

MOCET

IP3032 Standard IP Phone Administrators' Guide



FCC Statement

This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions in this guide, may cause interference to radio communications. This equipment has been tested and found to comply with the limits for a Class B computing device pursuant to Subpart J of Part 15 of FCC rules, which are designed to provide reasonable protection against radio interference when operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference, in which case the user, at its own expense, will be required to take whatever measures are necessary to correct the interference.

CE Declaration of Conformity

This equipment complies with the requirements relating to electromagnetic compatibility, EN55022 class B for ITE and EN 50082-1. This meets the essential protection requirements of the European Council Directive 89/336/EEC on the approximation of the laws of the Member States relating to electromagnetic compatibility.

Environment

The phone you have purchased, as well as any used batteries must not be disposed of with household waste. You should return these to your distributor if they are to be replaced or disposed of them in an approved recycling center.

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WARNING!



1. Read these installation instructions carefully before connecting the IP phone to its power.
2. To reduce the risk of electric shock, do not remove the cover from the IP phone or attempt to dismantle it. Opening or removing covers may expose you to dangerous voltage levels. Equally, incorrect reassembly could cause electric shock on re-use of the appliance.
3. Do not expose the IP Phone to Fire, direct sunlight or excessive heat.
4. Do not expose the IP Phone to rain or moisture and do not allow it to come into contact with water.
5. Do not install the IP phone in an environment likely to present a THREAT OF IMPACT.
6. You may clean the IP phone using a fine damp cloth. Never use solvents (such as trichloroethylene or acetone), which may damage the phone's plastic surface and LCD screen. Never spray the phone with any cleaning product whatsoever.
7. Take care not to scratch the LCD screen.
8. The IP phone is designed to work in temperatures from 5°C to 40°C.
9. The IP phone must be installed at least 1 meter from radio frequency equipment, such as TVs, radios, hi-fi or video equipment (which radiate electromagnetic fields).
10. Do not connect the LAN port to any network other than an Ethernet network.
11. Do not attempt to upgrade your IP phone in an unstable power environment. This could cause unexpected issues.
12. Do not work on the system or connect or disconnect cables during lightning storms.
13. Children don't recognize the risks of electrical appliances. Therefore use or keep the phone only under supervision of adults or out of the reach from children.
14. No repair can be performed by the end user, if you experience trouble with this equipment, for repair or warranty information, please contact your supplier.

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Introduction

About This Guide

The IP3032 Standard IP Phone Administrators' Guide provides instructions for installing, provisioning, and administering IP3032 phones. This guide will help you understand the MOCET VoIP network and telephony components, and provides descriptions of all available phone features.

Part 1: Getting Started of this guide gives you an overview of the IP3032 IP Phone.

Part 2: Setting Up Your System provides you essential information on how to setup your phone network and a provisioning server.

Part 3: Configuring Your System is devoted to descriptions of the phone features you can configure on the phones, which include brief examples of feature configurations.

Part 4: System Maintenance Tasks provides you firmware upgrade methods as well as phone maintenance tasks.

Part 5: References show IP3032 firmware menu structure.

This guide will help you perform the following tasks:

- Install and configure your phone on a network server
- Configure your phone's features and functions
- Configure your phone's user settings

This guide describes a method for provisioning IP3032 phones. Although there are other methods, the method described in this guide provides the most flexibility and manageability, and is the recommended approach for enterprise installations.

Who Should Read This Guide?

System administrators and network engineers should read this guide to learn how properly to set up IP3032 phones. This guide describes administration-level tasks and is not intended for end users.

How This Guide is Organized

This guide is organized into five parts. Each part contains multiple chapters. The parts are sequenced in the same way you would deploy IP3032 phones.

The parts contain the following chapters:

Part 1: Getting Started

Chapter 1, “Welcome to the MOCET IP3032 Standard IP Phone”, introduces the MOCET IP3032 Standard IP Phone.

Chapter 2, “IP3032 Standard IP Phone Firmware Architecture”, shows you how IP3032 IP Phones fit in your organization and details about IP3032 IP Phone firmware architecture.

Part 2: Setting Up Your System

Chapter 3, “Setting Up Your Phone Network”, describes how to set up your network.

Chapter 4, “Setting Up the Provisioning Server”, provides basic and advanced instructions on how to set up a provisioning server, deploy the IP3032 phones from the provisioning server, and upgrade the phone’s firmware.

Part 3: Configuring Your System

Chapter 5, “Setting Up Advanced Phone Features”, shows you how to configure and use advanced phone features like corporate directory and voice mail.

Chapter 6, “Setting Up Audio Features”, provides information on configuring and using audio features.

Chapter 7, “Setting Up User and Phone Security Features”, describes how to configure and use security features like locking the phone.

Part 4: System Maintenance Tasks

Chapter 8, “Upgrading Your IP3032 Phones Firmware”, give information about firmware upgrade methods.

Chapter 9, “Miscellaneous Maintenance Tasks”, shows you how to maintain the IP3032 phones, that includes configuration file backup and updates, dial plan, system log and reset to default.

Part 5: References

Chapter 10, “IP3032 Firmware Menu System”, shows the menu structure of the IP3032 IP Phone firmware.

Getting Help and Support

If you are looking for help or technical support for your phones, the following types of documents are available:

- Quick User Guide, which describes how to assemble IP3032 phone and the basic phone features
- User Guide, which describes both basic and advanced phone features
- Release Note, which describes the new and changed features and fixed problems in the latest version of the firmware

Part 1: Getting Started

Part 1 gives you an overview of the MOCET IP3032 Standard IP Phone and consists of the following chapters:

- Chapter 1: Welcome to the MOCET IP3032 Standard IP Phone
- Chapter 2: IP3032 Standard IP Phone Firmware Architecture

Chapter 1: Welcome to the MOCET IP3032 Standard IP Phone

This chapter introduces the MOCET IP3032 Standard IP Phone.

The MOCET IP3032 Standard SIP phone is an easy-to-use high quality desk phone with many advanced features including support for secure calling with trusted layer security (TLS) and Secure Real-time Transfer Protocol (SRTP), a built-in IP Security (IPSEC) virtual private network (VPN) client, and instant messaging capabilities. Utilizing a next generation capacitive touch sensitivity panel design, the IP3032 supports up to four simultaneous lines and can, multiple tilt angles and wall mount options as well as automatic support for power over Ethernet (PoE). The IP3032 can be configured through the simple built in menus displayed on the blue backlit LCD or from the phone's Web Configuration Interface. The IP3032 can be automatically provisioned from a local or Internet based server using the built-in MOCET auto-provisioning and management protocols.

The IP3032 IP Phone supports many advanced features including 3-way on-phone conferencing, can transfer and receive calls using industry-standard SIP protocols, and can provide built-in music-on-hold (MoH) over IP network. The IP3032 is interoperable with a wide range of SIP services and servers including those based on Metaswitch™, Broadsoft™, Freeswitch™ and Asterisk™. Therefore, the IP3032 can be deployed and used anywhere there is a suitable local area network (LAN) with Internet access and a local or remotely hosted SIP server. Since it is a stand-alone and “always-on” device, it does not require connection to a computer for it to work.

Key Features of IP3032 IP Phone

Features	Description
LCD	128 x 32 pixels 3 lines x 21 characters
Physical line keys	2 keys with the extra Line Group Switch button
Multiple line appearances	Up to 4 line appearances
Multiple call appearances	Up to 8 call appearances
Conference	Up to 3-party
Navigation Key (Up, Down, Left, Right and OK)	Yes (capacitive touch)
Programmable keys	8 keys
Audio quality (Speaker, Handset Receiver)	Narrow Band

Voice Codec	G.711 (A-law/Mu-law) G.723.1 G.726-32 G.729A/B iLBC
Ethernet	10M/100M (1 port)
Power Supply	Built-in IEEE802.3af PoE port Local Power (DC5V)
AC adaptor	5V/ 550mA
Tilt stand	Yes (3 steps. 30°, 51° and 60°)
Wall mount	Yes (Optional)

Chapter 2: IP3032 Standard IP Phone Firmware Architecture

This chapter provides an overview of the IP3032 Standard IP Phone firmware big picture, specifically an understanding of how the phone fits into the network configuration. If you want to begin setting up your IP3032 phones, go to [Setting Up Your Phone Network](#).

The IP3032 phone supports the following deployment scenarios.

The Session Initiation Protocol (SIP) is the Internet Engineering Task Force (IETF) standard for multimedia communications over IP. It is an ASCII-based, application-layer control protocol (defined in RFC3261) that can be used to establish, maintain, and terminate calls between two or more endpoints. Like other voice over IP (VoIP) protocols, SIP is designed to address the functions of signaling and session management within a packet telephony network. Signaling allows call information to be carried across network boundaries. Session management provides the ability to control the attributes of an end-to-end call.

For IP3032 phones to successfully operate as a SIP endpoint in your network, it will require:

- A working IP network
- Routers configured for VoIP
- VoIP gateways configured for SIP
- An active, configured call server to receive and send SIP messages

The rest of this chapter consists of the following sections:

- [Where IP3032 Phones Fit in Your Network](#)
- [Understanding IP3032 Phone Firmware Architecture](#)
- [Features Available on Your IP3032 Phones](#)

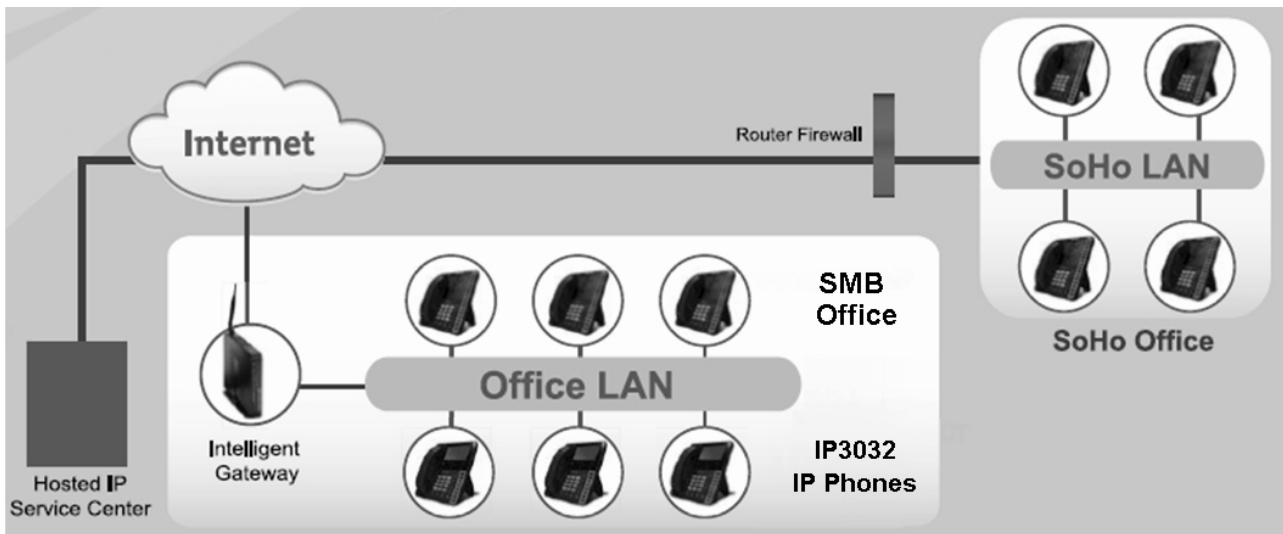
If you want to begin setting up your IP3032 phones on the network, go to [Setting Up Your Phone Network](#).

If you want to begin configuring the features available for your IP3032 phones, go to [Part 3: Configuring Your System](#).

Where IP3032 Phones Fit in Your Network

IP3032 phones connect physically through a Category 5 (Cat-5) cable to a standard office twisted-pair (IEEE 802.3) 10/100 megabits per second Ethernet LAN, and send and receive all data using the same packet-based technology.

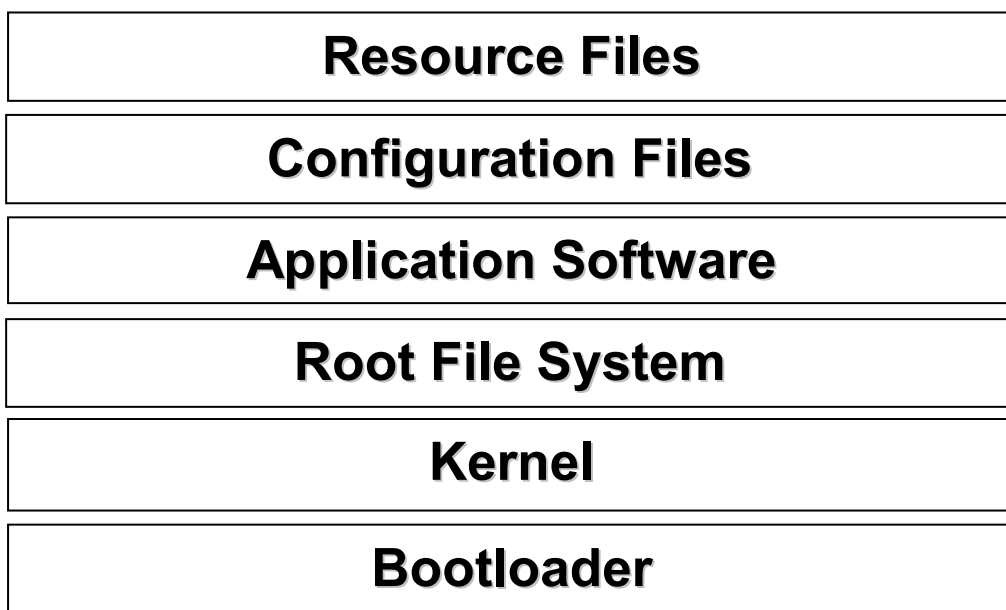
There are many ways to set up a phone network using IP3032 phones. The figure below is just one example of a network setup.



Understanding IP3032 Phone Firmware Architecture

The IP3032 phone firmware is made of six basic components:

- Bootloader – The firmware that loads first when the phone is powered on
- Kernel
- Root File System
- Application Software – The software that implements the phone functions and features
- Configuration files – Files that contain the phone's settings
- Resource files – Optional files that contain setting for advanced features



For each release of firmware, there is one combined image and one update pack.

- A combined image contains bootloader, kernel, root file system, application software and configuration files.
- An update pack is used when updating some, but not all, in a combined image; we also call it a “hot fix”; a “hot fix” is used when possible to reduce upgrade time and network bandwidth requirements.

What are the Configuration Files?

The configuration files are valid XML files that you can modify using an XML editor.

You can configure IP3032 phones automatically through configuration files stored on a central provisioning server. Or, you can manually configure a phone using the phone’s menu system via the local user interface. If you want, you can use the Web Configuration Interface, or use a combination of the automatic and manual methods.

We recommend that you configure phones automatically through a central provisioning server. If a provisioning server is not available, you can use one of the manual methods to update most phone settings.

What are the Resource Files?

Examples of resource files include:

- Contacts Directories
- Ringtones

Features Available on IP3032 Phones

This section briefly outlines the features available on IP3032 phones.

Basic User Features

- **Configuring Call Forwarding** – Provides a flexible call forwarding feature to forward calls to another destination.
- **Enabling Call Hold** – Pauses activity on one call so that you can use the phone for another task, such as making or receiving another call.
- **Configuring the Call Logs** – Contains call information such as remote party identification, time and date, and call duration in three separate lists, dialed calls, missed calls, and answered calls.
- **Configuring Call Park and Retrieve** – Park an active call – puts it on hold to a specific location, so it can be retrieved by any phone. This feature requires call server support.
- **Understanding the Call Timer** – Maintains a timer, in hours, minutes, and seconds, for each call in progress.

- Using Call Transfer – Transfer a call in progress to some other destinations.
- Configuring Call Waiting Alerts – Visually presents an incoming call on the screen, and plays a configurable sound effect, when you are in another call.
- Called Party Identification – Displays and logs the identity of the party in an outgoing call.
- Enabling Conference Management – Add, hold, mute, and remove conference participants, and obtain information about participants.
- Configuring Calling Party Identification – Displays a caller's identity, derived from the network signaling, when an incoming call is presented – if the information is provided by the call server.
- Connected Party Identification – Displays and logs the identity of the party to whom you are connected to (if the name is provided by the call server).
- Applying Distinctive Ringing – Enables you to select a ring tone for contacts in the contact directory.
- Configuring Do Not Disturb – Temporarily stops all incoming calls.
- Configuring the Handset and Speakerphone.
- Creating Local and Centralized Conferences – Join calls to create local conferences. The user can call into centralized conferences using conference bridge numbers. This feature requires call server support.
- Using the Local Contact Directory – The phone maintains a local private contact directory that can be edited locally, and a local public contact directory that can be downloaded from a dedicated web server.
- Microphone MUTE – Mutes the phone's microphone so other parties cannot hear you.
- Enabling Missed Call Notification – Displays the number of missed calls you have since you last looked at the Missed Calls list.
- Using the Speed Dial Feature – Enables you to place calls quickly from dedicated keys as well as from a speed dial menu.
- Setting the Time and Date Display – Time and date can be displayed in different format.

Advanced Features

- Enabling Shared Line Appearance – Allows a line extension or phone number to appear on multiple user's phones. This feature requires call server support.

- Using Busy Lamp Field – You can monitor the hook status of remote parties with the busy lamp field (BLF) LEDs and you can display your status on an attendant phone. This feature requires call server support.
- Resetting to Factory Defaults – Enables users to reset the phone to the factory default settings.
- Using the Corporate Directory – You can configure the phone to access your corporate directory if it has a standard LDAP interface.
- Using Instant Messaging – Supports the sending and receiving of instant text messages.
- Using Multiple Call Appearances – Supports multiple concurrent calls. You can place any active call on hold to switch to another call.
- Assigning Multiple Line Keys Per Registration – You can assign Multiple Line Keys to a single registration.
- Enabling Multiple Registrations – Supports multiple registrations per phone.
- Configuring Network Address Interface Translation – Phones can work with certain types of network address translation (NAT).
- Quick Setup – Provides a simplified interface to enter provisioning server parameters while your phone boots.
- Configuring Voicemail – Configures to access a compatible voice mail server.

Audio Features

- Acoustic Echo Cancellation – Employs advanced acoustic echo cancellation for handsfree operation.
- Audible Ringer Location – Choose how to play out audio tones.
- Audio Codecs – Enables access to a wide range of industry standard audio codecs.
- Volume Control – Choose the volume levels for the various audio outputs on the phone.
- Comfort Noise – Provides a consistent noise level to the remote user.
- Customizing Audio Sound Effects – Enables you to customize sound effects associated with incoming calls and other events.
- DTMF Event RTP Payload – Conforms to RFC2833, which describes a standard RTP-compatible technique for conveying DTMF dialing and other telephony events over an RTP media stream.

- Generating Dual Tone Multi-Frequency (DTMF) Tones – Generates dual tone multi-frequency (DTMF) tones in response to user dialing on the dial pad.
- IEEE 802.1p/Q – The phone may tag all Ethernet packets it transmits with an 802.1Q VLAN header.
- Voice Activity Detection – Conserves network bandwidth by detecting periods of relative “silence” in the transmit data path and replacing that silence with special packets that indicate silence is occurring.

Security Features

- Configuration File Encryption – Confidential information stored in configuration files can be protected (encrypted). The phone can recognize encrypted files, which it downloads from the provisioning server.
- Digital Certificates – Supports digital certificates and associated private keys.
- Local User and Administrator Passwords – Local settings menus are protected with two privilege levels – user and administrator – each with its own password.
- Locking the Phone – Prevent access to the phone menu and to key presses.
- Secure Real-Time Transport Protocol – Encrypting audio streams to avoid interception and eavesdropping.
- 802.1X Authentication – Authenticate devices connecting to a local area network (LAN).

For instructions on how to set up each feature on IP3032 phone, see the feature sections in Part 3: Configuring Your System.

Part 2: Setting Up Your System

Part 2 provides you with essential information on how to set up your phone network and a provisioning server. You will find basic and advanced instructions on how to set up a provisioning server, how to deploy IP3032 phones from the provisioning server, and how to upgrade the firmware.

Part 2 consists of the following chapters:

- Chapter 3: Setting Up Your Phone Network
- Chapter 4: Setting Up the Provisioning Server

Chapter 3: Setting Up Your Phone Network

This chapter shows you several automated and manual ways to configure IP3032 phones to operate in a LAN.

Connecting your IP3032 phone to the LAN will initiate a start-up sequence. Note that only step 1 is required and automatic. Steps 2, 3, and 4 are optional as all these settings can be manually configured on the phone. It is common to complete step 3 using a DHCP server within the LAN. The phone uses the following start-up sequence:

1. The phone establishes network connectivity.
IP3032 phone will establish 10M/100M network link with an Ethernet switch device. The phone will not be able to make and receive calls until this link is established. If IP3032 phone can not establish a link to the LAN, an error message *Check Network Connection* will display on the LCD.
2. Apply appropriate security and Quality of Service (QoS) settings (optional).
Assign the phone to a VLAN and/or 802.1X authentication.
3. Establish DHCP negotiation with the network and IP address, network addressing options, network gateway address, and time server.
4. Provisioning server discovery. (optional)

Once the provisioning server discovery is complete, the phone will initiate the provisioning process, which is described in the next chapter *Setting Up the Provisioning Server*.

These steps are described in more detail in the following sections of this chapter:

- Establishing Link Connectivity
- Security and Quality of Service Settings
- IP Communication Settings
- Provisioning Server Discovery
- Phone Network Menus

Establishing Link Connectivity

IP3032 phone supports the following Ethernet line rates: 10Mbps and 100Mbps. Ethernet line rates are automatically negotiated so that no special configuration is required.

Security and Quality of Service Settings

You have the option of using several layer-2 and layer-3 mechanisms that increase network security and minimize audio latency. This section describes each of the network security and quality of service options.

VLANs

A Virtual LAN (VLAN) can be used to separate and assign higher priority to a voice VLAN as a way of minimizing latency. IP3032 supports **Static** method. The VLAN ID can be manually set from the phone web or menu interface or from a configuration file. To change this parameter, go to VLAN Menu.

802.1X Authentication

802.1X authentication is a technology that originated for authenticating Wi-Fi clients. It has also been adopted for authenticating PCs and other devices in LAN deployments. To change this parameter, go to 802.1X Menu.

QoS with DSCP

Differentiated Services Code Point (DSCP) is a field in an IP packet that enables different levels of service to be assigned to network traffic. This is achieved by marking each packet on the network with a DSCP code and appropriating it to the corresponding level of service. To change this parameter, go to QoS Settings Menu.

IPSec

Internet Protocol Security (IPSec) is a protocol suite for securing Internet Protocol (IP) communications by authenticating and encrypting each IP packet of a communication session. It can be used in protecting data flows between a pair of hosts (host-to-host), between a pair of security gateways (network-to-network), or between a security gateway and a host (network-to-host). IP3032 supports IPSec in a host-to-host transport and in network tunnel modes. To change this parameter, go to IPSec Tunnel Items Menu.

IP Communication Settings

When the phone has established network connectivity, it needs to acquire several IP network settings to proceed with IP communication. These settings are typically obtained automatically from a DHCP server.

You have the option to set the IP communication settings manually from the phone web or menu interface.

Provisioning Server Discovery

By default, the IP3032 Auto-Provisioning function is disabled. If you need to use IP3032 Auto-Provisioning function, please go to Chapter 4: Setting Up the Provisioning Server.

Phone Network Menus

You have the option of modifying the phone network configuration.

After your phone starts and enters standby mode, the Network Setting menu is accessible from the phone's main menu.

Select **Menu > Admin Setting > Network Setting**.

To access the **Admin Setting** menu, you will have to enter the administrator's password. For more information, see Local User and Administrator Passwords in Chapter 7: Setting Up User and Phone Security Features.

Use the soft keys, the arrow keys, and the OK key to make changes.



Some advanced network configuration items, for example, QoS Settings, IPSec Tunnel Items and Security and Certificates Items, are only accessible through Web Configuration Interface. So, if possible, using Web Configuration Interface to do Network Setting and Security and Certificates Setting configurations is more convenient.

Admin Setting Menu

<i>Name</i>	<i>Description</i>
Network Type	See Network Type Menu.
IP Version Type	See IP Version Type Menu.
Static IP	See Static IP Menu.
802.1X	See 802.1X Menu.
SNTP	The Simple Network Time Protocol (SNTP) server is the phone obtains the current time from.
VLAN	See VLAN Menu.
Ping	The Ping function is for administrator to debug when the phone can not connect to LAN.

Network Type Menu

<i>Name</i>	<i>Description</i>
Static IP	If enabled, the phone will use the parameters configured in Static IP menu.
DHCP	The default setting. DHCP is used to obtain the parameters of IP Address, Subnet Mask, Default Gateway and DNS.

IP Version Type Menu

<i>Name</i>	<i>Description</i>
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IPv4	The default setting. When the phone is enabled Static IP, it will use the configured IPv4 address.
IPv6	If enabled as well as Static IP, the phone will use the configured IPv6 address.

Static IP Menu

<i>Name</i>	<i>Description</i>
IP Address	When the phone is enabled Static IP and IPv4, it will use the configured IPv4 address.
IPv6 Address	When the phone is enabled Static IP and IPv6, it will use the configured IPv6 address.
Subnet Mask	When the phone is enabled Static IP, it will use the configured subnet mask.
Default Gateway	When the phone is enabled Static IP, it will use the configured default gateway.
DNS	When the phone is enabled Static IP, it will use the configured DNS.

802.1X Menu

<i>Name</i>	<i>Description</i>
802.1X Type	Three options: Off, EAP-MD5 and EAP-TLS. The default is Off. The selected EAP type is used for authentication.
802.1X Identity	The identity is required for 802.1X authentication.
802.1X Password	The password is required for 802.1X authentication.

VLAN Menu

<i>Name</i>	<i>Description</i>
VLAN Mode	Two options: None and Static. The default is None. If Static is enabled, the phone will use the configured Voice VLAN ID and VLAN Priority in the TCP/IP stack, so that every Ethernet frame sent to network will contains VLAN tag.
Voice VLAN ID	The phone's 802.1Q VLAN identifier. The default value is 2. The range is from 1 to 4096.
VLAN Priority	The phone's 802.1Q VLAN priority. The default value is 0.

QoS Settings Menu

The QoS Settings menu is accessible from Web Configuration Interface only.

Name	Description
DSCP for RTP	Fourteen options: Best Effort, AF Class 1(Low Drop), AF Class 1(Medium Drop), AF Class 1(High Drop), AF Class 2(Low Drop), AF Class 2(Medium Drop), AF Class 2(High Drop), AF Class 3(Low Drop), AF Class 3(Medium Drop), AF Class 3(High Drop), AF Class 4(Low Drop), AF Class 4(Medium Drop), AF Class 4(High Drop), Expedited Forwarding. The default is Best Effort.
DSCP for SIP	

IPSec Tunnel Items Menu

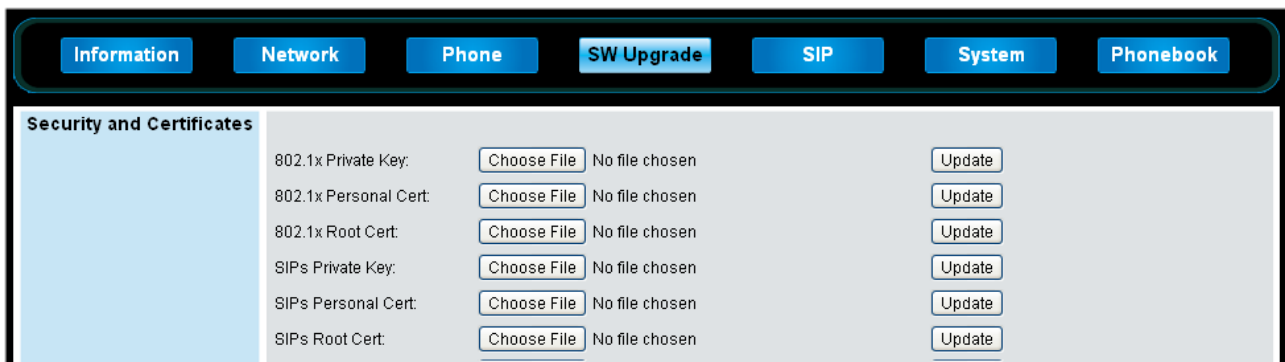
The IPSec Tunnel Items menu is accessible from Web Configuration Interface only.

Name	Description
Enable IPSec	The default is disabled. When enabled, the phone will use the configured parameters in IPSec Tunnel Items to create virtual private networks (VPN) for communications.
VPN Gateway	The default IP address of VPN Gateway and Remote Internal IP Selector is 192.168.0.1. In transport mode, you can set “VPN Gateway and Remote Internal IP Selector” as the same with remote phone’s IP address.
Remote Internal IP Selector	In tunnel mode, you need to set the correct VPN Gateway, so phone can setup VPN tunnel with this gateway, and communicate with Remote Internal IP Selector directly and secretly.

Preshared Key	The default is null.
IKE Encrypt Method	Three options: DES, 3DES, AES. The default is DES.
IKE Integrity Method	Two options: SHA, MD5. The default is SHA.
ESP Encrypt Method	Three options: DES, 3DES, AES. The default is DES.
ESP Integrity Method	Two options: SHA, MD5. The default is SHA.
Enable PFS	The default is enabled.
Phase 1 Lifetime (sec)	The default value is 28800.
Phase 2 Lifetime (sec)	The default value is 1800.

Security and Certificates Items Menu

The Security and Certificates items menu is accessible from Web Configuration Interface only.



<i>Name</i>	<i>Description</i>
802.1x Private Key	Select desired 802.1x Private Key (.pem) and press Update button to upgrade.
802.1x Personal Cert	Select desired 802.1x Client Cert (.pem) and press Update button to upgrade.
802.1x Root Cert	Select desired 802.1x Root Cert (.pem) and press Update button to upgrade.
SIPs Private Key	Select desired SIPs Private Key (.pem) and press Update button to upgrade.
SIPs Personal Cert	Select desired SIPs Client Cert (.pem) and press Update button to upgrade.
SIPs Root Cert	Select desired SIPs Root Cert (.pem) and press Update button to upgrade.

Chapter 4: Setting Up the Provisioning Server

This chapter provides basic instructions for setting up your IP3032 phones with a provisioning server.

This chapter consists of the following sections:

- Why Use a Provisioning Server
- Preparation for Auto-Provisioning Service
- Provisioning Procedure

Why Use a Provisioning Server?

A provisioning server allows for flexibility in installing, upgrading, maintaining, and configuring the IP3032 phones. The provisioning server can be set up on the local LAN or anywhere on the Internet. The IP3032 phone is designed such that if it can not locate a provisioning server when it boots up, it will operate with internally saved parameters. This is useful when the provisioning server is not available.

Preparation for Auto-Provisioning Service

To use Auto-Provisioning Service, you need to make sure provisioning files, provisioning server and IP3032 Auto-Provisioning Service setting are configured ready. See below sections to know how to prepare each of them.

Provisioning Files

IP3032 Auto-Provisioning function supports configuration file update and firmware upgrade. The provisioning configuration files are in XML format.

Only those settings to be provisioned are required to be included in the XML file. All settings in the XML file will be applied and treated as changes even the values are the same as the current settings on the phone.

Auto-Provisioning Service Settings

The Auto-Provisioning Service function uses FTP/ TFTP/ HTTP/ HTTPS as transport protocol. As to HTTP/ HTTPS, it will support Basic and Digest authentication.

There are eight options for auto-provisioning setting through the Web Configuration Interface.

Information	Network	Phone	SW Upgrade	SIP	System	Phonebook
Auto Provisioning Settings						
Enable APS:	<input type="checkbox"/>		APS on Boot:	<input checked="" type="checkbox"/>		
APS Server Type:	TFTP		Files Directory:	<input type="text"/>		
APS Server:	192.168.1.100		APS Server User Password:	<input type="password"/>		
APS Server User Name:	root					
APS Interval time(Sec.):	1440					

Name	Description
Enable APS	Enable or disable APS function.
APS Server Type	Select APS server type: HTTPS/ HTTP/ TFTP/ FTP.
APS on Boot	Do APS or not when phone on boot.
APS Server	The server address can be an IP address or FQDN, or together with port.
File Directory	The file directory of APS server.
APS Server User Name	The login name of APS server.
APS Server User Password	The login password of APS server.
APS Interval Time	The interval time (minutes) to retry APS.

The Hierarchy of File System in Provisioning Server

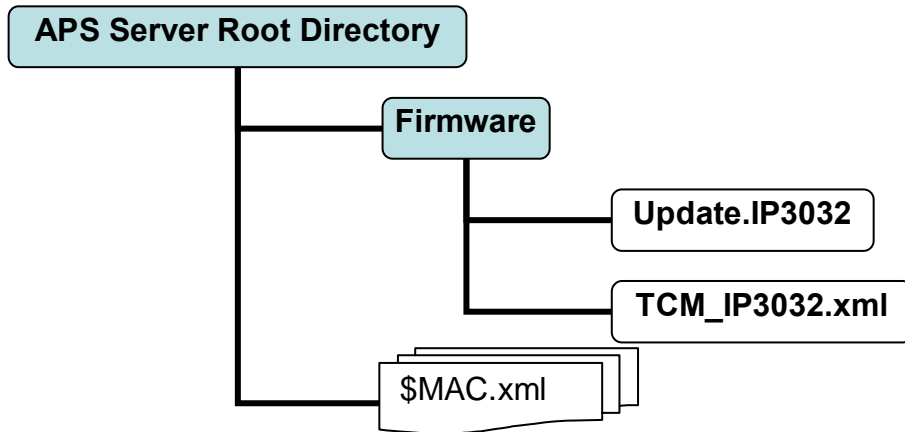
Following part will only describe how to provision via HTTP.

The configuration file name for IP3032 is TCM_IP3032.xml. It is an XML document that indicates the latest version number and the latest firmware's URL of IP3032.

For example, the following content indicates its newest version is F_5_3_1.0.30, and its firmware URL is http://172.16.11.254:5000/provisioning/firmware/update.ip3032.

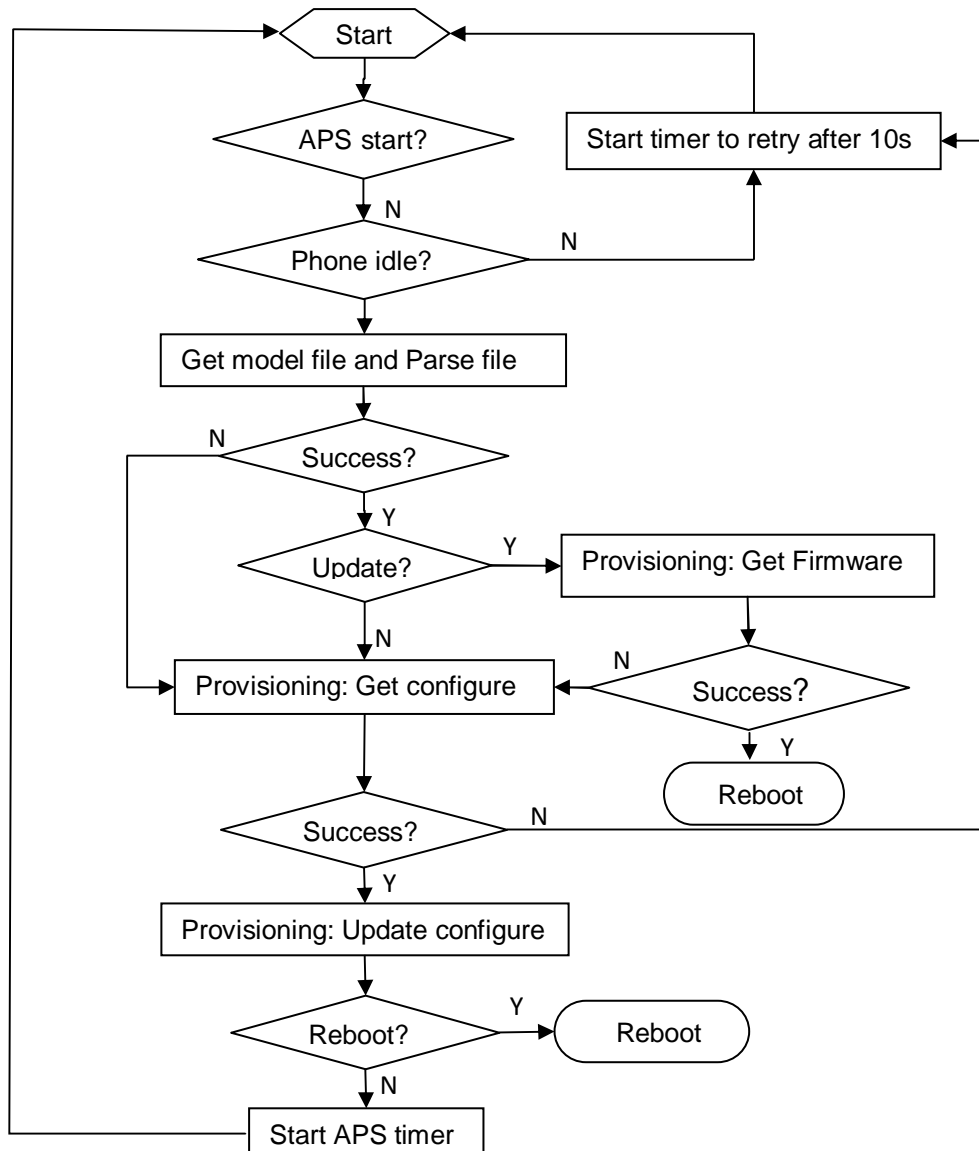
```
<?xml version="1.0" encoding="utf-8" standalone="yes"?>
<Firmware>
<Firmware_Version>F_5_3_1.0.30</Firmware_Version>
<Firmware_URL>http://172.16.11.254:5000/provisioning/firmware/update.ip3032</Firmware_URL>
</Firmware>
```

The hierarchy of file system in Auto-Provisioning Service server is as below. Please note that if firmware is put in another server, the file location will still need to follow this hierarchy.



Provisioning Procedure

Provisioning Work Flow



Getting the APS Server Address

The default APS server must be provided by the Operator/ ITSP or DHCP options.

Users are allowed to change the APS server by option "APS Server Address" via Web Configuration Interface, and they can disable APS function by clear this option or uncheck the "Enable APS" option. "Enable APS" option is false by default.

For example, find options below in \$MAC.xml, and modify the values.

```
<APS_Enable type="integer">1</APS_Enable>
<APS_Server_Type type="integer">0</APS_Server_Type>
<APS_Server type="string">172.16.1.198</APS_Server>
```

DHCP Type (Default Network Type)

IP3032 phone will try to get the APS server address from DHCP option 66/150/159/160. The option checking priority will be 160/159/150/66, and APS Server address supports both IP address and URL.

If the phone does not get this information, it will use the local APS server address setting.



Static IP

The phone will use the APS server address specified in the web interface of the phone.

Firmware Upgrade

After the APS is enabled manually, IP3032 phone will try to download TCM_IP3032.xml from the APS server with the format of http://server-address/firmware/TCM_IP3032.xml.

For example, it will request the file as http://aps.customer.com/firmware/TCM_IP3032.xml.

When the provisioning IP3032 XML file is downloaded successfully, then it will perform the firmware upgrade check by checking <Firmware_Version>.

1. If the version of firmware in the phone is different from the what is specified in the provisioning file, the phone will try to download the new firmware from the address specified in the <Firmware_URL>. After the firmware upgrade, the phone will reboot and perform the same provision action above again.

2. If the firmware version is the same, the phone will skip the firmware upgrade and proceed to configuration update.
3. If the <Firmware_URL> is not specified in the xml file, it will use the same server address as specified for downloading the TCM_IP3032.xml.

Getting the Configuration File

If the firmware version requested is the same as the version on the phone, or the firmware update fails, it will try to download the \$MACADDRESS.xml file from the APS server with the format of [http://server-address/\\$MAC.xml](http://server-address/$MAC.xml) .

For example, the phone will request <http://aps.customer.com/001915123456.xml>.

APS Check Timing

The IP3032 phone will check and download the \$MACADDRESS.xml periodically when the Interval Time expires and phone is in an idle state (i.e., not on a call). The polling interval can be set in the Auto Provisioning section of the web interface in seconds. If it is not specified in \$MACADDRESS.xml, IP3032 phone will use default setting of 1440 minutes; if it is set to 0, the phone will only poll the \$MACADDRESS.xml when the phone boots up.

```
<APS_Period type="integer">1440</APS_Period>
```

The timing for the phone to perform the APS checking:

1. Boot up and be initialized if the APS_On_Boot is set to "1"
<APS_On_Boot type="integer">0</APS_On_Boot>
2. Interval time elapse.
3. APS Server address is changed or "APS Enable" is switched to "on".
4. APS check retry (Refer to the next section).

The APS checking UI:

When upgrading firmware from APS server, success or fail, LCD will show alerting information to let user know the upgrading state and result.

For example, IP3032 LCD shows "Updating... Please Wait!" or "Update Fail!" or "Update Success!".

APS Check Retry

Now we have defined some provisioning parameters as defaults.

- APSCheckPeriod: 1440 minutes.
Note: It could be configured by the provision file: 30~65535 minutes.

- APSErrorRetryTimes: 5.
- APSErrorRetryDelay: 60 seconds.

If the phones send the Provisioning Request, but fails to get the \$MACADDRESS.xml file, or decrypting the file fails, the APS will be contacted again after APSErrorRetryDelay, and try APSErrorRetryTimes. If not successful after that, the phone will then provision after APSCheckPeriod.

If the provisioning is successful, the phone will poll the server again after APSCheckPeriod to check if there is newer firmware or a newer configuration file.

Part 3: Configuring Your System

Part 3 describes the advanced phone features you can configure for your IP3032 phones. These features include a number of phone features that add efficiency and convenience, audio and video features, and several security features.

Part 3 consists of the following chapters:

- Chapter 5: Setting Up Advanced Phone Features
- Chapter 6: Setting Up Phone Audio Features
- Chapter 7: Setting Up User and Phone Security Features

Chapter 5: Setting Up Advanced Phone Features

This chapter will show you how to configure all available advanced phone features and call features. It provides important information you need to know in order to successfully perform configuration changes for the following advanced features:

- Assigning Multiple Line Keys Per Registration
- Assigning Call Progress Tones
- Configuring Network Address Translation
- Using a Corporate LDAP Directory

This chapter also shows you how to make configuration changes for the following advanced call server features:

- Configuring Shared Line Appearances
- Using Busy Lamp Field
- Enabling Voicemail Intergration
- Enabling Multiple Registrations
- Setting Up Backup Servers

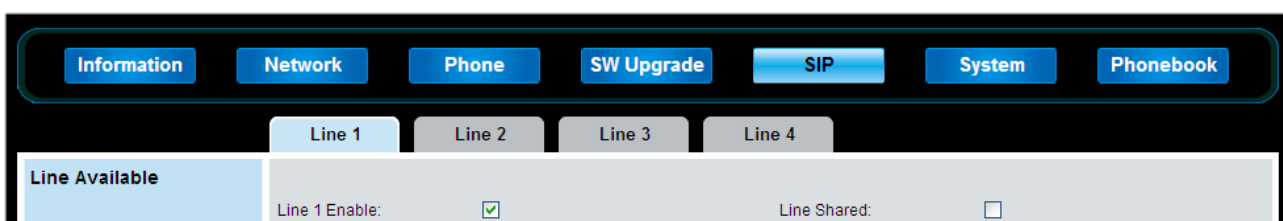
Assigning Multiple Line Keys Per Registration

You can assign a single registered phone line to multiple line keys on IP3032. This feature can be useful for managing a high volume of calls to a line.

Example Multiple Line Keys Configuration

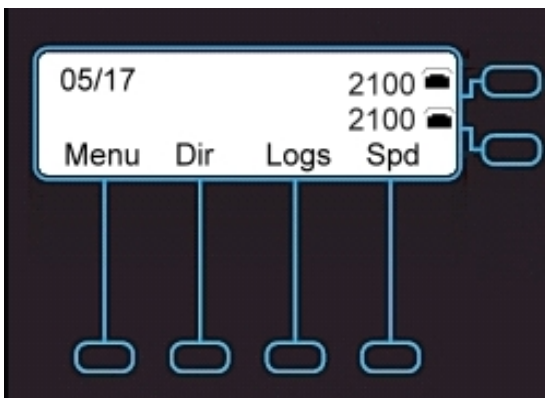
IP3032 can support up to 4 line keys with the same registered line address. In this example, it will show you how to enable four line keys with the same registered line number, 2100.

1. Configure Line1 as the registered line number, 2100.
 - A. Access the Web Configuration Interface by entering the phone's IP address in a web browser, for example, `http://<phone IP address>`. For administrators, log in as **Admin**, and the default password is **1234**.
 - B. Select **Line1** tab which is under **SIP** tab.



- C. Scroll down to **Line Available**, and then tick the **Line1 Enable** item.
 - D. Scroll down to **SIP Proxy Server, SIP Registrar Server, Subscriber Information**,....etc, and then enter all required registration information of the line number, 2100.
 - E. Scroll down to the bottom, and then press "**Save Settings**" button to save the configuration parameters.
2. Duplicate Line1 configuration file to Line2.
- A. Scroll down to **Line Configuration Mirroring** in Line1 tab.
 - B. Select "Line2" option, and then press "**Dup**" button to duplicate Line1 configuration parameters to Line2.
 - C. Select **Line2** tab which is under **SIP** tab.
 - D. Scroll down to **Line Available**, and then tick the **Line2 Enable** item.
 - E. Scroll down to the bottom, and then press "**Save Settings**" button to save the configuration parameters.
3. Duplicate Line1 configuration file to Line3 and Line4
- A. Follow Step2 procedure to duplicate Line1 configuraiton file to Line3 and Line4.

The IP3032 phone will display the registered line number 2100 on four line keys, as shown next.



Assigning Call Progress Tones

IP3032 phone plays call signals and alerts, called call progress tones, such as busy signals, ringback sounds, and call waiting tones. It can support 11 different call progress tones for the following countries.

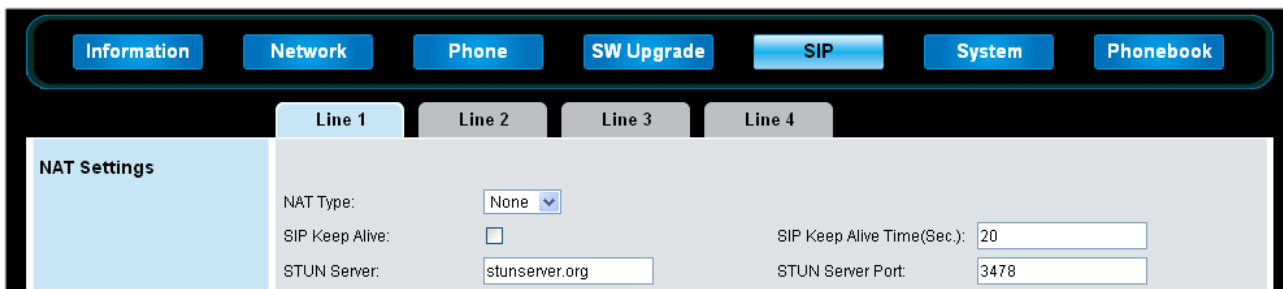
- United States
- China

- United Kingdom
- Canada
- Japan
- France
- Korea
- Germany
- Singapore
- HongKong
- Taiwan

The default call progress tones on IP3032 phone match standard United States tones. You can assign your IP3032 phone to match the standard tones in your region.

Configuring Network Address Translation

IP3032 phone can support STUN (Simple Traversal of UDP through Network Address Translators - RFC 3489) type of NAT. The STUN type of NAT allows IP3032 phone operating behind a NAT to discover the presence of the network address translator and to obtain the mapped NAT address and port number that the NAT has allocated for the IP3032 phone's UDP connections to remote hosts. IP3032 phone requires assistance from a third-party network server (STUN server) located on the opposing (public) side of the NAT, usually the public Internet. However, please note that STUN does not work with symmetric NAT which is often found in the networks of large companies.



The screenshot shows the NAT Settings configuration page for Line 1. The page has a navigation bar with tabs for Information, Network, Phone, SW Upgrade, SIP, System, and Phonebook. Below the navigation bar are tabs for Line 1, Line 2, Line 3, and Line 4. The NAT Settings section includes the following fields:

NAT Type:	None	SIP Keep Alive Time(Sec.):	20
SIP Keep Alive:	<input type="checkbox"/>	STUN Server Port:	3478
STUN Server:	stunserver.org		

Using Corporate LDAP Directory

You can connect your IP3032 phone to a corporate directory server that supports Lightweight Directory Access Protocol (LDAP). The corporate directory is a flexible feature and the following table links you to the parameters you can configure. When the configuration is done, you can call number you retrieve from the LDAP server on the IP3032 phone.

Please note that if you would like to make the corporate directory be used for number lookup on incoming calls, you need to configure Number Guessing Option as **LDAP**, and specify a parameter for LDAP Number Filter field.

The screenshot shows the 'LDAP Parameters' configuration page. It includes several input fields and dropdown menus for configuring LDAP settings. The 'Number Guessing Option' is set to 'Phonebook'. Other fields include LDAP Server Address (192.168.1.100), LDAP Login Name (cn=Manager,o=Tecom,c=c), LDAP Base (o=Tecom,c=cn), LDAP Name Filter, LDAP Security (None), LDAP Server Port (389), LDAP Login Password (masked), LDAP Max. Hits (6), and LDAP Number Filter.

Hereunder is an example. When there are many phone number types for a contact in the LDAP server, you should specify a parameter for the LDAP Number Filter field according to your need.

The screenshot shows the 'Person Details' window for 'DUGUGANG DUGUGANG'. It displays fields for First Name, Last Name, Nickname, E-mail, Home Address, Phone Home (6789), Mobile, Description or Note, Business Address, Phone Business (9876), and Fax. A table of contacts is shown with columns for Name, Phone, DOB, and Tags. Red boxes and arrows highlight the 'Phone Home' and 'Phone Business' fields, with text explaining how to set the LDAP Number Filter based on the selected phone type.

If the number that you would like to look up is Phone Home, 则 Number filter 为 homePhone

If the number that you would like to look up is Phone Bussiness 则 Number filter 为 telephoneNumber



IP3032 phone currently supports Open LDAP Directory Servers, but others may work as well.

To learn how to configure Open LDAP Directory server in your network, please refer to the content in the website, <http://www.openldap.org/>.

Configuring Shared Line Appearances

Shared Line Appearances connect calls and lines to multiple phones. With the shared line appearances feature enabled, an active call displays simultaneously on multiple phones in a group. By default, the answering phone has sole access to the incoming call, called line seize. You can enable another phone in the group the ability to enter a conversation, called a barge in. If the answering phone places the call on hold, that call becomes available to all phones of the groups. The parameters you can configure are listed as below. All call state of a call – active, inactive, on hold – are displayed on all phones of a group.

Example Shared Line Appearances Configuration

This feature is dependent on support from a SIP call server. IP3032 phone currently supports **Broadsoft PBX** only.

To enable shared call appearances on your phone, you will need to obtain a shared line address from your SIP service provider and make settings on SIP tab of the Web Configuration Interface.

Information	Network	Phone	SW Upgrade	SIP	System	Phonebook
<div style="display: flex; justify-content: space-around;"> Line 1 Line 2 Line 3 Line 4 </div>						
Line Available Line 1 Enable: <input checked="" type="checkbox"/> Line Shared: <input checked="" type="checkbox"/>						
SIP Proxy Server Server Mode: Broadsoft SIP Proxy Server: 60.250.158.234 Port: 5060 Outbound Proxy Server: Port: 5060 Backup Proxy Server: Port: 5060						
SIP Registrar Server Registrar Server: 60.250.158.234 Port: 5060 Registrar Outbound Server: Port: 5060 Backup Registrar Server: Port: 5060 Registrar Expire Time (Sec.): 60						

Using Busy Lamp Field

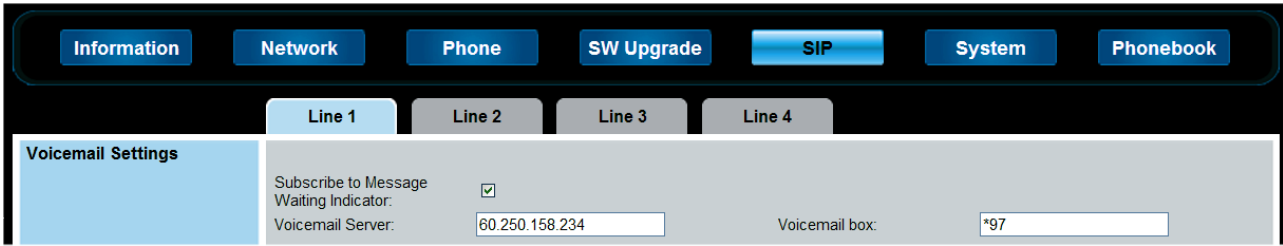
The busy lamp field (BLF) feature enables user to monitor the status of lines on remote phones, display remote party information, and answer incoming calls to remote phones (called directed call pickup). The BLF feature must be supported by a call server and the specific functions will vary with the call server you use. You may need to consult your SIP server partner to find out how to configure BLF.

Enabling Voicemail Integration

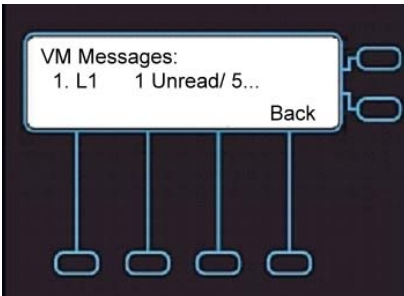
The phone is compatible with voicemail servers. You can configure each line registration per phone to access voicemail with a sequence numbers. When IP3032 phone gets new message(s), it will give a visual alert on MWI LED and text display on LCD screen.

Example Voicemail Configuration

The following illustration shows you how to enable one-touch to access voicemail server. In the illustration, line1 is configured to subscribe to Message Waiting Indicator and to the voicemail server at 60.250.158.234; the configured sequence numbers to access Voicemail box is *97.

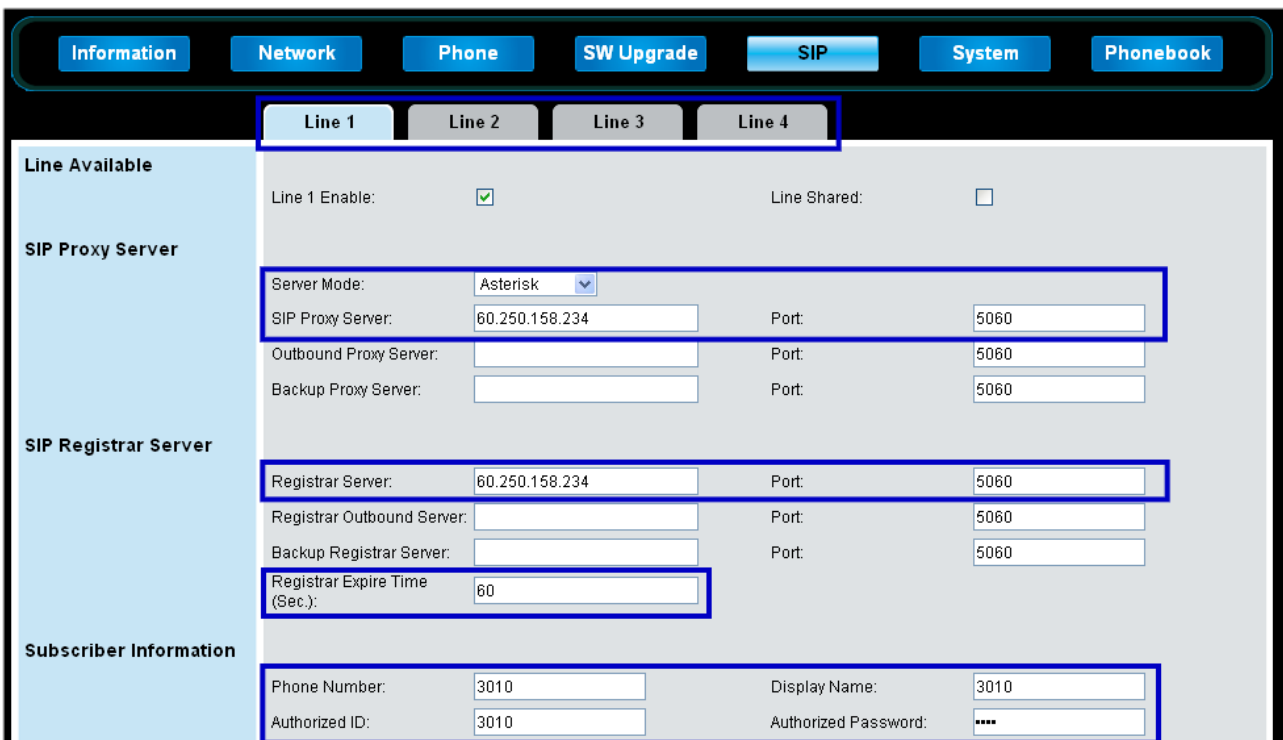


When user presses the Message key of IP3032, the phone will display the configured line1 Voicemail Messages, for example as below.



Enabling Multiple Registrations

IP3032 phone can support up to four registrations. The following illustration explains the registration parameters and options on the Web Configuration Interface. Each registration can be mapped to one or more line keys. Note that a line key can be used for only one registration. The user can select which registration to use for outgoing calls or which to use when initiating new instant message dialogs.



Setting Up Backup Servers

Backup Servers are often required in VoIP deployments to ensure continuity of phone service if, for example, the call server needs to be taken off-line for maintenance, the server fails, or the connection between the phone and the server fails.

Backup Servers Settings

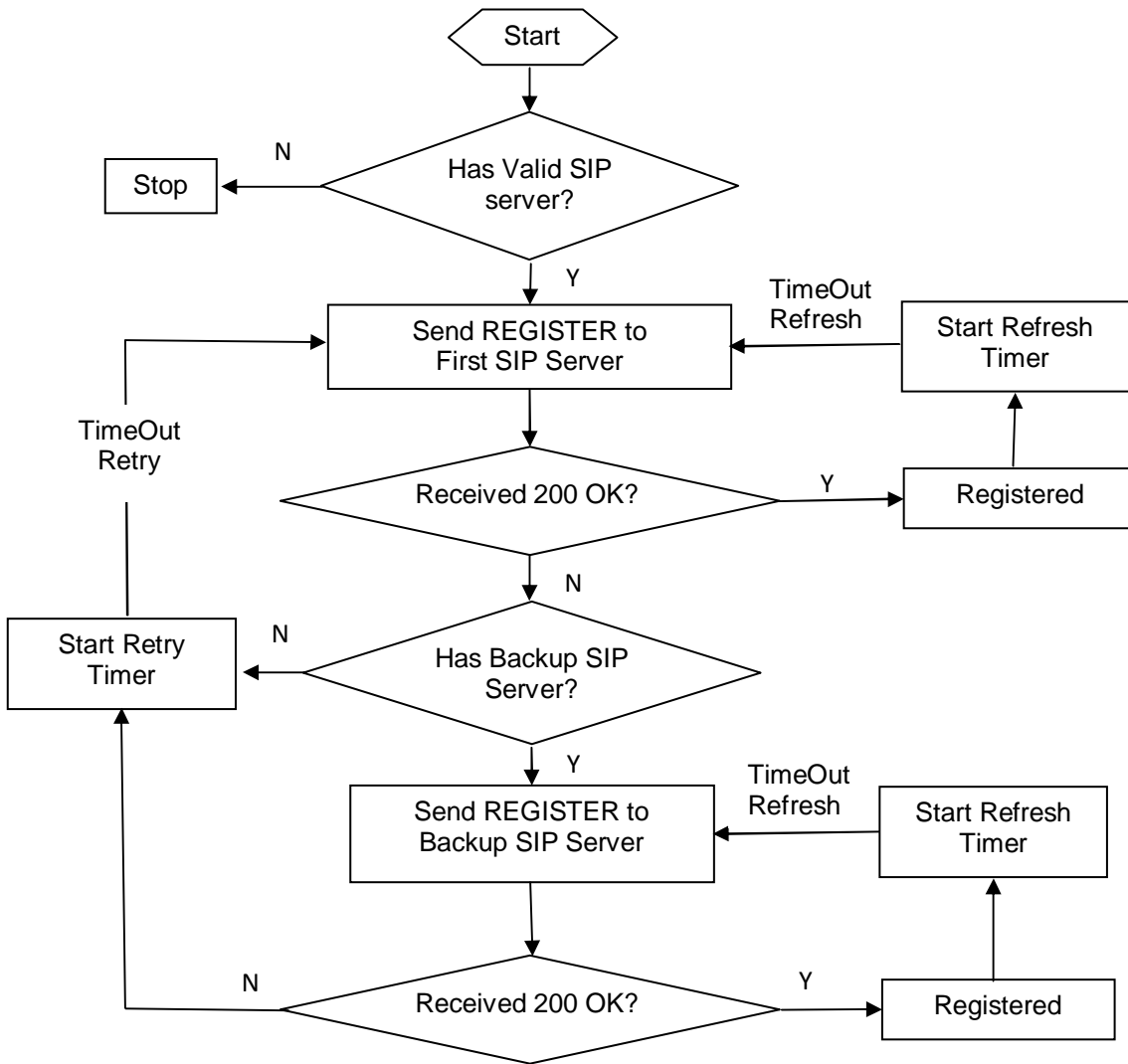
To use backup servers, you need to configure the Backup Proxy Server and Port fields and the Backup Registrar Server and Port fields as the following illustration.

The screenshot displays the SIP configuration interface for Line 1. The interface includes a navigation bar with tabs for Information, Network, Phone, SW Upgrade, SIP, System, and Phonebook. Below the navigation bar, there are tabs for Line 1, Line 2, Line 3, and Line 4. The main configuration area is divided into two sections: SIP Proxy Server and SIP Registrar Server. The SIP Proxy Server section includes fields for Server Mode (set to Asterisk), SIP Proxy Server (60.250.158.234), Port (5060), Outbound Proxy Server, Port (5060), Backup Proxy Server, and Port (5060). The SIP Registrar Server section includes fields for Registrar Server (60.250.158.234), Port (5060), Registrar Outbound Server, Port (5060), Backup Registrar Server, Port (5060), and Registrar Expire Time (Sec.) (60). The Backup Proxy Server and Backup Registrar Server fields are highlighted with a blue border.

Section	Field	Value
SIP Proxy Server	Server Mode	Asterisk
	SIP Proxy Server	60.250.158.234
	Port	5060
	Outbound Proxy Server	
	Port	5060
	Backup Proxy Server	
Port	5060	
SIP Registrar Server	Registrar Server	60.250.158.234
	Port	5060
	Registrar Outbound Server	
	Port	5060
	Backup Registrar Server	
	Port	5060
Registrar Expire Time (Sec.)	60	

SIP Servers Registration Procedure

From the following illustration, you can learn how a phone registers to the first SIP servers and when it will use the backup servers.



Chapter 6: Setting Up Phone Audio Features

After you set up your IP3032 phones on the network, users can send and receive calls using the default configuration. However, you might consider modifications that optimize the audio quality of your network.

Frequency bandwidth is one of the most critical elements affecting the intelligibility of speech in telephony. Complicating the intelligibility of telephony speech in today's world is background noise, variations in environmental reverberation, and communication among persons speaking a variety of native languages. While VoIP technology can broaden the frequency bandwidth and improve sound quality and intelligibility, it can also increase the network load and create a demand for lower raw bit rates.

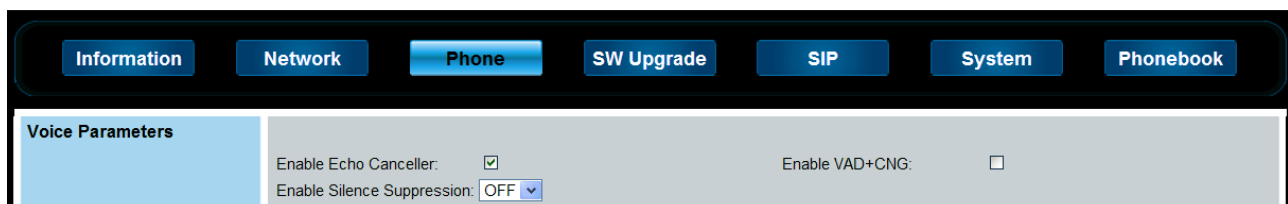
This chapter describes the audio sound quality features and options you can configure for your IP3032 phones. Use these features and options to optimize the conditions of your organization's phone network system.

This chapter shows you how to update your configuration for the following audio-related features:

- Acoustic Echo Cancellation and Voice Activity Detection
- Generating Dual Tone Multi-Frequency (DTMF) Tones
- Audio Codecs

Acoustic Echo Cancellation and Voice Activity Detection

The IP3032 phone uses advanced acoustic echo cancellation (AEC) for handsfree operation using the speakerphone. In addition, it also provides voice activity detection (VAD) which is to detect periods of silence in the transmit data path, so the phone doesn't have to transmit unnecessary data packets for outgoing audio. This process conserves network bandwidth. The AEC and VAD parameters in the following illustration will help you set up this feature.



Generating Dual Tone Multi-Frequency (DTMF) Tones

The IP3032 phone generates dual tone multi-frequency (DTMF) tones in response to user dialing on the dial pad. These tones are transmitted in the real-time transport protocol (RTP) streams of connected calls. The phone can encode the DTMF tones using the

active voice codec or using RFC2833-compatible encoding. The DTMF parameters in the following illustration will help you set up this feature. For the DTMF Type selection, you may need to consult your SIP server partner to find out which is the suitable one.

The screenshot shows the SIP configuration interface for Line 1. The 'DTMF Type' dropdown menu is highlighted, showing 'RFC2833' as the selected option. Other visible fields include 'Phone Number: 3016', 'Display Name: 3016', 'Authorized ID: 3016', 'Authorized Password: ****', 'Enable CLIP: [checked]', 'RFC2833 Payload: 97 (96-127)', 'Intercom Code: *80', and 'Sync Feature Key: SIP-INFO'.

Audio Codecs

The following table details the audio codec support for IP3032 phone.

<i>Audio Codec</i>	<i>Raw Bit Rate</i>	<i>Default Payload Size</i>
G.711 u-law	64 Kbps	20 ms
G.711 A-law	64 Kbps	20 ms
G.723.1	5.3 Kbps	30 ms
	6.3 Kbps	30 ms
G.726-32	32 Kbps	20 ms
G.729	8 Kbps	20 ms
iLBC	15.2 Kbps	20 ms
	13.33 Kbps	30 ms

You can find Codec Settings in the Web Configuration Interface, as shown in the following illustration.

The screenshot shows the 'Codec Settings' page for Line 1. It features a list of seven codecs with dropdown menus for selection: 'First Codec: None', 'Second Codec: G.711 u-law', 'Third Codec: G.711 a-law', 'Fourth Codec: G.729', 'Fifth Codec: G.723.1', 'Sixth Codec: G.726-32', and 'Seventh Codec: iLBC'. The 'G.723.1 Bit Rate' is set to '5.3kb/s'. Other settings include 'Packet Time: 20(ms)' and 'G.726-32 Payload: 114 (96-127)'.

Chapter 7: Setting Up User and Phone Security Features

After setting up your IP3032 phones on your network, users can place and answer calls using the default configuration. However, you may require some security-related changes to optimize your system for best results.

This chapter shows you how to update your configuration for the following security-related features:

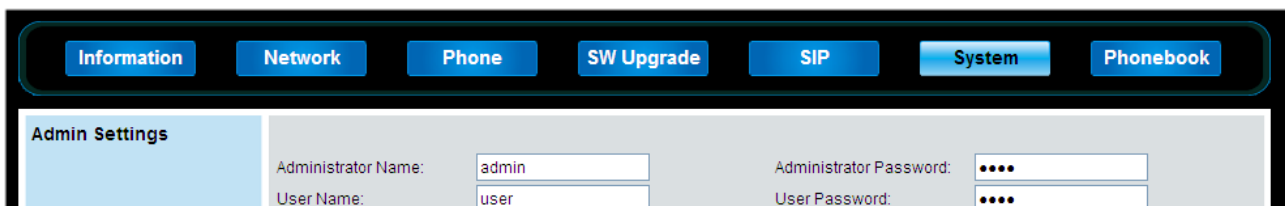
- Local User and Administrator Passwords
- Locking the Phone

Local User and Administrator Passwords

Several local settings menus are protected with administrator password. The phone will prompt for administrator password before granting access to the menu options. The Web Configuration Interface is protected by the user and administrator password and different page will display depending on which is used. The default user name is **user**, and the default user password is **1111**. The default administrator name is **admin**, and the default administrator password is **1234**. You should change the administrator password from the default value.

Web Configuration Interface

To change the user or administrator password, navigate to **System** tab. To change the administrator password, you must log in to the Web Configuration Interface as an administrator.



The screenshot shows the Web Configuration Interface with the **System** tab selected. The **Admin Settings** section is visible, containing the following fields:

Administrator Name:	<input type="text" value="admin"/>	Administrator Password:	<input type="password" value="...."/>
User Name:	<input type="text" value="user"/>	User Password:	<input type="password" value="...."/>

Local Phone User Interface

To change the administrator password, navigate to **Menu > Admin Setting**, enter the current administrator password, and select **Account Setting > Admin Name and Admin Password**.

To change the user password, navigate to **Menu > Admin Setting**, enter the current administrator password, and select **Account Setting > User Name and User Password**.

Locking the Phone

User can lock their phones, and prevent access to the menu or key presses, by entering **Phone Locked** mode through the phone menu.

Once the phone is locked, all user features and access to menus are disabled. The phone screen will show as below.



However, the IP3032 phone still can receive incoming calls.

To unlock the phone, the user enters the **PIN** code (Default: 1111 as user password for Web Configuration Interface access), and presses **OK** key; if it is entered correctly, the phone returns to its normal idle state.

In case the user forgets their PIN code, the system administrator can unlock their phone by entering admin password (Default: 1234 as administrator password for Web Configuration Interface access), and pressing the **OK** key.

Part 4: System Maintenance Tasks

Part 4 provides you with the information for upgrading your IP3032 phones firmwares and some maintenance tasks such as configuration file backup and updates, dial plan, system log and reset to default.

Part 4 consists of the following chapters:

- Chapter 8: Upgrading Your IP3032 Phones Firmware
- Chapter 9: Miscellaneous Maintenance Tasks

Chapter 8: Upgrading Your IP3032 Phones Firmware

IP3032 phones support several different ways to update its firmware, please refer to the following table.

No.	Method	Description	User	Admin.	Dist.
1	Auto-Provision Upgrade with MOCET APS	Software Patch (update.ip3032) upgrade via TFTP/ FTP/ HTTP/ HTTPS mass provisioning by using Tecom APS protocol with encrypted XML configuration file.	✓	✓	✓
2	Upgrade using Web Browser on A Specified Computer	Download SW Update Pack, includes Linux kernel, application pack and software patch, from web browser on a PC.	✗	✓	✓
3	Upgrade Using TFTP/FTP/HTTP/HTTPS Server	Software Patch (update.ip3032) and XML configuration file update via TFTP/FTP/HTTP/HTTPS server.	✗	✓	✓
4	Engineering Key Sequences on Root Menu	Download Software Patch (update.ip3032) from TFTP server.	✗	✓	✓
5	Emergency Upgrade on Boot	When the system partition on flash is damaged, press the special keys combination in booting to download complete flash image from TFTP server at fixed IP address.	✗	✓	✓
6	Updating images through Console by U-boot	This method is required to have a dedicated console cable and take apart housings. It downloads image from TFTP server by issuing download commands through a dedicated console cable.	✗	✗	✓

Auto-Provision Upgrade with MOCET APS

To learn how to set up the provisioning server, please refer to Chapter 4: Setting Up the Provisioning Server.

Please note that for general users, they are unable to use auto-provision upgrade if their Operator/ ITSP do not provide Auto-Provisioning System (APS).

Upgrade Using Web Browser on a Specified Computer

You can upgrade IP3032 phone firmware through Web Configuration Interface on administrator mode.

To save upgrade time, IP3032 supports individual firmware upgrade as following.

- Kernel Upgrade
- Application Pack Upgrade
- Software Patch Upgrade



Please note that you have to wait for the firmware update completed and make sure there is no interrupt during the update progress.

Kernel Upgrade

1. Select SW Upgrade tab of Web Configuration Interface.
2. Press **Choose File** button to select a kernel image named as, [kernel.ip3032](#), for Linux Kernel option.
3. Press **Update** button to start Kernel Upgrade.

Information	Network	Phone	SW Upgrade	SIP	System	Phonebook
HTTP Upgrade						
Linux Kernel:		<input type="button" value="Choose File"/>	No file chosen	<input type="button" value="Update"/>		
Application Pack:		<input type="button" value="Choose File"/>	No file chosen	<input type="button" value="Update"/>	<input type="button" value="Update&Reset"/>	
Software Patch:		<input type="button" value="Choose File"/>	No file chosen	<input type="button" value="Update"/>	<input type="button" value="Update&Reset"/>	

Application Pack Upgrade

It is used to upgrade related folder in which the applications are changed frequently such as /tcmhome, /etc.

1. Select SW Upgrade tab of Web Configuration Interface.

2. Press **Choose File** button to select an application pack named as, [apps.ip3032](#), for Application Pack option.
3. Press **Update** button to start Application Pack Upgrade, or press **Update&Reset** button to make the update start and also reset the configuration settings to the factory default.

Information	Network	Phone	SW Upgrade	SIP	System	Phonebook
HTTP Upgrade						
Linux Kernel:	<input type="button" value="Choose File"/>	No file chosen	<input type="button" value="Update"/>			
Application Pack:	<input type="button" value="Choose File"/>	No file chosen	<input type="button" value="Update"/>	<input type="button" value="Update&Reset"/>		
Software Patch:	<input type="button" value="Choose File"/>	No file chosen	<input type="button" value="Update"/>	<input type="button" value="Update&Reset"/>		

Software Patch Upgrade

It is used to upgrade some applications such as tcconfig, tecomphone, webs, network, tcmgui, www and so on for bug hot fixes.

1. Select SW Upgrade tab of Web Configuration Interface.
2. Press **Choose File** button to select a patch image named as, [update.ip3032](#), for Software Patch option.
3. Press **Update** button to start Software Patch Upgrade, or press **Update&Reset** button to make the update start and also reset the configuration settings to the factory default.

Information	Network	Phone	SW Upgrade	SIP	System	Phonebook
HTTP Upgrade						
Linux Kernel:	<input type="button" value="Choose File"/>	No file chosen	<input type="button" value="Update"/>			
Application Pack:	<input type="button" value="Choose File"/>	No file chosen	<input type="button" value="Update"/>	<input type="button" value="Update&Reset"/>		
Software Patch:	<input type="button" value="Choose File"/>	No file chosen	<input type="button" value="Update"/>	<input type="button" value="Update&Reset"/>		

Upgrade Using TFTP/FTP/HTTP/HTTPS Server

When you have many phones to update, but the Auto-Provisioning System is not available, then the TFTP/FTP/HTTP/HTTPS Server Upgrade function will help you to save time when updating phones.



Tip for You

Please note that the upgrade method only supports Software Patch upgrade; the software file name is [update.ip3032](#).

Information **Network** **Phone** **SW Upgrade** **SIP** **System** **Phonebook**

TFTP/FTP/HTTP/HTTPS Upgrade

Server Type:

Server IP Address:

Server User Name:

Software File:

Configuration File:

Files Directory:

User Password:

Engineering Key Sequences on Root Menu

During the phone reboot, press engineering key sequences on root menu to download Software Patch (update.ip3032) from TFTP server.

The default Engineering Key Sequences for updating from TFTP server:

“*” + “873283” (“update” in the keypad) + “1”**

You can configure the Key Sequences through Web Configuration Interface on administrator mode. As for the IP address of TFTP server, you can input its IP address in a popup window of the phone menu after pressing the Key Sequences on root menu.

Information **Network** **Phone** **SW Upgrade** **SIP** **System** **Phonebook**

Key Sequences

Reboot:

Update By TFTP:

Debug Mode:

Reset:

Emergency Upgrade on Boot

When your phone is unable to boot up, you may try this emergency upgrade method to recover your phone. By pressing special keys combination in booting, the phone will download complete flash image from a specified download server.

Please note that IP address of the specified download server and IP address of the phone are fixed, and are unable to configure.

- IP address of the phone: 192.168.1.10
- IP address of the specified download server: 192.168.1.100

Hereunder is the procedure for emergency upgrade on boot.

1. Set up a TFTP server with IP address: 192.168.1.100.
2. Put the image file and the corresponding file of MD5 checksum in the same directory of the TFTP server.

3. Connect the phone and the TFTP server in the same LAN.
4. Unplug power from the phone.
5. Press the following key sequences, and do not release the buttons.
 - Press button "*" and "3" at the same time, it will update U-BOOT image.
The image file: nandboot.flash.ip3032
The corresponding file of MD5 checksum: nandboot.flash.ip3032.md5
 - Press button "*" and "6" at the same time, it will update Kernel image.
The image file: ulmage.ip3032
The corresponding file of MD5 checksum: ulmage.ip3032.md5
 - Press button "*" and "9" at the same time, it will update Root File System image.
The image file: dspg.jffs2.ip3032
The corresponding file of MD5 checksum: dspg.jffs2.ip3032.md5
 - Press button "*" and "Line" at the same time, it will update blob image.
The image file: ip3032.image.blob
The corresponding file of MD5 checksum: ip3032.image.blob.md5
6. Plug power to the phone.
7. Release the key sequence buttons after the phone's MWI LED turns on.

Updating Images through Console by U-boot

This firmware upgrade method is only applicable for distributor who has been authorized in the local repair right. For its upgrade guide, please refer to Appendix A - Updating Images through Console by U-boot.

Chapter 9: Miscellaneous Maintenance Tasks

This chapter shows you how to maintain the IP3032 phones. This includes:

- RTP Port Base
- Configuration File Backup
- Configuration File Updates
- Optional SIP Header
- SIPs Parameters
- RTP Options
- Dial Plan
- System Log
- Session Timer
- Reset to Default

Real-Time Transport Protocol (RTP) Port Base

You can specify the IP3032 phone's RTP starting port. Since IP3032 phone supports conferencing and multiple RTP streams, the phone can use several ports concurrently. The default parameter of RTP starting port is 10000. You can configure it based on your requirement. Please note that the starting port number must be entered as an even integer.

The screenshot shows the 'RTP Parameters' section of the Web Configuration Interface. The 'RTP Port Base' is set to 10000. The interface includes a navigation bar with tabs for Information, Network, Phone, SW Upgrade, SIP, System, and Phonebook.

Configuration File Backup

To backup the configuration file of your phone, you can use the backup function through Web Configuration Interface. The backup configuration file name is **PhoneConfig.xml**.

The screenshot shows the 'HTTP Upgrade' section of the Web Configuration Interface. It includes fields for Linux Kernel, Application Pack, Software Patch, and Configuration File, each with a 'Choose File' button and a 'No file chosen' status. There are 'Update' and 'Update&Reset' buttons for each field. The 'Backup' button is highlighted with a red box.

Configuration File Updates

When you have many phones to deploy, but Auto-Provisioning System is not available, then Configuration File Updates function will help you to save time to configure phones. Hereunder is an example with the minimum necessary information to configure a working phone on a SIP server.

1. Configure Line1 settings under SIP tab.

The following fields marked by blue rectangles are the minimum necessary configuration to make the phone work on a SIP server.

The screenshot shows the SIP configuration page for Line 1. The 'SIP' tab is selected. The configuration is organized into sections: Line Available, SIP Proxy Server, SIP Registrar Server, and Subscriber Information. Blue rectangles highlight the following fields:

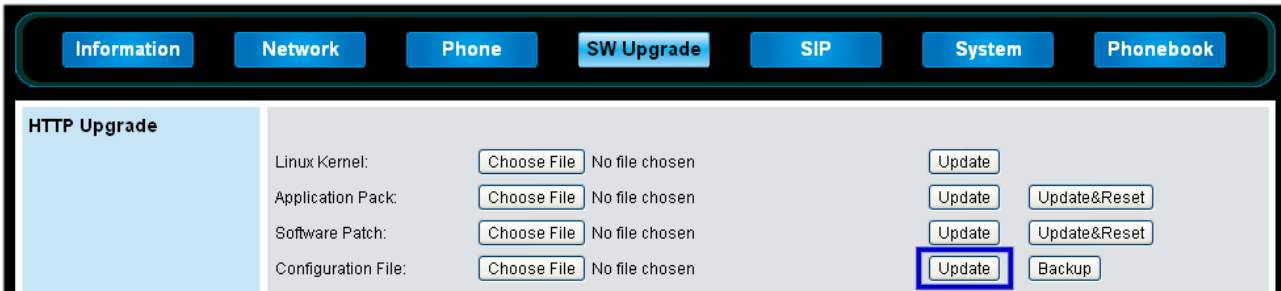
- Line 1 Enable:
- Line Shared:
- Server Mode: Asterisk (dropdown)
- SIP Proxy Server: 60.250.158.234
- Port: 5060
- Outbound Proxy Server: (empty)
- Port: 5060
- Backup Proxy Server: (empty)
- Port: 5060
- Registrar Server: 60.250.158.234
- Port: 5060
- Registrar Outbound Server: (empty)
- Port: 5060
- Backup Registrar Server: (empty)
- Port: 5060
- Registrar Expire Time (Sec.): 3600
- Phone Number: 2102
- Display Name: 2102
- Authorized ID: 2102
- Authorized Password: ***

2. Use Configuration File Backup function to download its configuration file as a template.
3. Use XML editor to modify the configuration file for other phones. The minimum necessary configuration includes the following four fields:

Line_Extension, Line_User, Line_Password and Line_DisplayName

```
<LineSettings index="0">
  <Line_Server_Mode type="integer">3</Line_Server_Mode>
  <Line_Enable type="integer">1</Line_Enable>
  <Line_Extension type="string">2102</Line_Extension>
  <Line_User type="string">2102</Line_User>
  <Line_Password type="string">2102</Line_Password>
  <Line_DisplayName type="string">2102</Line_DisplayName>
  .....
  .....
  .....
  .....
</LineSettings>
```

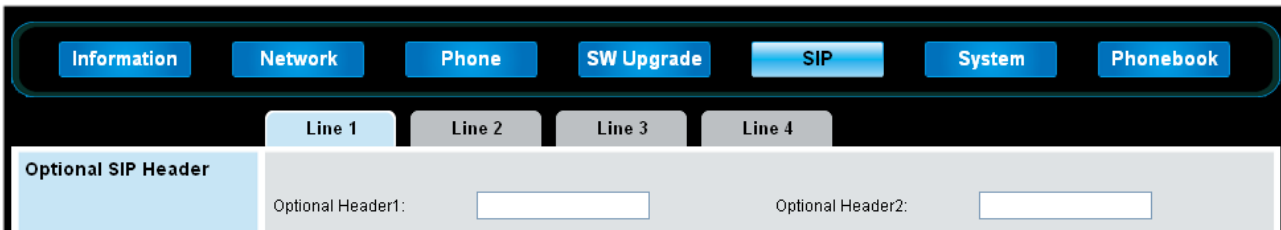
4. Use Configuration File Updates function to upload the modified configuration file to its corresponding phone.



Optional SIP Header

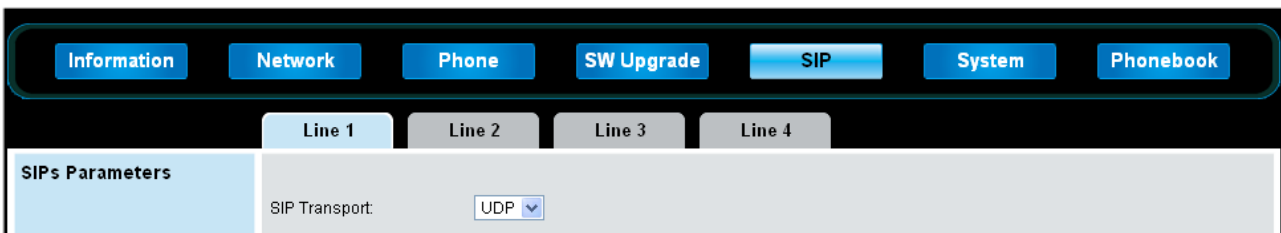
Optional SIP Header allows the service provider or administrator to define a custom SIP header. The SIP header will be contained in invite packets when making calls. The SIP Header format is: **header_name: header_value**.

For example: PhoneMAC: 00:19:15:33:ff:cc.



SIPs Parameters

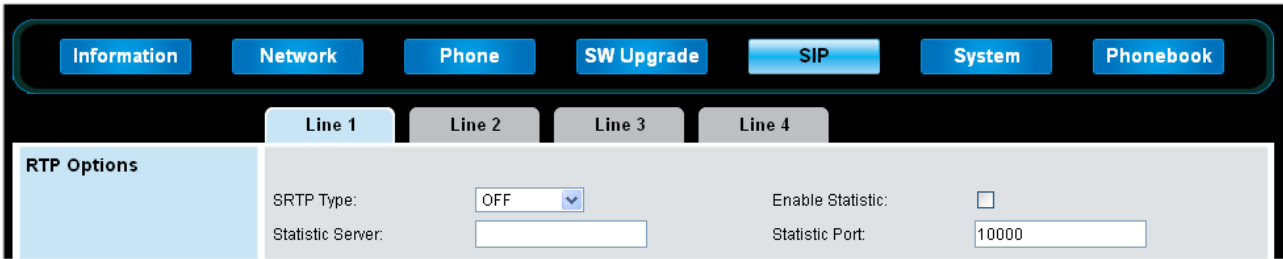
SIP Parameters defines what kind of transmission method to be used for SIP signals. There are options: UDP, TCP and TLS.



RTP Options

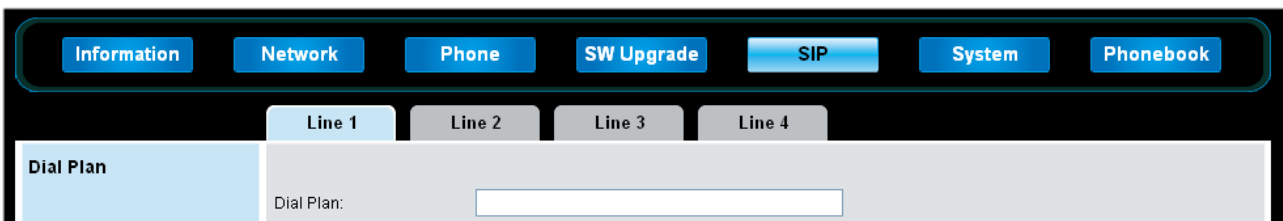
RTP Options allows the service provider or administrator to configure:

1. SRTP for RTP
2. RTP Statistics to be sent to a specific server



Dial Plan

This parameter allows you to create a specific routing path for outgoing SIP calls. When outgoing SIP call fits the dial plan, it will be called out immediately without waiting for Dial Timeout time.



A dial plan is a set of rules for determining whether a complete set of numbers has been entered. IP3032 uses industry-standard Regular expressions to match dialed numbers.

Dial Plan Entries

Dial plan entries have three parts. The parts may be separated with character “[|]”.

Regex Pattern	Result pattern	Flags
---------------	----------------	-------

The first part contains a regex pattern which is used for matching the dialed number. The second part contains the result or the dial plan step, and the third (optional) part contains flags that set additional processing attributes. The following flags are available:

The “d” flag means that the number is complete and can be dialed. For example: “([0-9]{5})\|1|d” means that a number with five digits will be dialed automatically.

The “i” flag means that the comparisons should be done case-insensitive.

The “t” flag means that an additional time out period (default is 4 seconds) should take place before automatic dialing starts.

A new value (2-9) after “t” flag can overwrite the default time. For example: “t8” means the time out is 8 seconds.

On the phone, a dial plan can contains multiple dial plan entries, use space character to separate them.

For example:

```
^911$|sip:911@someserver.com|d ([0-9]{4})\|1|di ([a-z]{8})|sip:\1@sipserver.com|t5
```

This dial plan has three entries:

- `^911$|sip:911@someserver.com|d`
- `([0-9]{4})\1|di`
- `([a-z]{8})|sip:\1@sipserver.com|t5`

Substitutions

Substitutions are found in the second part of the dial plan entry. Substitutions are marked by a leading character “\”.

The “d” replacement inserts the name of the SIP proxy.

For example:

`“[0-9]{4}|sip:999@\d|d”` inserts the SIP proxy behind the “@” symbol.

Numbers are back references to match-groups of the Regex part according to RFC2915.

For example:

`“[0-9]*|sip:\1@1.1.1.1|t”` inserts a “sip:” before the string (which is the first match), and add “@1.1.1.1” after it.

Note: Only the back references 1-4 are available.

Examples for Dial Plans

Convert an emergency number into a SIP URL.

This pattern could look like this: `^911$|sip:emergency@\d|d`

Separated by the exclamation mark, it contains the pattern for the 911 and the resulting SIP URI. The d flag indicates that there is no need to press the OK key after dialing this number.

Make the phone dial a number when the pound key is pressed.

The pattern could look like this: `([^\#]+)#|sip:\1@\d|d`

This dial plan entry will look for a pattern ending in a pound symbol and use this as the user name in a SIP URI (not including the pound symbol).

Matching an International Number

Just put the 011 pattern at the front of the pattern: `^011([0-9]*)$|sip:+\1@\d|t`

System Log

For System Log settings, you can set System Log to as Console, Local or Remote.

Information	Network	Phone	SW Upgrade	SIP	System	Phonebook
Admin Settings						
Administrator Name:	<input type="text" value="admin"/>		Administrator Password:	<input type="password" value="...."/>		
User Name:	<input type="text" value="user"/>		User Password:	<input type="password" value="...."/>		
System Log to:	<input type="text" value="Console"/>		Log Level:	<input type="text" value="Emergency"/>		
System Log Address:	<input type="text"/>		System Log Port:	<input type="text" value="514"/>		

When System Log to is set as Console, the system log will be printed to console.

When System Log to is set as Local, the system log will be saved in local PC.

When System Log to is set as Remote, the system log will be send to system log server. Please remember to fill in the fields of System Log Address and System Log Port, they are the system log server address and system log server port.

In addition, you can choose what log level that you want to view. There are eight levels for selection, Emergency, Alert, Critical, Error, Warning, Notice, Informational and Debug.

Session Timer

In Session Timer settings, it has two fields, one is for Session Expires time, and the other is for minimal Session Expires time. With the feature support, it adds IP3032 phone the capability to periodically refresh SIP sessions by sending repeated INVITE requests.

- Session Timer: the default time is 300 seconds.
- Minimum Session Expiration (Min-SE): the default time is 100 seconds.

Information	Network	Phone	SW Upgrade	SIP	System	Phonebook
Session Timer						
Session Expires(Sec.):	<input type="text" value="300"/>		Min-SE(Sec.):	<input type="text" value="100"/>		

Reset to Default

For Reset to Default settings, you can choose Reset Configuration only, Reset Phonebook only or Reset All (Reset Configuration and Phonebook).

Information	Network	Phone	SW Upgrade	SIP	System	Phonebook
Reset to Default						
Reset Configuration:	<input type="button" value="Reset Configuration"/>		Reset Phonebook:	<input type="button" value="Reset Phonebook"/>		
Reset All:	<input type="button" value="Reset All"/>					

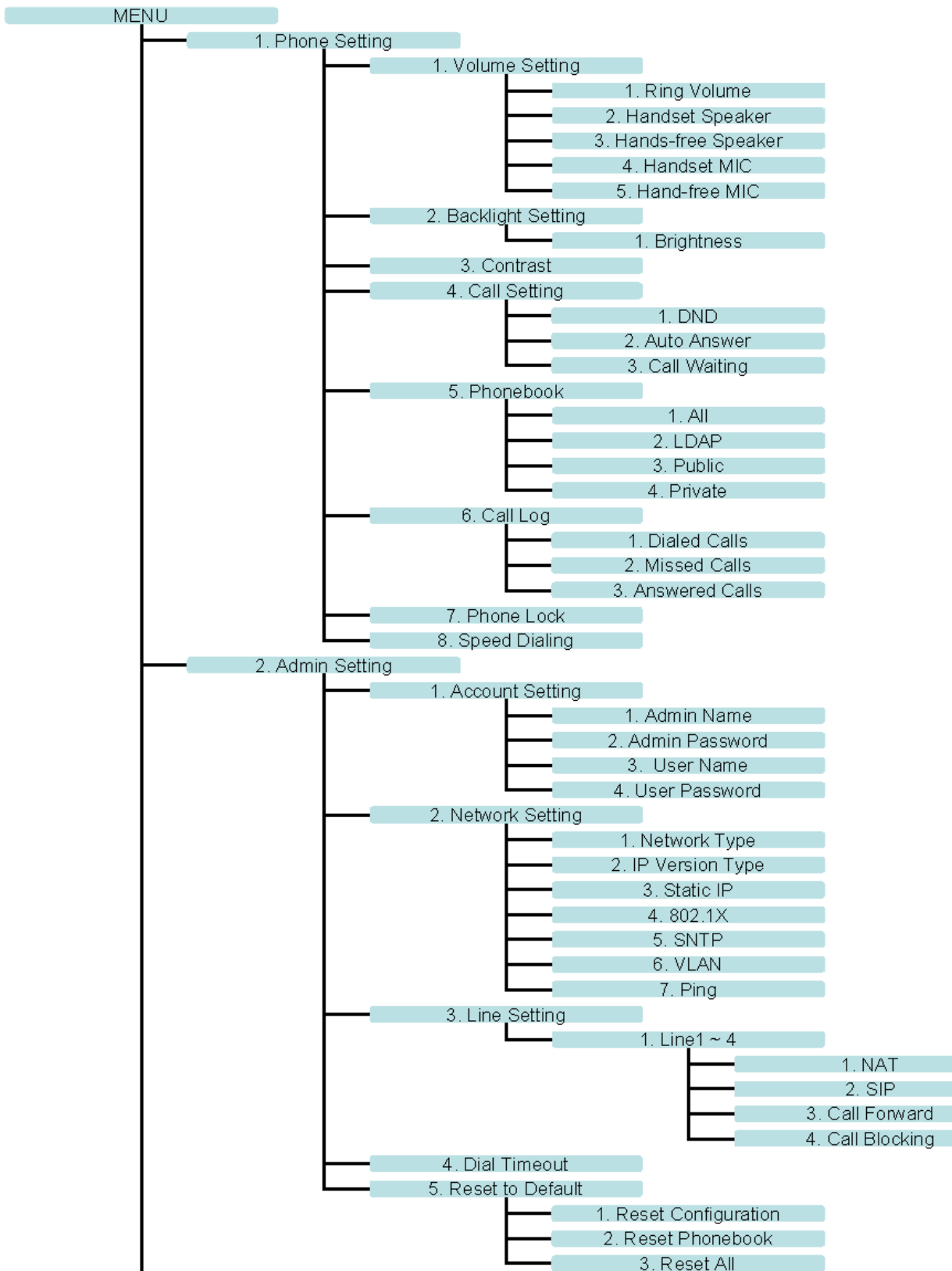
Part 5: References

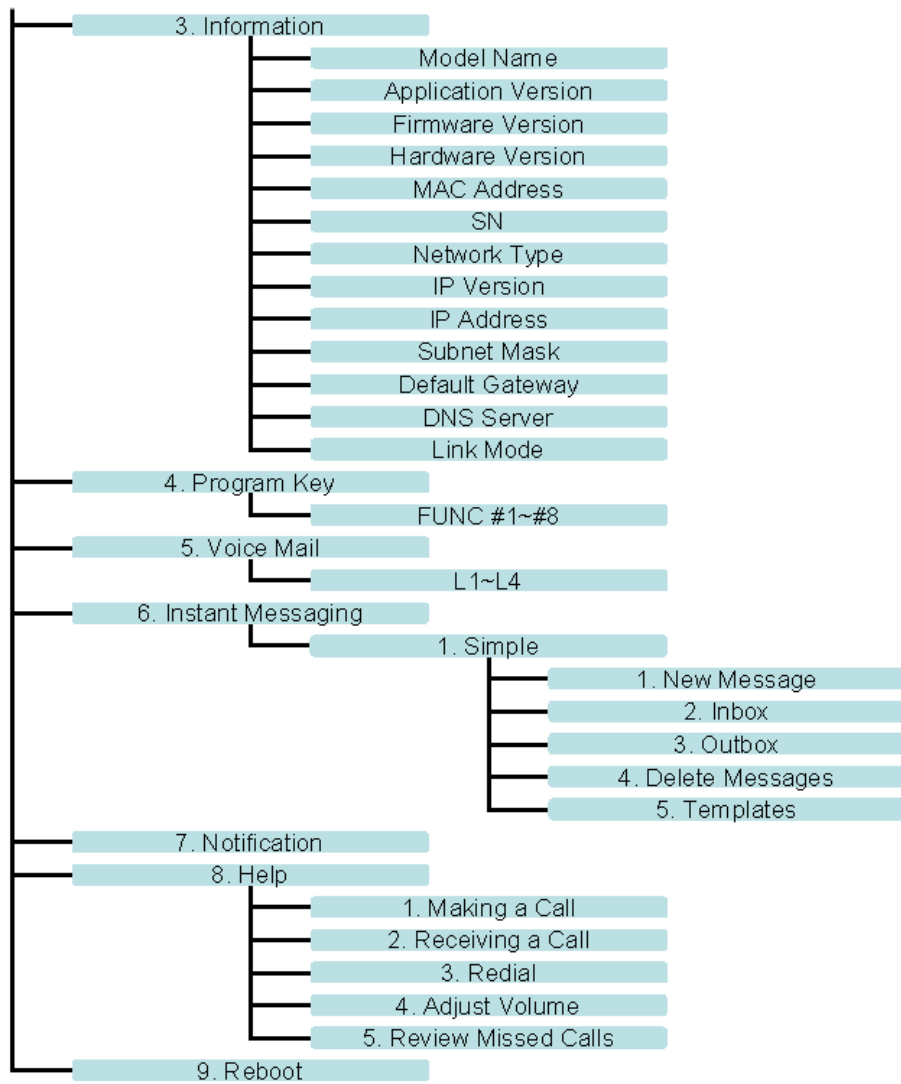
Part 5 provides you with reference information about the IP3032 firmware menu structure as it appears on IP3032 phones.

Part 5 consists of the following chapter:

- Chapter 10: IP3032 Firmware Menu System

Chapter 10: IP3032 Firmware Menu System





Appendix A – Upgrading Images through Console by U-boot

This firmware upgrade method is only applicable for distributor who has been authorized in the local repair right.

Preparing Materials

Before starting upgrade, please make sure you have the following materials ready.

Software List

- ✓ TFTP software (file name: tftpd32.328)
- ✓ Console cable driver (file name: Console cable driver-CP210x_Drivers)
- ✓ Telnet client software (file name: teraterm_utf8-4.58.exe)
- ✓ IP3032 combined image file (file name: ip3032.image.blob)
- ✓ The corresponding file of MD5 checksum for IP3032 combined image file (file name: ip3032.image.blob.md5)

Hardware List

- ✓ IP3032 phone and its adapter----- 1 unit
- ✓ PC or Notebook with one USB port and Ethernet port -----1 unit
- ✓ Console cable for IP3032 -----1 unit
- ✓ Switch hub and its adapter----- 1 unit
- ✓ Ethernet cable ----- 2 pcs

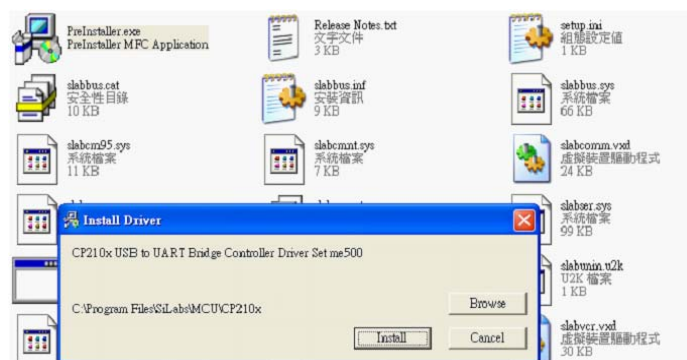
Software Environment Setup

Installing the Console Cable Driver

Step1: Open the file of “Console cable driver-CP210x_Drivers”.

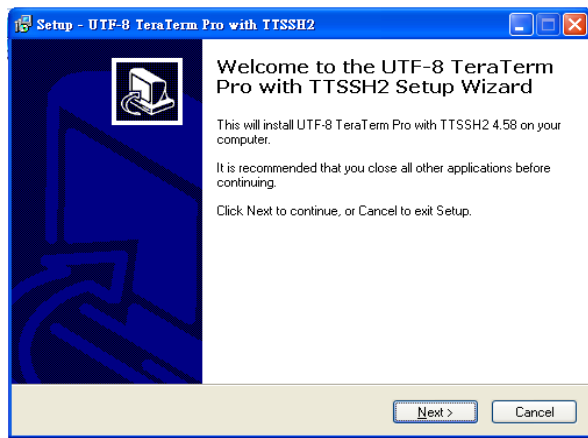
Step2: Open the file of “PreInstaller.exe”, and then the screen below will show up.

Step3: Press “Install” to start installation. (You may encounter some warning messages during installation. Please skip it. Finally, the installation completion message will show up.)



Installing the Telnet Client Software

Step1: Open the file of “teraterm_utf8-4.58.exe”, and then the screen below will show up.



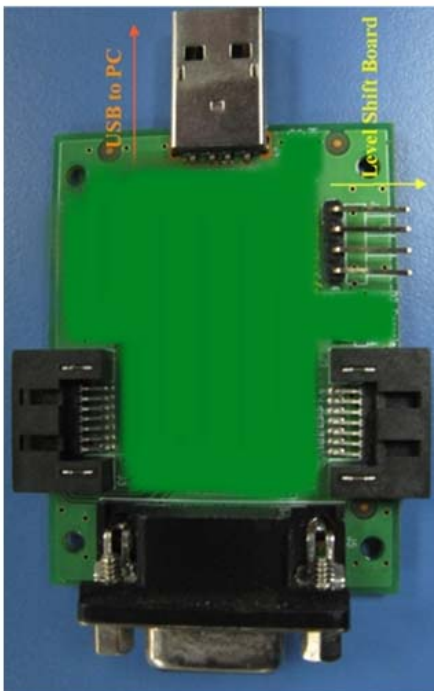
Step2: Press “Next” to continue the installation until installation completed.

Step3: Put “ip3032.image.blob” and “ip3032.image.blob.md5” in the file of tftpd32.328.

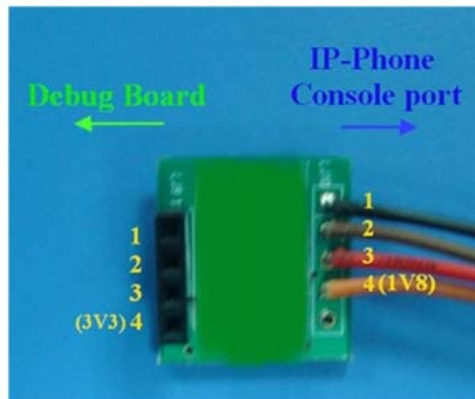
Hardware Environment Setup

Make Sure You Use the Correct Console Cable

Check your console cable is consisted of a debug board and a level shift board.



Debug Board



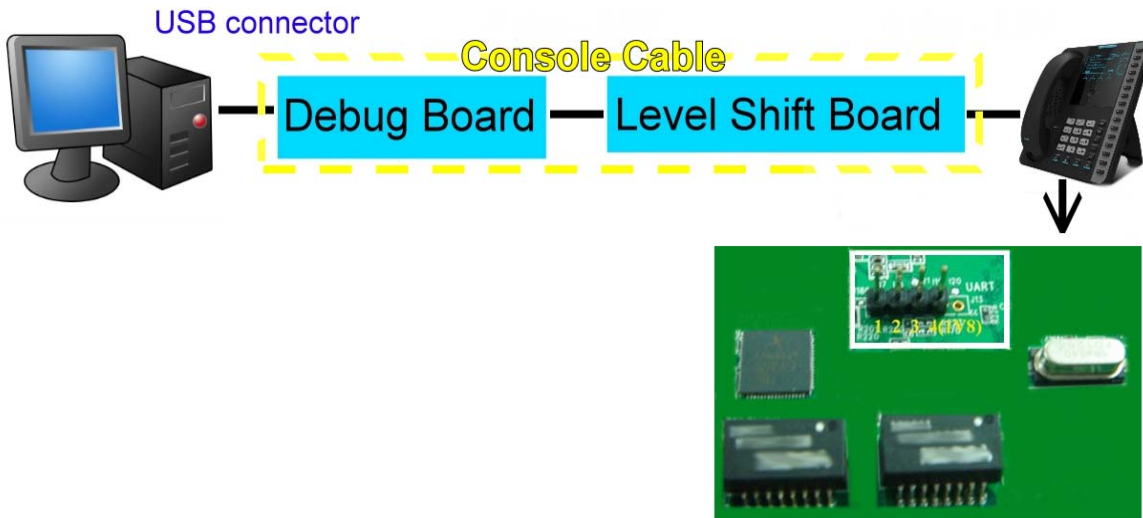
Level Shift Board

Connecting the Console Cable with PC/Notebook and the IP3032

Step1: Take apart IP3032 phone housing.

Step2: Insert USB plug of the console cable into USB port of PC/Notebook.

Step3: Insert 4 pin connector of the console cable into the 4 pin header on the main board of IP3032. Please note the direction of the connector.



Connecting an Ethernet Cable with PC/Notebook and Switch Hub

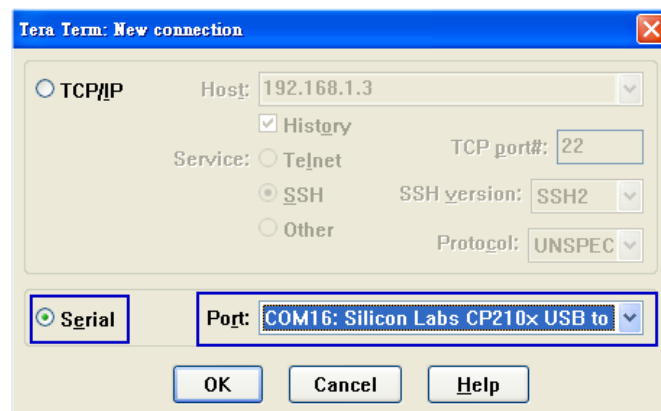
Connecting an Ethernet Cable with IP3032 Phone and Switch Hub

Upgrading Image through Console Port

Opening Tera Term Client Window

Step1: Select "UTF-8 Tera Term Pro".

Step2: Select "Serial", and set Port to "COM?" (COM? depends on the USB port that the console cable is connecting with.)

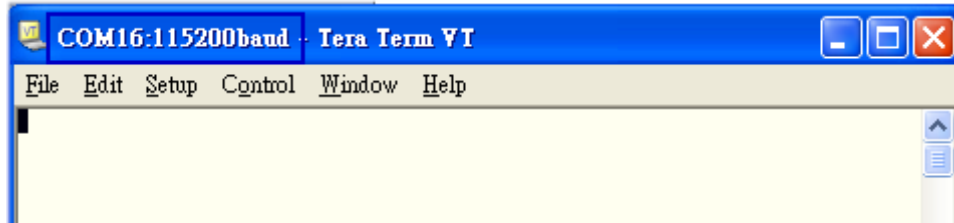


Step3: Press “OK”. The new window will pop up.

Step4: Select “Serial port” in the “Setup” menu of the new window.

Step5: Change baud rate from default, 9600, to 115200. Then press “OK”.

Step6: Then the screen below will show up. Please keep the window open.



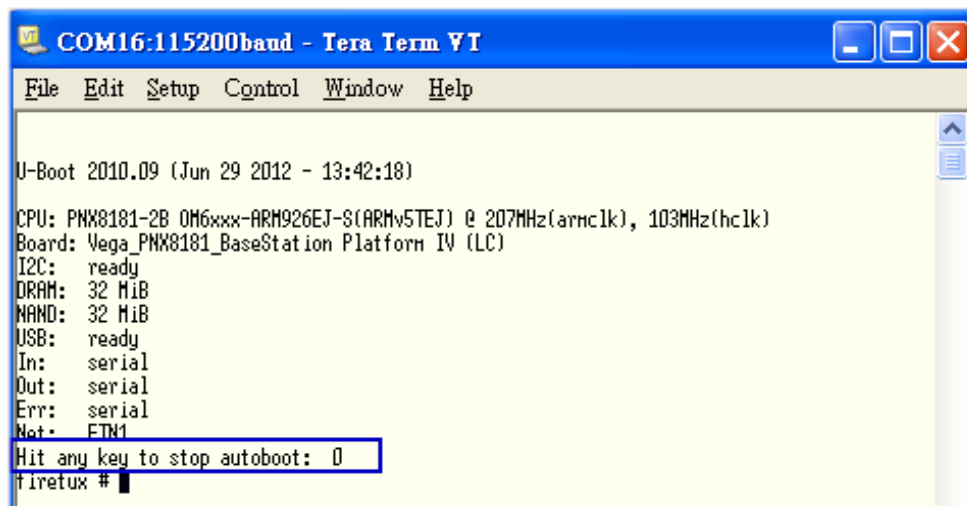
Opening TFTP Window

Step1: Open “tftpd32.exe”.

Step2: Change “server interface IP” to the IP address of PC/Notebook. Please keep the window open.

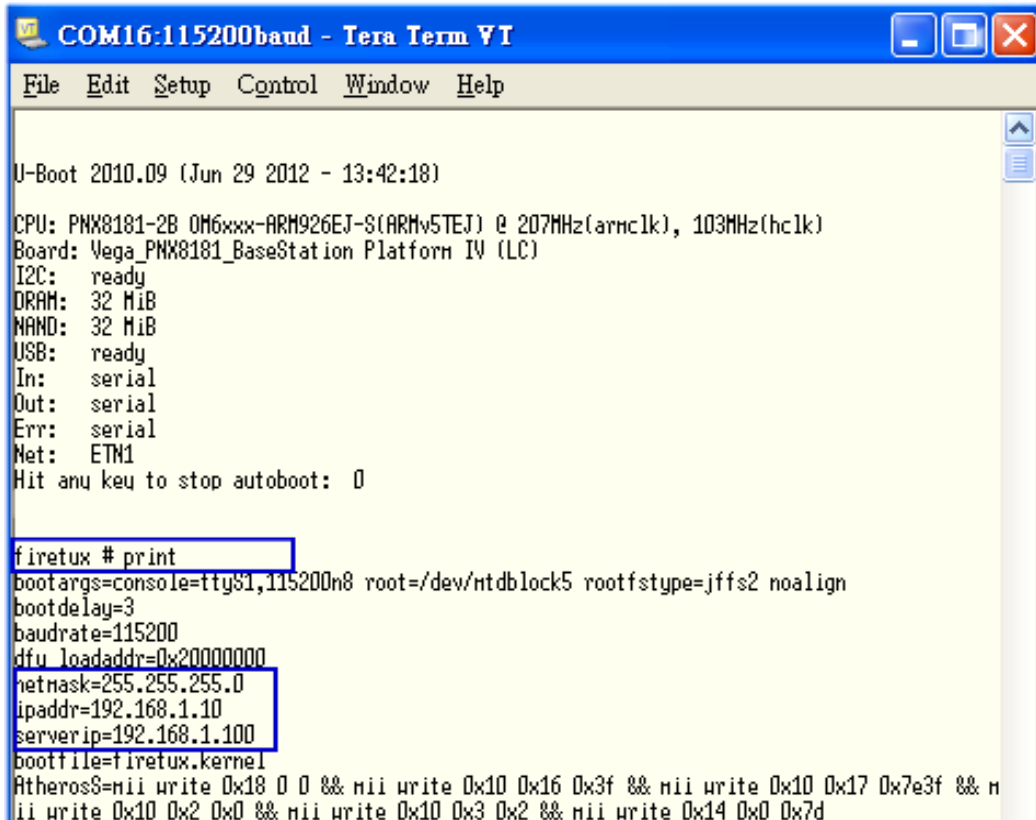
Environment Setting

Step1: Power on IP3032 phone, and then the telnet window will show the IP3032 phone running status. Before the count down number to “0”, press any key to stop the count down, so that you can enter commands through the console cable.



Step2: Set up the environment.

Enter a command: **print**, and press “Enter” key to confirm.



```
COM16:115200band - Tera Term VT
File Edit Setup Control Window Help

U-Boot 2010.09 (Jun 29 2012 - 13:42:18)

CPU: PNX8181-2B 0M6xxx-ARM926EJ-S(ARMv5TEJ) @ 207MHz(armclk), 103MHz(hclk)
Board: Vega_PNX8181_BaseStation Platform IV (LC)
I2C: ready
DRAM: 32 MiB
NAND: 32 MiB
USB: ready
In: serial
Out: serial
Err: serial
Net: ETM1
Hit any key to stop autoboot: 0

firetux # print
bootargs=console=ttyS1,115200n8 root=/dev/ntdblock5 rootfstype=jffs2 noalign
bootdelay=3
baudrate=115200
dfu_loadaddr=0x20000000
netmask=255.255.255.0
ipaddr=192.168.1.10
serverip=192.168.1.100
bootfile=firetux.kernel
AtherosS=mii write 0x18 0 0 && mii write 0x10 0x16 0x3f && mii write 0x10 0x17 0x7e3f && m
ii write 0x10 0x2 0x0 && mii write 0x10 0x3 0x2 && mii write 0x14 0x0 0x7d
```

Step3: Set TFTP server IP address of your computer.

Enter a command: **setenv serverip 172.18.149.62**, and press “Enter” key to confirm.

Step4: Set IP address of the IP3032 phone.

Enter a command: **setenv ipaddr 172.18.149.100**, and press “Enter” key to confirm.

Upgrading the Firmware

Step1: Upload the combined image file to the phone.

Enter a command: **tftpboot 0x20000000 ip3032.image.blob**, and press “Enter” key to confirm.

Exiting Debug Mode

Step1: After booting, the IP3032 phone will enter “Debug Test” mode.

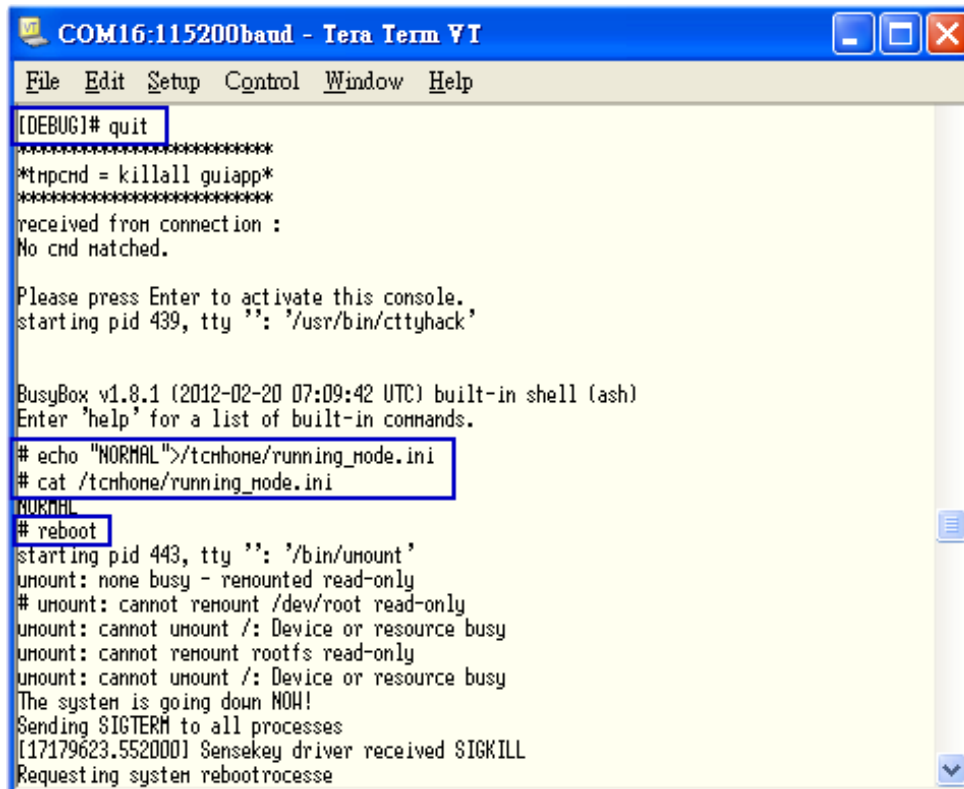
Step2: Exit from “Debug Test” mode to “Normal” mode.

Enter a command: **quit**, and press “Enter” key to confirm.

Enter a command: **echo “NORMAL”> /tcmhome/running_mode.ini**, and press “Enter” key to confirm.

Enter a command: **cat /tcmhome/running_mode.ini**, and press “Enter” key to confirm.

Enter a command: **reboot**, and press “Enter” key to confirm.



```
COM16:115200band - Tera Term VT
File Edit Setup Control Window Help
[DEBUG]# quit
*****
*tmpcmd = killall guiapp*
*****
received from connection :
No cmd matched.

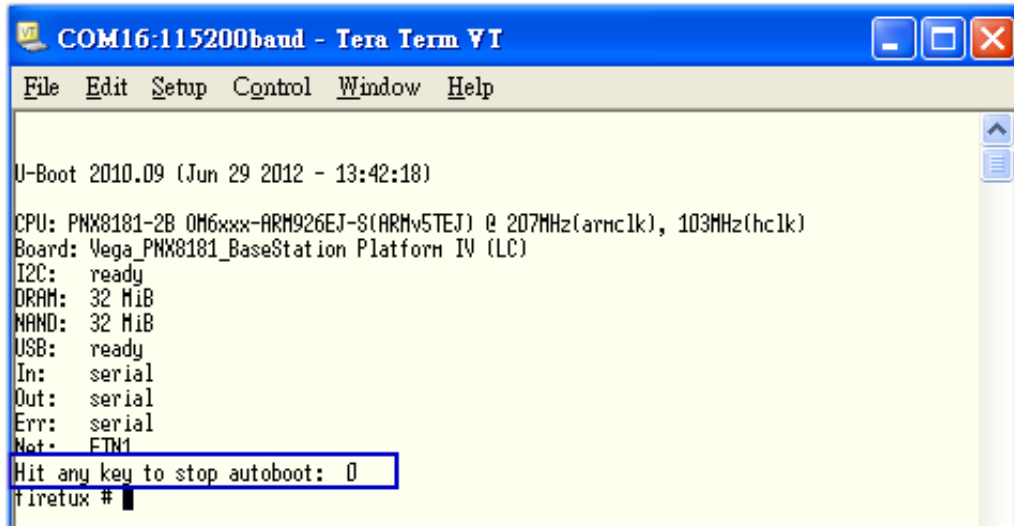
Please press Enter to activate this console.
starting pid 439, tty "": '/usr/bin/cttyhack'

BusyBox v1.8.1 (2012-02-20 07:09:42 UTC) built-in shell (ash)
Enter 'help' for a list of built-in commands.
# echo "NORMAL">/tcmhome/running_mode.ini
# cat /tcmhome/running_mode.ini
NORMAL
# reboot
starting pid 443, tty "": '/bin/umount'
umount: none busy - remounted read-only
# umount: cannot remount /dev/root read-only
umount: cannot umount /: Device or resource busy
umount: cannot remount rootfs read-only
umount: cannot umount /: Device or resource busy
The system is going down NOW!
Sending SIGTERM to all processes
[17179623.552000] Sensekey driver received SIGKILL
Requesting system rebootprocese
```

Setting MAC Address

Please note that the IP3032 phone’s MAC address will be reset to default after you complete the combined image upgrade. You should set the original MAC address of the phone through console before you start to use the phone. Otherwise, you will encounter MAC address conflict when using the phone. Please follow the steps below to set MAC address.

Step1: During the IP3032 reboots, press any key to stop the count down before the count down number to “0”, so that you can enter commands through the console cable.



```
COM16:115200baud - Tera Term VT
File Edit Setup Control Window Help

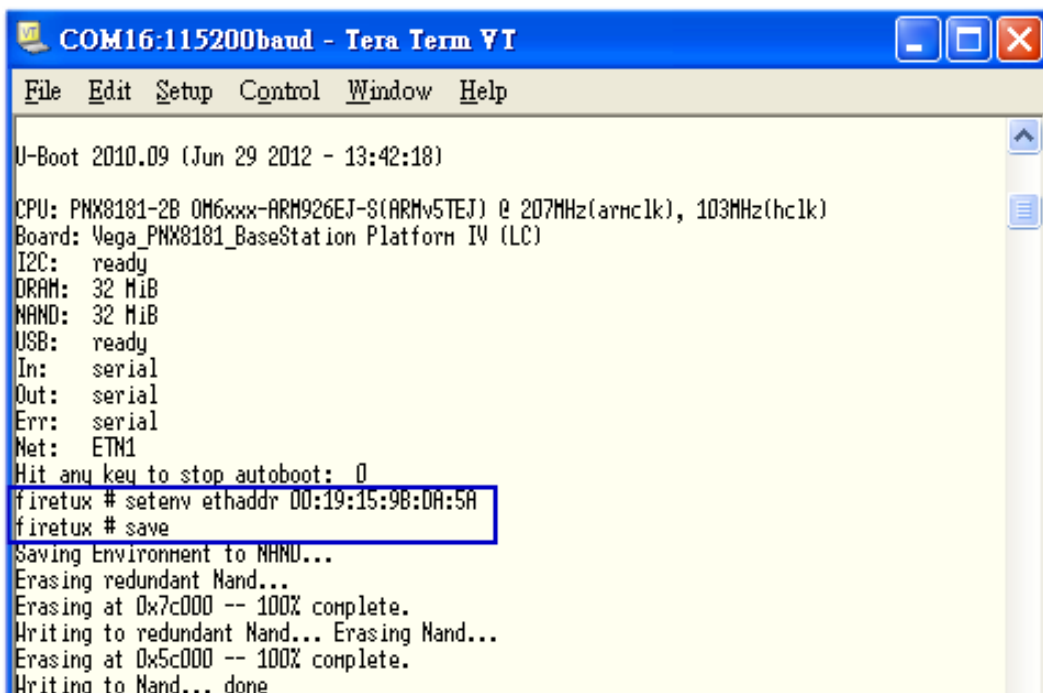
U-Boot 2010.09 (Jun 29 2012 - 13:42:18)

CPU: PNX8181-2B 0M6xxx-ARM926EJ-S(ARMv5TEJ) @ 207MHz(arnc1k), 103MHz(hc1k)
Board: Vega_PNX8181_BaseStation Platform IV (LC)
I2C: ready
DRAM: 32 MiB
NAND: 32 MiB
USB: ready
In: serial
Out: serial
Err: serial
Net: ETN1
Hit any key to stop autoboot: 0
firetux #
```

Step2: Set and Save MAC address on its U-boot.

Enter a command: **setenv ethaddr 00:19:15:9B:DA:5A**, and press “Enter” key to confirm.

Enter a command: **save**, and press “Enter” key to confirm.



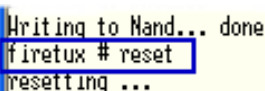
```
COM16:115200baud - Tera Term VT
File Edit Setup Control Window Help

U-Boot 2010.09 (Jun 29 2012 - 13:42:18)

CPU: PNX8181-2B 0M6xxx-ARM926EJ-S(ARMv5TEJ) @ 207MHz(arnc1k), 103MHz(hc1k)
Board: Vega_PNX8181_BaseStation Platform IV (LC)
I2C: ready
DRAM: 32 MiB
NAND: 32 MiB
USB: ready
In: serial
Out: serial
Err: serial
Net: ETN1
Hit any key to stop autoboot: 0
firetux # setenv ethaddr 00:19:15:9B:DA:5A
firetux # save
Saving Environment to NAND...
Erasing redundant Mand...
Erasing at 0x7c000 -- 100% complete.
Writing to redundant Mand... Erasing Mand...
Erasing at 0x5c000 -- 100% complete.
Writing to Mand... done
```

Step3: Reset the phone after the MAC address written done.

Enter a command: **reset**, and press “Enter” key to confirm.



```
Writing to Mand... done
firetux # reset
resetting ...
```