, the terminals can get in contact with the Auto Configuration Servers (ACS) and establish the configuration automatically. Accordingly other service functions can be provided.

## 3. The Network Topological of TR-069



## 4. Functions supported by TR-069

- Auto configuration and dynamic service activation
  - initial CPE configuration
  - remote CPE configuration
- Firmware management
  - version management
  - update management

- Status and Performance Monitoring
  - control logfile analysis and dynamic messages
  - connectivity and service control
- Diagnostics

## 5. TR-069 Extensions

- TR-098-data model for internet gateway devices (DSL modems with built in routers)
- TR-104-data model and any specific items for VoIP devices
- TR-106-a base object structure for TR-069 enabled devices
- TR-110-a reference model for VoIP configurations
- TR-111-covers applying TR-069 to remote management of home network devices

## 6 About Products

Now all the products of Flying Voice can support standard TR-069 protocols, and user can configuration the parameters of TR069 in Administration/TR069 webpage and upload security file in Administration/Security webpage. The TR069 configuration page is:

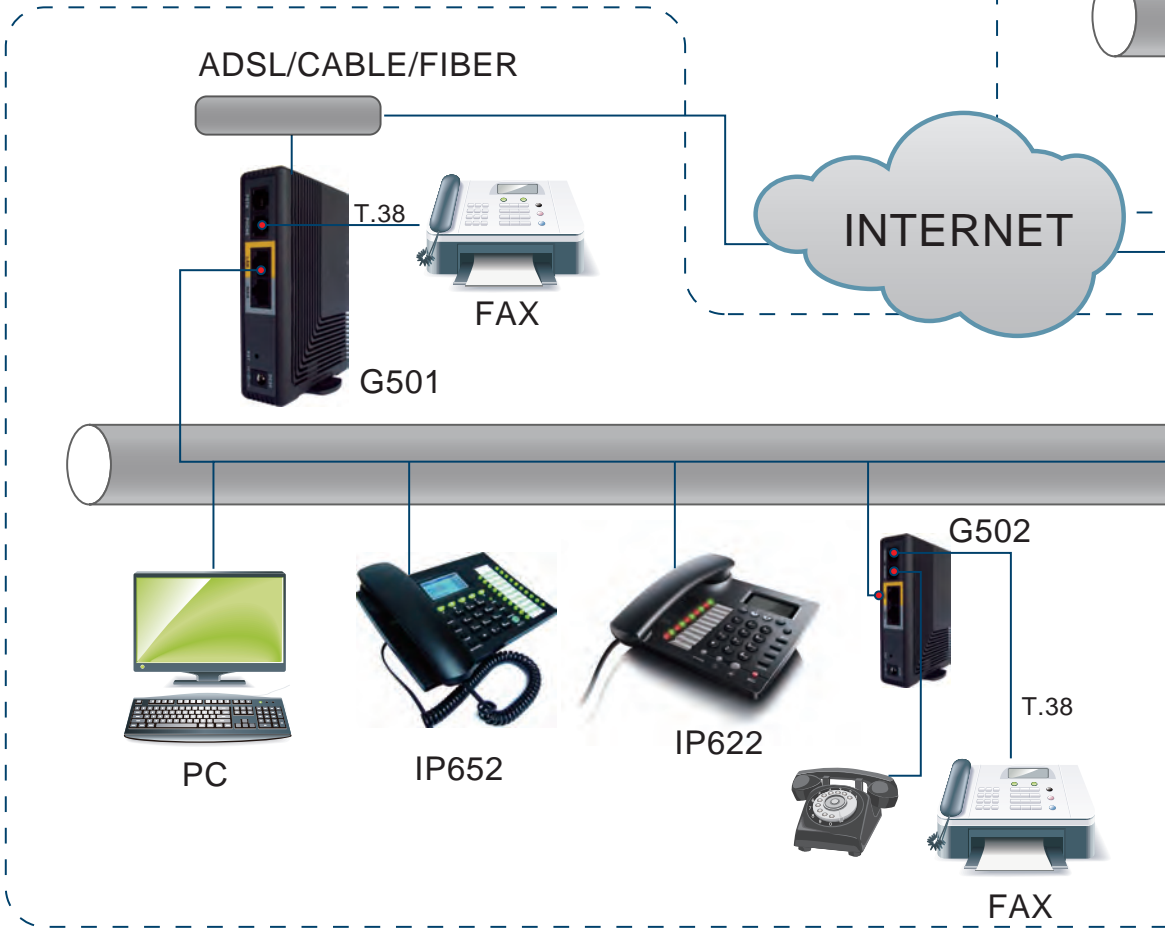
ACS	
TR069 Enable:	Disable
CWMP:	Enable
ACS URL:	
User Name:	
Password:	
Periodic Inform Enable:	Enable
Periodic Inform Interval:	30
Connect Request:	
User Name:	
Password:	
SSL Key:	

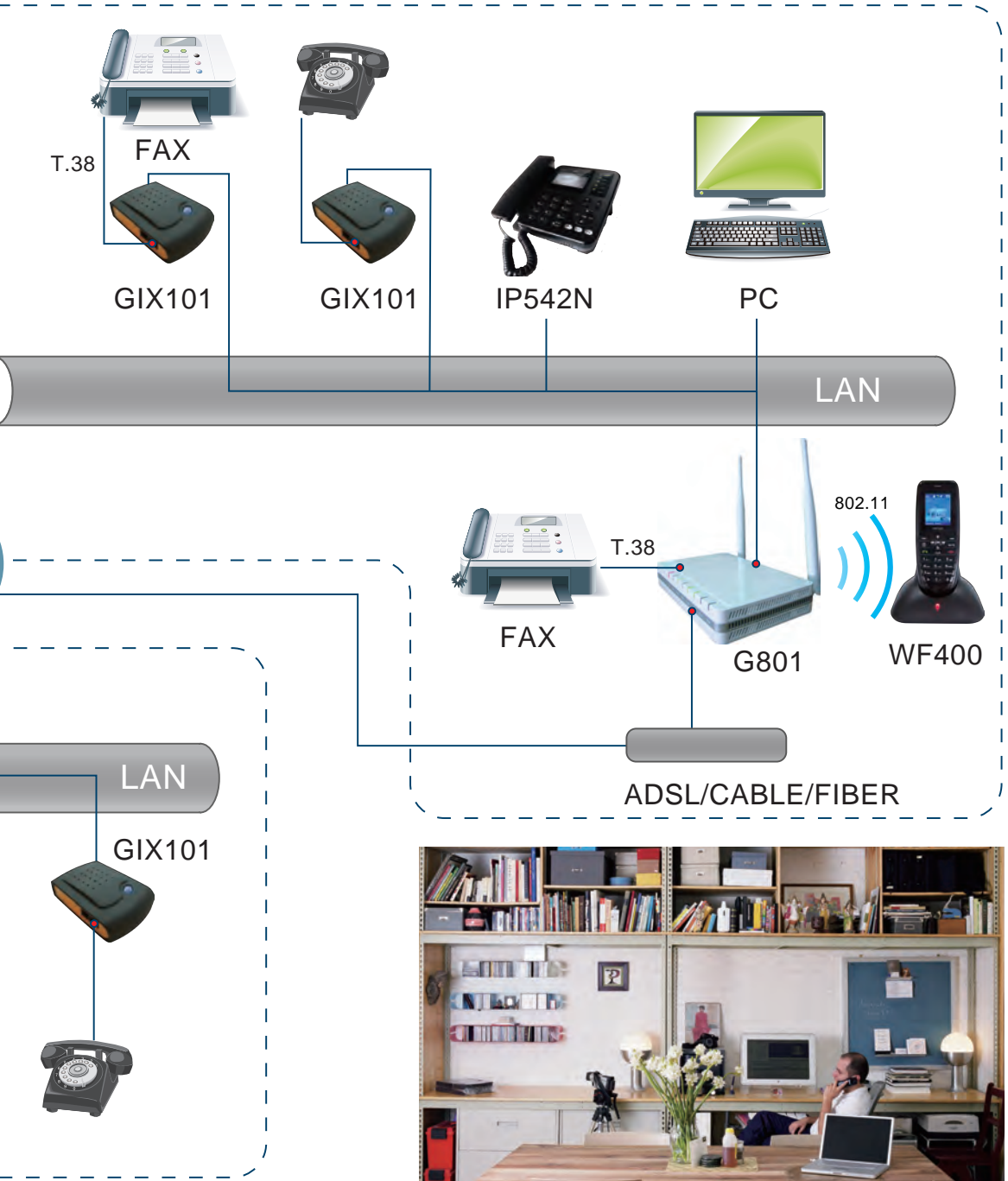
The Security page is

id	cert	cert	cert
2	none	none	none
3	none	none	none
4	none	none	none

id	cert	cert	cert
2	Example Certificate Authority	Example Certificate Authority	May 7 01:59:00 2011 GMT
3	user@example.com	Example Certificate Authority	May 7 03:59:00 2011 GMT
4		Here (label)	

Certificate Update  
Update Type: TR069 CA Certificate  
Local Upload:







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