# **MOCET**

## IP3032 Standard IP Deskphone **User 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 as 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 is own expense, will be required to take whatever measures are necessary to correct the interface.

#### **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 replaced or dispose of them in an approved recycling center.

#### **Trademarks**

All company, brand and product names, like Metaswitch™, Broadsoft™, Freeswitch™ and Asterisk™ are registered trademarks of their respective owners.

#### **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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## **About this Guide**

This guide explains how to use the basic features of your new IP3032 phone. Not all features listed are available by default. Contact your system or network administrator to find out which features and services are available to you on your system.

Your System Administrator has the ability to customize some features on this phone. For information on more advanced settings and configurations, administrators should refer to the IP3032 Standard IP Deskphone Administrators' Guide.

## **Overview**

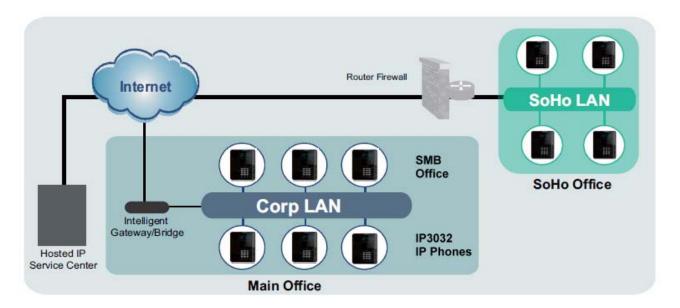
The MOCET IP3032 Standard IP Deskphone is the lowest cost of IP3000 series IP Phones. The following table highlights its key features:

Features	IP3032
Appearance	
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 keys (Up, Down, Left, Right and OK)	Yes (capacitive touch)
Programmable keys	8 keys
Audio quality (Speaker, Handset Receiver)	HD
Voice codec	G.711 (A-law/Mu-law), G.722 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)
Tilt stand	Yes (3 steps. 30°, 51° and 60°)
AC adaptor	5V/ 550mA (Optional)
Wall mount	Yes (Optional)

## 1. Introduction

The MOCET IP3032 Standard IP Deskphone 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 be positioned in multiple tilt angles and has a wall mount option 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-green backlit LCD or from the phone's web user interface. The IP3032 can be automatically provisioned from a local or Internet based server using the built-in MOCET auto-provisioning system 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 an 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.



#### 1.1. Phone Features and Specifications

- Complete VoIP and networking protocol support
- Rich supplementary call services and phone features
- HD acoustic hardware design for handset and speakerphone
- G.722 wideband audio codec support
- 4 SIP Lines with 8 call appearances
- 8 programmable keys with LEDs
- Interoperable with Asterisk, Freeswitch, Metaswitch and Broadsoft
- Auto-provisioning, remote management and security enhancements

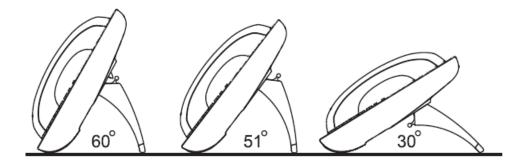
#### 1.2. Requirements

The IP3032 IP Phone requires the following environments:

- Compatible SIP-based IP PBX system or Internet-based hosted SIP service account
- Ethernet/Fast Ethernet LAN (10/100 Base-T)

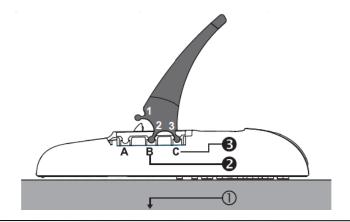
#### 1.3. Installation and Setup

The IP3032 can support three different tilt angles of 60°, 51°, and 30° (see below).



#### 1.3.1. Attaching the Stand to the Phone

After unpacking the box, attach the stand to the IP3032 phone first. Below this paragraph we illustrate an example below of 60° angle installation with the stand. There are three sets of "antlers" on the top of the stand (named 1, 2 and 3) and there are three sets of mounting slots on the back of the phone (named A, B and C). See the illustration below.



The procedures for attaching the stand to the phone are as follows:

Step ①: Place the main body of the IP3032 face down on your lap or a firm sofa;

Step 2: First snap the "2<sup>nd</sup>" antler into the slot "B" hole;

Step **3**: Then snap the "3<sup>rd</sup>" antler into the slot "C" hole (it may take a bit of pressure, but when it locks in place, you will hear a click and the stand will not wiggle on the phone.)

For other stand angle, such as 51° and 30°, please refer to the Installation Combination Table below to find correct slots and correct antlers to use.

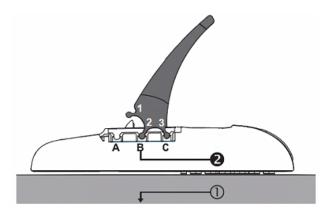


Snapping the slot "B" hole with the antler of stand first is the most important step for a smooth installation. The product sticker on stand should be facing inward towards the phone. When a stand is installed at 60° or 30°, you can rotate the stand between 30° and 60° quickly without pulling the antler in the slot B out.

#### 1.3.2. Detaching the Stand from the Phone

To remove the stand from the IP3032 phone, follow the procedures below (we are using a 60° stand as an example):

Step ①: Place the main body of the IP3032 face down in your lap or a firm sofa;



Step **2**: Place with two hands on the stand of the IP3032 and firmly press down the stand, the "2<sup>nd</sup>" antler will be pulled straight out of the slot "B" hole; Keep pressing down the stand, the "3<sup>rd</sup>" antler will be pulled straight out of the slot "C" hole.



## 1.3.3. Installation Combination Table

Different Tilt Angle Installation	Slot Holes on the Back Shell	Antlers of Stand
B2 C3 C3	B B C	3 2 2
60°	В	2
	С	3
A2 B3/D2	A B	2 2 3 2
51°	В	3
	Α	2
30°	A B	2 1
	В	2
	Α	1

#### 1.3.4. Connect the Handset to the Phone

After installing the stand, connect the handset to the curly handset cord and then connect the other end of the curly handset cord to the port marked HAND with the handset icon on the bottom of the IP3032.

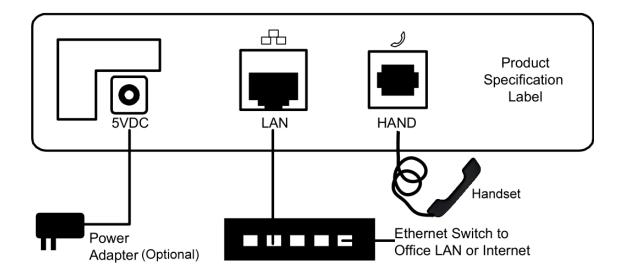
#### 1.3.5. Connect the Ethernet Cable

Using a general CAT-5 Ethernet cable, follow the installation steps below:

- (1) If your Ethernet Switch supports PoE:
  - Please connect an Ethernet cable to the Switch port from the dd LAN port of the IP3032. Then you will see the phone LEDs and buttons light up momentarily and the phone will boot.
- (2) If your Ethernet Switch doesn't support PoE:
  - Please connect an Ethernet cable to the Switch port from LAN port.
  - Plug in the IP3032 power adaptor to the wall power outlet then plug the barrel plug to the power port on the back of the phone. You should see the phone LEDs and buttons light up momentarily and the phone will boot.

#### **Notice:**

- 1. The IP3032 takes just under two minutes to start up and becomes operational. There are a series of LED and buttons that will light up periodically to provide boot progress information. Please be patient.
- 2. The IP3032 does not ship with a power adapter. If you are not using Power over Ethernet (PoE), you must order the power adapter separately from MOCET.

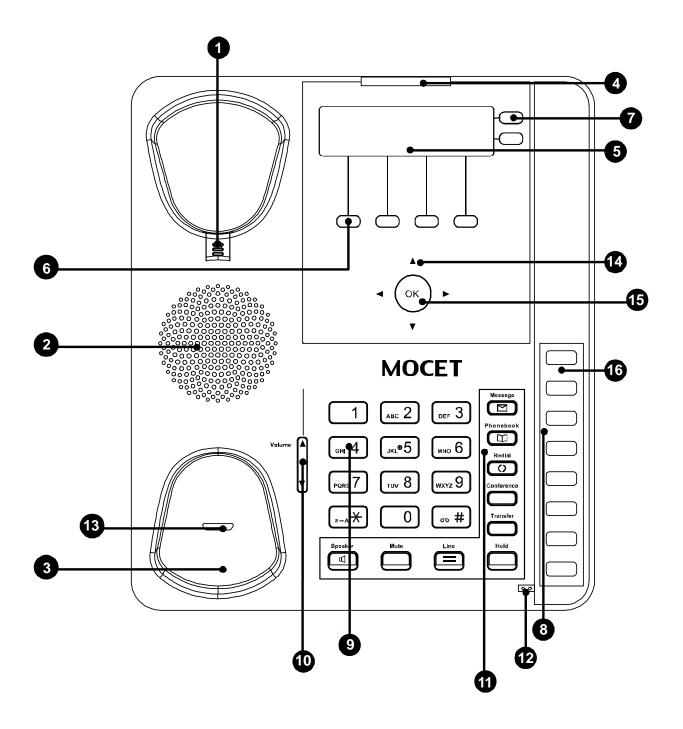




Do not use other power adapters. Use only the MOCET IP3032 Power Adapter with your IP3032 IP Phone. Adapters for other devices may damage IP3032 IP Phone.

## 1.4. Appearance and Function Description

The IP3032 Standard IP Deskphone comes with a blue-green graphic LCD display, capacitive touch keys, including 4 soft keys, 2 line keys, and 5 navigator keys, as well as a traditional keypad, 8 programmable keys and 9 function keys. In the box, a handset, curly handset cord and an Ethernet cable supplied. It can be installed and placed on the desktop or mounted on the wall. The wall mount accessory is available separately. The figure below illustrates the front view of the IP3032 IP Phone. With the point numbers, you can find its name and a simple description of specified part in the following table.



No	Part Name	Description	
1	Hanger	The hanger can be pulled; it is reversed for wall-mount installation.	
2	Speaker	For ring and hands-free talking.	
3	Hook switch & Handset bottom cradle	Handset The handset bottom cradle is for the placement of handset;	
4	Message LED	The message LED. Steady Blue means the phone is booting or upgrading.	
5	LCD Display	The LCD screen is used for displaying phone's settings, phone number, call status and so forth.	
6	Soft keys	Soft keys displayed on the LCD are used for item selection or control. The various soft key functions depend on the current activity on the phone as they are context-sensitive.	
7	Line keys  These keys are used for line selection. A blue LED is associated with each key to indicate its line/call status.		
8	Programmable keys can be used for indicating busy lamp field (BLF) status of other phones, speed dial phone numbers, or for activating features of a service or IP-PBX. A blue LED is associated with each key to indicate its status.		
9	1 = 2 = 3 = 4 = 5 = 8 = 7 = 8 = 9 Numeric keypad is for numbers.		
10	Volume Control key	The Volume Control key is used to set the loudness of the ringer, handset and speakerphone functions. While the phone is in idle state, the ringer is adjustable. While in a call, the handset or speaker volume is adjustable.	
11	Line Group key	The Line Group key is used to toggle between line group 1,2 to line group 3,4. Or switch back from line group 3,4 to line group 1,2.	
	MUTE key	The MUTE key is used to activate or deactivate the microphone on the IP3032. A blue LED is used to indicate whether the phone is muted or not.	

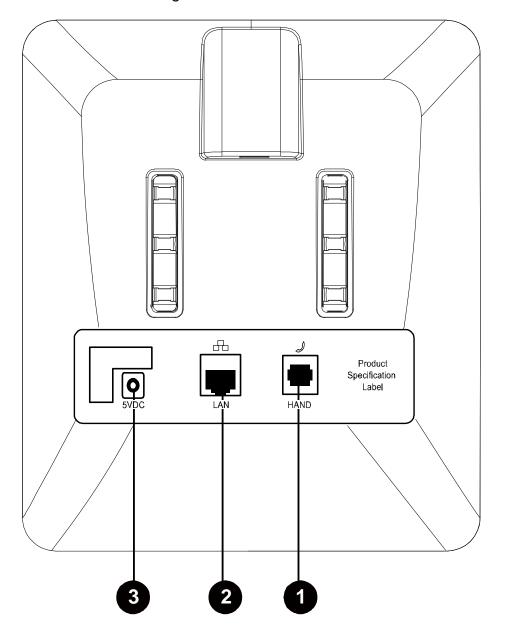
	Speaker Phone key	The SPKR key is used to activate or deactivate the Hands-free feature. A blue LED is used to indicate whether the speakerphone is active or not.
	Hold key	The Hold key is used to place the active call on hold. A blue LED is used to indicate whether the call is on hold or not.
	()・) Transfer key	The Transfer key is used to transfer a call to another IP phone.
	Redial key	The Redial key dials the last dialed number automatically.
	Message key	The Message key is used to access the Voice Mail System for message retrieval.
	Conference key	The Conference key is used to place multiple calls on the phone into a conference on the phone.
	Phonebook key	The Phonebook key is used to enter into the Phone Book menu to call, add, edit or delete a contact in a selected phonebook.
12	Hands-free MIC	Hands-free MIC hole.  Note: You can use a paperclip to pick up the programmable key plate label (overlay) right here.
13	○ OK ▷ □ □ □ □ □ □ □ □ □ □ □ □ □ □ □ □ □ □	The Navigation Control Keys are used for navigating the menus on the phone; menu items are displayed on the LCD screen.
14	OK Navigator OK Key	The OK button is used to confirm and save a setting on the phone or to dial a phone number.
15	Programmable key plate label	The programmable key plate label is plastic. For best results, we recommend using a fine permanent marker (such as Sharpie brand) to write on it. It can be cleaned with industrial alcohol.
	key plate label	<u>Note:</u> Custom templates and programming software for MOCET phones are available from DESI (see, <a href="http://labels.desi.com">http://labels.desi.com</a> )

## 1.5. IP3032 Port Functions

The back side view and the connectors of the IP3032 are as follows:

No	Part Name	Description
1	Handset Connector	RJ-9 Jack for connecting handset cord.
2	LAN Port	RJ-45 Jack 100/10Mbps Ethernet port for connecting to the local area network (LAN). This port can support power over Ethernet (PoE) if the LAN switch provides it.
3	Power Jack	If a power source from adaptor is required, please order the IP3032 power adapter separately from MOCET.

Their locations are shown in the figure below:



## 1.6. IP3032 LED Functions

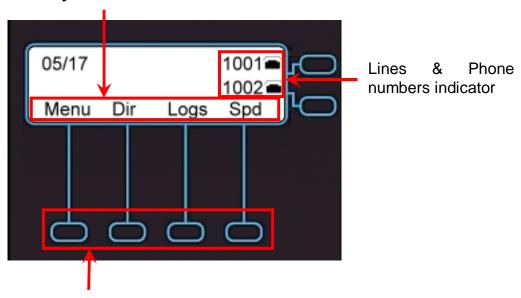
The following table describes all functions of the LED indicators:

LED	Color	Status	Description
		Off	No new message(s).
Message LED	Blue	Steady	Phone is booting or upgrading.
		Blinking slow	New message(s) indication.
		Steady	Feature is set to "ON – Active". Or Phone is busy.
668833	Blue	Blinking slow	Incoming call notification.
Programmable LED		Off	Feature is set "Off-inactive".
	Blue	Steady	Line is active, dialing or in a call.
Line LED		Blinking slow	Call is on hold.
		Flashing faster	Incoming call.
Hold LED	Blue	Off	No call is on hold.
Hold LED	Diue	Blinking slow	Call is placed on hold.
	_	Off	Speakerphone is not in use.
Speakerphone LED	Blue	Steady	On-hook dialing or hands-free mode.
MUTE LED	Blue	Off	Microphone is active.
MOTE LED	Diue	Steady	Microphone is inactive.
	Blue	Off	The line 1, 2 status is displayed on LCD.
Line Group LED	Diue	Steady	The line 3, 4 status is displayed on LCD.

#### 1.7. LCD Screen Indicators

The following figure shows the standard dispaly of the LCD. There are four soft keys associated with the operation of LCD display. For different menu or status items, the display items will change accordingly.

#### Soft-key menu:



#### Soft-key buttons:

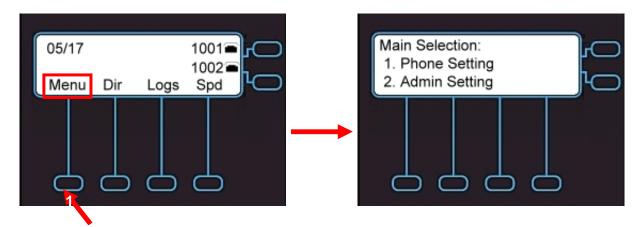
No	Line Status Icon	Description
1		The line is registered to the SIP server or service.
2	<b>A</b> x	The line is not registered to the SIP server or service.
3		The line is registered to the SIP server or service and is in use.
4	$\leftarrow$	The line is set for "Call Forwarding" to the configured number.

## 2. Getting Started

#### 2.1. Customizing Your IP Phone from Menu

You can customize your IP phone by adjusting the settings including display contrast, ring type, device volume adjustment, call settings, and add, edit or delete contacts in the phonebooks of the IP3032 using the on-screen menus, buttons and navigation keys.

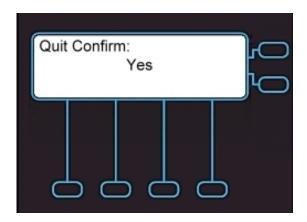
To configure your IP Phone from the menu, you can press **Menu** soft-key under the LCD screen.



Press this key

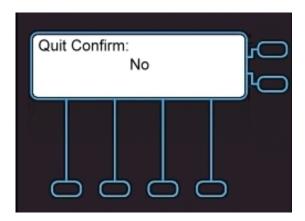
You can navigate through the menu with the navigation keys. The following sections will describe how you can setup your IP Phone through this menu. If you require additional information or assistance with your new phone, contact your system administrator.

<u>Note:</u> Changes to the phone settings will only become active after you exit from the menu and confirm the changes. The LCD will show the following screen to let you confirm your changes and exit the menu.



If you press **Navigation OK**, phone will save the settings.

If you press **Navigation Up and Down**, phone will change the screen to the following screen:



You will exit the screen and can continue to configure your phone.

#### 2.2. Configuring Basic Settings

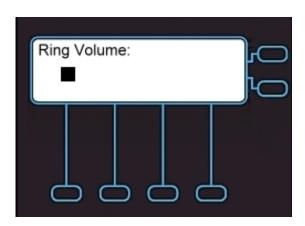
#### 2.2.1. Volume Setting

You can configure following volume settings:

- Ring Volume
- Handset Speaker
- Hand-free Speaker
- Handset MIC
- Hand-free MIC

#### To configure volume:

- 1. Press "Menu".
- 2. Select **Phone Setting > Volume Setting**. And select which device volume you want to set. Select from Ring Volume, Handset Speaker, Hand-free Speaker, Handset MIC, Hand-free MIC.



- 3. Use the **Navigation Up** and **Down** or **Volume Up** and **Down** keys to change the volume levels.
- 4. Press the **Navigation OK** key to confirm the change and exit volume change screen.

#### 2.2.2. LCD Brightness

To configure the brightness of the LCD to a comfortable level:

- Press "Menu".
- 2. Select Phone Setting > Backlight > Brightness.
- 3. Press **Navigation Up** and **Down** keys to increase or decrease the display brightness.
- 4. Press the **Navigation OK** key to confirm the changes and exit the menu.

#### 2.2.3. LCD Contrast

To configure the contrast of the LCD to a comfortable level:

- 1. Press "Menu".
- 2. Select Phone Setting > Contrast.
- Press Navigation UP and Navigation Down keys to increase or decrease the display contrast.
- 4. Press Navigation OK key to confirm the changes and exit the menu.

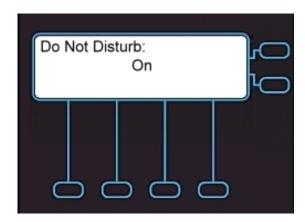
#### 2.2.4. Call Setting

You can configure following call features:

- Do Not Disturb (DND)
- Auto Answer
- Call Waiting

To configure those features:

- 1. Press "Menu".
- Select the Phone Setting > Call Setting: then select which feature you want to change. Select from DND, Auto Answer, or Call Waiting. Following screen is DND setting screen.



- 3. Use the **Navigation Up** and **Down** keys to select **On** or **Off**.
- 4. Press the **Navigation OK** key to confirm the changes and exit the menu.

#### 2.2.5. Lock Your Phone

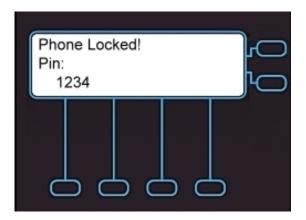
Sometimes you don't want other people to use your IP Phone. You can lock your phone and configure a personal identification key (PIN) to unlock the phone.

To lock your phone:

- 1. Press "Menu".
- 2. Select Phone Setting > Phone Lock.
- 3. Press the Navigation Up and Down keys to select On or Off.
- 4. Press Navigation OK key to confirm the change and exit the menu

You will see your phone is locked.

Following is a locked phone screen.



#### 2.2.6. Speed Dialing Setting

The Speed Dialing feature let you store up to 10 phone numbers that you can access easily using a single digit Speed Dial number from 0 to 9.

To configure a speed dial number:

- Press the "Spd" sofk-key below the LCD screen, or select Menu > Phone Setting > Speed Dialing.
- 2. Select the speed dial number to configure (from 0 to 9). And press the **Navigation OK** or "**Edit**" soft-key to edit the setting, and enter the phone number.
- 3. Press the **Navigation OK** key to confirm the changes and exit the menu.

#### 2.2.7. Reboot Your phone

Sometimes you will need to reboot your phone to apply new settings.

To reboot your phone:

- 1. Press "Menu".
- 2. Use the **Navigation Up** and **Down** keys to select **Reboot**.
- 3. Press **Navigation OK** to reboot your phone. Phone will show a "**Reboot**" message and the phone will restart.

**Note:** The IP3032 takes about 2 minutes to startup after rebooting; there will be a series of diagnostic lights on the phone during the process. If your phone does not become operational within 2 minutes, contact your administrator or Service Provider support line.

#### 2.3. Managing Contacts

The IP3032 supports four different contact lists:

- All Contacts
- LDAP Directory
- Public Contacts
- Private Contacts

You can store up to 100 **Private Contacts** in your phone's directory, and you can add, edit, delete, dial, or search for a contact in this directory.

**Public Contacts** are provided for you by the administrator or Service Provider. They can be viewed, and used, but not changed or deleted. For more information, please contact your administrator or Service Provider.

The IP3032 can also access a local or remote **LDAP Directory**. Please contact your administrator or Service Provider for more information on how to enable, configure or access this feature.

The **All** directory lets you search and dial a contact from all the configured contacts databases.

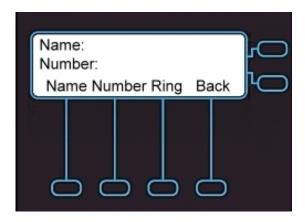
There are three ways to enter your phone's directory:

- Press the "Phonebook" hot key.
- Press "**Dir**" soft-key.
- Navigate to Menu > Phone Setting > Phonebook.

#### 2.3.1. Adding Contacts

To add a new contact:

- 1. Go to the **Phonebook** > **Private**.
- 2. Press the "Add" soft-key.
- 3. Enter "Name", "Number", and "Ring", from the key pad.



4. Press the "Save" soft-key when you are finished.

#### 2.3.2. Editing Contacts

To edit your contact:

1. Use the **Navigation Up** and **Down** keys to select the contact your want to edit.

- 2. Press the "Edit" soft-key.
- 3. Use the **Navigation Up** and **Down** keys to select the fields you want to change.
- 4. When finished making changes, press the "Save" soft-key to save the changes.

#### 2.3.3. Deleting Contacts

To delete your contact:

- 1. Use the **Navigation Up** and **Down** keys to select the contact your want to delete.
- Press "Del" soft-key.

The contact is deleted from the directory.

#### 2.3.4. Searching for a Contact

To search for a particular contact:

- 1. Go to **Phonebook** > **Private**.
- 2. Press the "**Search**" soft-key.
- 3. Input the first character of the contact's name and the IP3032 will move to the first contact matching the character.
- 4. Using the **Navigation Up** and **Down** keys, choose the desired contact.

#### 2.3.5. Placing a Call to a Contact

To dial from a directory:

- 1. Navigate to a contact (private, public or LDAP).
- 2. Press the "**Dial**" soft-key.

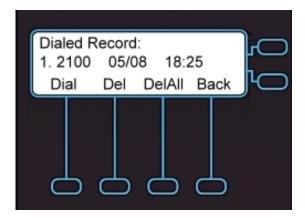
The phone will dial the number selected.

#### 2.4. Managing Call Logs

In the IP3032, there three call logs:

- Dialed Calls
- Missed Calls
- Answered Calls

The call logs save the last 30 numbers dialed, missed or answered. For example, the following is the Dialed Calls list screen.



From the call log lists, you can view, delete, and dial.

To manage the selected call log:

- Use the **Dial** soft-key to redial the call.
- Use the **Del** soft-key to delete the selected item.
- Use the **DeIAII** soft-key to delete all items in the list.
- Use the **Back** soft-key to exit the call list screen.

#### 2.5. Viewing Your Phone's Information

Using the menu of the IP3032, you can check the IP3032 hardware version, firmware version, and network status.

To view this information:

- 1. Press the "Menu".
- Select "Information".
- 3. Use the **Navigation Up** and **Down** keys to view the information.

#### 2.6. Configuring Programmable Keys

The IP3032 has 8 programmable keys, which can be configured for different features. From the Menu, you can view the settings for each key.

To see the setting status:

- 1. Press "Menu".
- 2. Select the "Program Key".
- 3. Select the FUNC#1-FUNC#8 to see their settings.

#### 2.7. Managing Voice Mail

Some hosted SIP services and IP PBXs provide a message waiting indicator (MWI) to alert you to any new voicemail on your phone. Your phone can receive MWI information from supported services and servers.

To view your voice mail status:

- 1. Press "Menu".
- 2. Select "Voice Mail".

You can see the voice mail count for your accounts.

To get more information on MWI status:

- 1. Press Menu > Voice Mail.
- 2. Select the account you want to check and press the **Navigation OK** key.

#### 2.8. Managing Instant Message

The IP3032 supports SIP SIMPLE Instant Messaging (IM). With a supported service or server, you can send and receive IM right from your phone using the on-screen dialpad; there is also a way to configure some useful Message Templates, which are pre-made messages for common requests or responses, to simplify use of IM on the IP3032.

There is a message icon and a popup message on the LCD screen when there is a new message received.

To view the messages, you can:

1. Press the Navigation OK key.

The messages are listed for you to view.

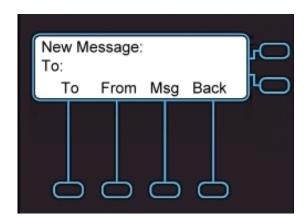
Or

- 1. Press "Menu".
- 2. Select Message > Instant Messaging > Simple > Inbox.

The messages are listed for you to view.

To create a new message:

- 1. Press "Menu".
- 2. Select Message > Instant Messaging > Simple > New Message.
- 3. Enter "To", "From", and "Msg", from the key pad.



To delete a message:

- 1. Press "Menu".
- Select Message > Instant Message > Simple > Delete Messages.
- Select what data you want to delete, you can select Delete Inbox Messages,
   Delete Outbox Messages or Delete All Messages.

To edit a Message Template:

- 1. Press "Menu".
- 2. Select Message > Instant Message > Simple > Text Templates.

The IP3032 can save and use up to 30 Message Templates.

#### 2.9. Viewing Notifications

The IP3032 supports Broadsoft Server Notifications. With this feature, administrators or service providers can send messages to users connected to a Broadsoft server. For more information, please contact your administrator or Service Provider.

## 3. Using Basic Features

The IP3032 Standard IP Deskphone is designed to be used like a regular phone on a public switched telephone network (PSTN). You can place calls, transfer a call to someone else, or conduct a conference call.

The IP3032 supports up to four simultaneous calls. However, only one of these calls can be active at a time. The active call is the one you are using to speak or listen to. The inactive calls can be:

- On hold
- Joined in a conference (with up to three callers)
- "Incoming call" or "Ringing"

This chapter provides basic operating instructions for the phone including:

- Placing a Call
- Placing an urgent call
- Answering a Call
- Answering an urgent call
- Ending a Call
- Redialing a Number
- Putting a Call on Hold
- Setting Up a Conference
- Transferring a Call
- Forwarding Calls
- Enabling or Disabling Do Not Disturb

#### 3.1. Common Terms

To use the IP3032, you need to follow some conventions that we will mention in this guide. In the following descriptions, we will introduce some common terms for your understanding.

#### 3.1.1. Lines

A "Line" in this guide represents how many phone numbers supported in one phone. For example, the IP3032 can support up to four lines configured for the same or different service providers or servers and even make up to four concurrent calls. Therefore, the IP3032 is said to support **multiple line appearances**.

#### 3.1.2. Calls

A "Call" in this guide represents how many simultaneous connections can be made to a single phone number. Each line of service in the IP3032 can support up to two calls simultaneously. This allows the user to place one caller on hold and talk to the other person in another call on the same Line. Therefore, the IP3032 phone is said to support **multiple call appearances**.

#### 3.1.3. Register to a server

The IP3032 must be configured before it can perform some basic functions. Although the phone can make a peer-to-peer SIP call by dialing the other user's IP address (i.e., 192.168.0.12) directly on the keypad, this is inconvenient and it is hard to remember all the IP addresses of phones on the system. This is why a hosted SIP server or local IP PBX implements a Registrar service, which allows the connected phones to find and dial each other more easily by extension numbers or names instead of IP addresses.

In addition, the hosted SIP server or IP PBX helps keep track of active phones, their IP addresses for routing calls, as well as keeping track of which phones are busy or idle. Most hosted SIP services and IP PBXs support direct system service (DSS) as well as showing the busy lamp field (BLF) status of system phones. These allow you to directly dial another extension and can show you whether the other phone is on a call or not.

Usually, the hosted SIP service or IP PBX will also use the Registrar service to send messages to your phone to alert you that you have voicemail. This is called message waiting indication (MWI).

Refer to Configuration through Menus or Configuration through Web for detail information of how to configure the phone to register to server.

#### 3.1.4. Caller ID

When you receive a call, the caller's phone number is shown on the screen, if the caller hasn't chosen to hide his number and if the network supports the Caller ID feature. The IP Phone can display both the Caller ID (CLID) and the Caller Name (CNAM) of the caller if it is available. Not all services and servers support CLID and CNAM. For more information about this topic, please contact your service provider or system administrator.

#### 3.2. Installing Your IP Phone

Before placing the phone into operation, either you or your administrator or Service Provider has to install the phone on your network. Please contact your administrator or Service Provider for more details, or refer to the included hardware installation guide.

#### 3.3. Configuring Your IP Phone for Service

The IP3032 must be configured for the hosted SIP service or IP PBX before operation. The phone is usually pre-configured by the administrator or hosted SIP service provider. For an advanced or experienced user, you may refer to the IP3032 Administrators' Guide for full information on how to configure all the settings of the IP3032.

#### 3.4. Line Selection

The IP3032 can support up to 4 lines.

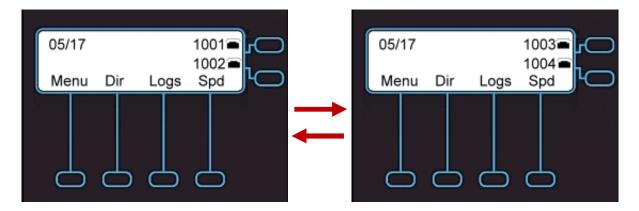
To select a line you can:

1. Use the preferred line setting. That means when you pick up handset, or press the speakerphone key, or press headset key when not on a call, the phone will

automatically use the preferred line for the call. If the preferred line is not the line you want to use, you can press **Line Key** buttons ( ) on the right hand side of the LCD to switch to the desired line.

2. Press the **Line Key** buttons ( ), which will connect to the chosen line.

<u>Note:</u> Because the IP3032 LCD displays only two phone lines at a time, to access the second two lines, either use the **Navigation Up** or **Down** keys, or press the **Line Group** button ( ) on the phone to switch between them.



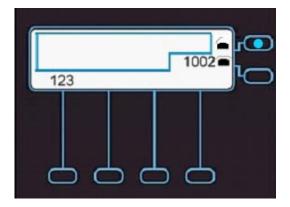
#### 3.5. Placing a Call

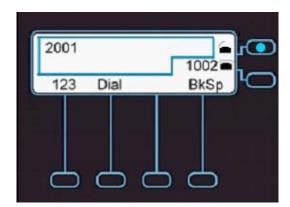
You can place a call in many ways from the IP3032:

- Using the handset
- Using the speakerphone
- Pressing a line key

You can also dial the number first and then choose the method you will use to dial the other party. This is called pre-dialing. During a call, you can alternate between using the handset or speakerphone modes by pressing the speakerphone key( ), or picking up the handset while on a speakerphone call. The call duration on an active call is shown on the LCD during the call.

To place a new call:



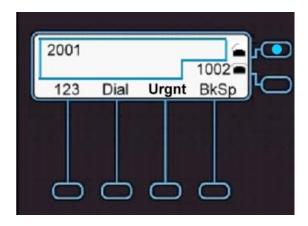


Operation	Description		
	or Line Key	<ul> <li>1. Pick-up the handset or press a line key or press the speaker key.</li> <li>→ You will hear a dial tone.</li> </ul>	
Making a call	1 as 2 as 3 as 4 as 5 as 6 as 7 as 8 as 9 as 7 as 8 as 9	<ul> <li>2. Use the keypad to dial the phone number.</li> <li>→ The LCD window displays the number you enter.</li> <li>Note: You may use the "BkSp" soft-key to delete the last digit.</li> </ul>	
	<b>\$</b> < )	3. On-hook the handset when your conversation is over.	

#### 3.6. Placing an Urgent Call

To place an urgent call, you need to enable Urgent Call feature in the Web User Interface first. Please refer to the **Section 4.4. Configuring Phone Settings** to know how to configure it.

If your IP phone is enabled "Urgent Call", when you dial a number, there is an "Urgnt" softkey displayed on the LCD screen.



If you press the "Urgnt" softkey to dial the number out, the call will be marked as an urgent call. The called party can neither deny nor block the call even the called party's phone is on DND.

#### 3.7. Adjusting Call Volume

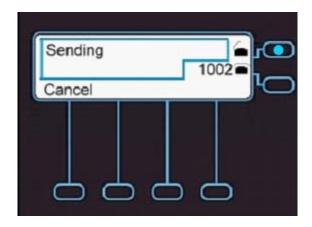
During a conversation, if the voice volume is too low or too high, you may adjust it.

To adjust volume while on a call:

Operation	Description
Adjusting Call Volume	Press the Volume control key to adjust the volume.

## 3.8. Canceling a Call

After you dial a call, but the called party has not answered, you can Cancel the call.

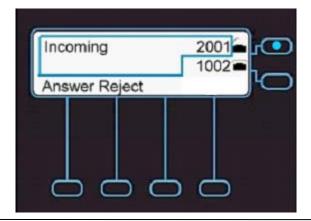


Operation	Description	
Canceling a call		Press "Cancel" soft-key to reject the call.
	"Cancel"	If only the calling call on the phone, you can just press speakerphone or put down handset to cancel the call.

## 3.9. Answering a Call

When there is a new incoming call on the phone, you can answer a call by:

- Using the handset
- Using the speakerphone
- Pressing "Answer"



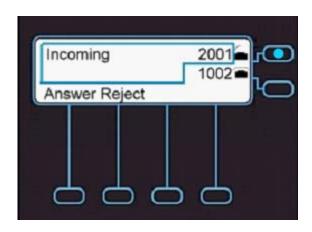
Operation	Description	
Answering a call	or or Answer	If there is no other call, just pick-up the handset or speakerphone or press the "Answer" soft-key upon hearing the phone ringing.  If there is another call, press "Answer" soft-key to accept the call, the previous call will be placed on hold automatically.

#### 3.10. Answering an Urgent Call

When there is an incoming urgent call on the phone, you can answer a call as you would normally; however, some features will be ignored automatically including DND and call blocking.

#### 3.11.Rejecting a Call

When there is an incoming call on the phone, you can reject the call.



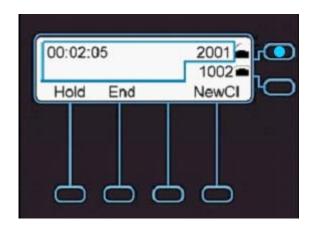
Operation	Description	
Rejecting a call	"Reject"	Press "Reject" soft-key to reject the call.  After pressing the "Reject" soft-key, the call will be dropped directly. The record will be logged in answered call list. Other call features are not affected by this operation.

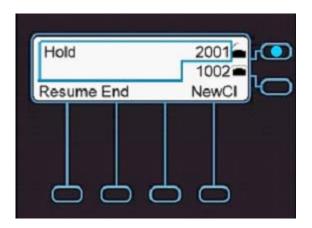
## 3.12. Ending and Holding and Resuming a Call

When a call is in connected state, you can end a call by:

- Using the handset
- Using the speakerphone

#### Pressing "EndCall" soft-key





Operation	Description	
Ending a call	or or EndCall	If there is only one call, just place the handset back into the cradle, press speakerphone or press "EndCall" to end the call.  If there is another call, press the "EndCall" soft-key to end the call.
Holding a call	or Hold	Press the "Hold" soft-key or <b>Hold</b> key on the phone to place an active call on hold.
Resuming a call	or Resume	Press the "Resume" soft-key or <b>Hold</b> key on the phone again to place an active call on hold.

#### 3.13. Muting and Un-muting a Call

While in a conversation, you may mute the microphone by pressing the **MUTE** ( button. The LED of the button will become red. When muted, the other caller will not hear anything from your phone.

Pressing the **MUTE** ( key again will Un-Mute the phone.

#### 3.14. Redialing a Number

To redial the last numbers you dialed:

- 1. Press **Redial**( ) hot key.
- 2. Phone will enter "Dialed Calls" list.
- 3. Select the number you want to redial.
- 4. Press "Dial" soft key to dial out.

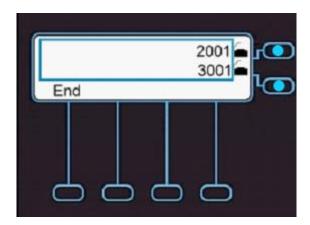
For more information, please see 2.4. Managing Call Logs.

#### 3.15. Setting up a Conference Call

The IP3032 can support a 3 party conference call.

To set up a conference:

- 1. Call the first party and Hold the call
- Press "NewCall" soft key
- 3. Call another party
- 4. Press the Conference ( ) hot key to set up the 3-party conference



#### 3.16.Transferring a Call

A call can be transferred in one of three ways:

**Blind Transfer**: The call is automatically transferred after you dial the number of the party to whom you want to transfer the call.

**Semi-Attended transfer**: The party to whom you want to transfer the call does not answer their phone before you transfer the call (when you hear ring-back tone).

**Attended transfer**: The party to whom you want to transfer the call answers their phone before you transfer the call. You can consult with them before completing the transfer.

#### 3.16.1. Blind Transfer

To complete a blind transfer:

- During an active call, press the Transfer ( ) hot key.
- 2. The active call is placed on hold, and an entry screen is displayed for the number you want to transfer to.
- 3. Press **Navigation OK** or "**Send**" soft-key to complete the transfer. If you want to cancel transfer operation, just press the "**Back**" soft-key.

#### 3.16.2. Semi-Attended Transfer

To do Semi-Attended transfer:

1. During an active call, press "**NewCall**" soft-key to call another party.

2. When you hear the ring-back tone, press the **Transfer** ( ) hot key.

#### 3.16.3. Attended Transfer

To do Attendant transfer:

- 1. During an active call, press "NewCall" soft-key to call another party.
- 2. Wait for the other party to answer the call.
- 3. Press the Transfer ( ) hot key.

Note: Sometimes, you may place more than 2 calls on your phone, when you press

Transfer ( ) hot key, phone will show a call list to let you select a call to finish the transfer operation.

#### 3.17. Forwarding a Call

You can configure your phone to forward your incoming call to another party, which is sometimes called as static forwarding. You can also dynamically forward calls while your phone is ringing.

There are three types of static forwarding:

- Unconditional
- No answer
- Busy

For more information on how to setup static call forwarding, please contact your administrator or Service Provider support center.

To use dynamic forwarding:

- 1. When the phone rings with an incoming call, press the "Divert" soft key.
- 2. Enter a number to forward the incoming call to.
- Press OK to forward the call.

## 3.18. Using Voice Mail

Your voicemail is saved on either your hosted SIP service or on the IP PBX, but you can access it from the IP3032 using the Voicemail button.

The presence of new voice mail messages is indicated by a flashing message waiting indicator (MWI) LED on the front of the phone and an icon is shown on LCD.

**Note:** Voicemail is an optional feature configured on a hosted SIP service or IP PBX and may not be supported on your particular system. To use voicemail on the IP3032, some settings need to be configured first. For more information please contact your administrator or Service Provider support center.

To listen your voice messages:

1. Press the Message ( ) hot key to display the mailbox list.

- 2. Select which SIP line's mailbox you want to listen to and press "**OK**". A call will be placed to your voice mail server.
- 3. Follow the interactive voice response (IVR) instructions to retrieve and listen, forward or delete your messages.

#### 3.19. Placing a Speed Dial Call

To configure a speed dial number, please see 2.2.6 Speed Dialing Setting.

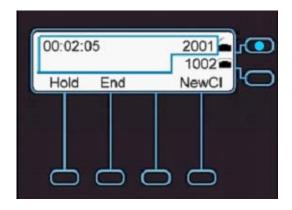
To make a speed dial call:

- 1. Pickup the handset, or press a line key, or press the speaker key or press the headset key.
- 2. When you hear dial tone, dial the one digit (0-9) speed dial key you previously configured.
- 3. Press "**OK**" or the "**Dial**" soft-key and the phone will look up the key in the speed dial table, find the matching item and dial the number assigned to the speed dial key.

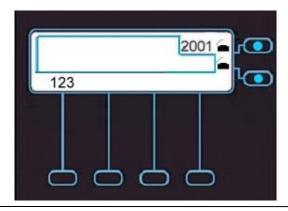
#### 3.20. Using Multiple Lines

IP3032 can support up to four lines configured for the same or different service providers or servers. It allows you to make up to four concurrent calls.

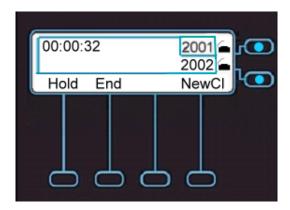
For example, the following screen shows you line1001 has one active call with number 2001.



You can make a new call by using another line, 1002. Press the **Line Key** button, 1002, on the right hand side of the LCD, and then make a call to number 2002.



After the called party answers the call, the following screen shows you line 1001 has one held call with number 2001, and line 1002 has one active call with number 2002.



To switch between the two calls, press **Hold** soft-key to hold the line 1002 which calls to number 2002 first. Then press the **Line Key** button, 1001, on the right hand side of the LCD, which calls to number 2001 to unhold the call.

By the steps as above, you can make up to four concurrent calls.

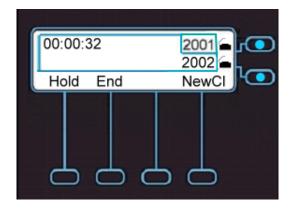
### 3.21. Using Multiple Calls

Each line of service in the IP3032 can support up to four calls simultaneously. It allows you to place one caller on hold and talk to the other person in another call on the same line.

For example, the following screen shows you line 1001 has one active call with number 2001.

You can make a new call on the same line, just press **NewCI** soft-key to call another party.

After the called party answers the call, the following screen shows you line 1001 has two calls, the call to number 2001 is on hold, and the call to number 2002 is active.



By the steps as above, you can use one line to make 4 calls simultaneously.

### 4. Using Advanced Features

In the **Getting Started** chapter, this guide describes the steps used to customize the IP3032 from the LCD menu of the phone. However, there are many advanced features that need to be set up using the phone's web user interface. This chapter provides instructions on how use these features.

### 4.1. Login Web UI

- To use your web browser to configure the IP3032, you need to know the IP address
  of the phone on your local area network (LAN). Using the LCD menu, find the IP
  address of the phone under Menu > Information > IP Address.
- Point your web browser to the IP address of the IP3032.
- 3. Enter the correct Username and Password information into the dialog box and press OK.



**Note:** The default Username is "user" and the default Password is "1111" and the default network settings are:

Default IP address: 192.168.1.10
Default Subnet mask: 255.255.255.0
Default Gateway: 192.168.1.1
Default DNS: 8.8.8.8

### 4.2. Changing the User's Password

Some features of the IP3032 require you to enter a personal identification number (PIN) for security. The default PIN of the IP3032 User account is "1111".

To change the User password:

- 1. Navigate to the **System** tab
- 2. Locate User **Settings > User Password**:
- Change the password to use a new PIN.

### 4.3. Viewing Phone Information on Web User Interface

**Information** is the first page you will see when you login to the phone's web user interface. This page lets you check the status of the IP Phone including network and product related information as well as account status information.



For SIP Line status:

- When this line is disabled, the line status will not be displayed.
- "Registration State" will be marked "Registered" with green text.
- "Unregistered" will be marked with red text.

### 4.4. Configuring Phone Settings

Phone Settings enables you to configure private settings for phone. Select the **Phone** tab of the web user interface to begin configuration.

From this page you can change following settings:

- Device Volume
- Tone and Ring types
- Voice parameters
- Advanced phone features (DND, Auto-answer, Call Waiting, etc)
- Speed dial settings
- Programmable key settings

Here is the description of each field.

Field Name	Function
Handset Mic	Set the input level of handset microphone.
Handset Speaker	Set the output level of handset speaker.
Speaker Mic	Set the input level of hand-free microphone.

Speakerphone	Set the output level of hand-free speaker.	
Ring Tone Volume	Set the output level of ring.	
Side Tone Volume	Set the output level of side tone.	
Tone Type	Select the tone type. There are 11 types of standard Tone for selection.	
Ring Type  Select the ring type. There are 11 types of standard Ring for selections.		
Preferred Line Select the SIP line automatically to dial out if not proline manually when dialing.		
<b>Enable Music on Hold</b>	Plays hold music on phone when held by remote.	
Enable Auto Answer	Turn on auto answer function.	
Enable DND	Turn on DND function, block any incoming all	
Enable Call Waiting	Turn on call waiting function.	
<b>Enable Call Waiting Tone</b>	Play call waiting tone when there's another incoming call.	
Enable Hold Reminder	Turn on the Hold Reminder function.	
Hold Reminder Time (sec) Set the time (10~60 sec) that phone will remind us tone when the call hold remote every this time peri		
Hotline Number Set the hotline number		
Hotline Timeout(sec)	Set the time (1~60 sec) when phone off hook but not dial, phone will dial out the hotline number automatically after this time period.	
Dial Timeout (sec)	Set the time (1~30 sec) that phone will dial out automatically after this time period.	
Enable Phone Lock	Lock the phone.	
Enable Urgent Call	Enable Urgent Call to add urgent call option in dialing screen.	
Phone Number Display style	Set the display style for calling name and number.	
RTP Port Base	Set phone local RTP Base Port.	
Speed Dialing Entry 0~9	There are 10 speed dial entries for user to store. Each entry map to the numeric keypad on the IP Phone.	
Programmable Key Setting 1~8	Please refer to Using Programmable Keys in the Administrator Guide.	

### 4.5. Managing Programmable Keys

Your IP3032 Phone has 8 programmable keys which can be configured for different features.

To use the programmable keys, just press the key, then the configured feature will be

executed, some features will blink the LEDs or turn on the related features.

To configure programmable keys, go to **Phone** tab of the web user interface, and find the "**Programmable Key Settings**" section:

Option Name	Function	
Line Supported	Select which line supports the feature (Do Not Disturb, Lock IP-Phone, Auto Answer have no this item).	
Do Not Disturb Enable/disable DND feature.		
Lock IP-Phone	Lock your phone.	
Abbreviated Codes Speed dial the codes.		
Busy Forward Set static forwarding number when busy.		
Always Forward	Set static forwarding number unconditional.	
No answer forward	Set static forwarding number when no answer.	
DND Forward	Set forwarding number when DND feature is enabled.	
Auto Answer	Enable/disable auto-answer feature.	
Caller Blocking	Set a caller blocking item.	
BLF	Set number for Busy lamp field feature. Contact administrator for help.	
Call Park  Set call park number, PBX related feature. Contact administrator for help.		
SLA	Set SLA number. PBX related feature. Contact administrator for help.	
Asterisk Parking Number	Set Asterisk Park number, PBX related feature. Contact administrator for help.	
Asterisk Parking Slot	Set Asterisk Park Slot number, PBX related feature. Contact administrator for help.	
Intercom	Set Intercom number, PBX related feature. Contact administrator for help.	
Feature Code	Set speed dial for feature codes.	
DTMF	Set DTMFs for call.	
Save Settings	Save changes in this page to the phone.	
Cancel	Discard all changes in this page.	

### 4.6. Configuring SIP Line Settings

On the **SIP** tab of the web user interface, you can change various SIP settings for each Line.

A user can block specific calls by Caller ID (CLID). For each SIP Line, you can configure up to 10 blocking entries.

Field Name	Function	
Always Call forwarding	Enable/disable always call forwarding	
Busy Call forwarding	Enable/disable busy call forwarding	
No-Answer call forwarding	Enable/disable no-answer call forwarding	
DND call forwarding	Enable/disable DND call forwarding	
No answer timeout	Set no answer timeout time (seconds)	
Enable block anonymous call	Enable/disable block anonymous call feature.	
Always forward Destination	Set always forward number	
Busy forward Destination	Set busy forward number	
No-Answer forward Destination	Set no-answer forward number	
DND forward Destination	Set DND forward number	
Caller Blocking Entry 0~9	Set caller blocking numbers	
Save Settings	Save changes in this page to the phone.	
Cancel	Discard all changes in this page.	

### 4.7. Configuring System Settings

On the **System** tab, the user can access or change:

- Password
- Time settings
- Debug logs
- Reboot the phone

Field Name	Function
User Name	The user username. It is fixed, and can't be changed.
User Password	The user password
Auto DST	Enable Automatic daylight saving time flag
Starts on	Set DST start Month
Starts on	Set DST start Day
Starts on	Set DST start Time
Starts end	Set DST end Month

Starts end	Set DST end Day	
Starts end	Set DST end Time	
Time Format	Set LCD date/time display modes	
Time zone	Set Phone Time Zone.	
System Log Download	Click to download System Log	
SIP Log Download	Click to download SIP Log	
Reboot	Click to reboot IP Phone	
Save Settings	Save changes in this page to the phone.	
Cancel	Discard all changes in this page.	

### 4.8. Managing Phonebook

On the **Phonebook** tab, the user can manage his phone directories.

### 4.8.1. Private Phonebook

For the Private Phonebook, the user can add, edit, delete or dial an entry directly from the web page. The user can also upload his contacts from a file using a web browser. The Private Phonebook can have up to 100 contacts.

### 4.8.2. Public Phonebook

For the Public Phonebook, a user or administrator can upload a file using a web browser.

### 4.8.3. LDAP Directory

The LDAP Directory can be configured using a web browser; please contact your administrator or Service Provider support desk for more information.

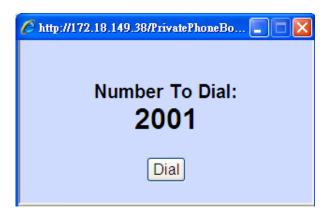
Field Name	Function
Number Guessing Option	Set number guessing feature data source.
LDAP Security	Set LDAP security type, SSL2, SSL3 or TLSv1.
LDAP Server Address	Set LDAP server address.
LDAP Server Port	Set LDAP Server port.
LDAP Login Name	Set LDAP user name.
LDAP Login Password	Set LDAP user password.
LDAP Base	Set LDAP Search Base.
LDAP Max. Hits	Set retrieve record counts from Server one time.
LDAP Name Filter	Set Name guessing filed in LDAP server.
LDAP Number Filter	Set Number guessing filed in LDAP server.

Test Configuration	Click to test if the LDAP server available. Show "Error" in red when unavailable, "Available" in green when available,
	"Not Configured" in black when hasn't set select LDAP.

### 4.9. Placing a Call from the Private Phonebook

There is a "Dial" button for each contact entry under Pricvate Phonebook tab, you can press it and place a call to the contact.

A new window will show up when you click the "Dial" button,



If you click "Dial" to place the call, when the number dials out, the window will close.

### 4.10.Sending IM

On the **Phonebook** tab, the user can send instant messages.

To send an instant message:

- 1. Select a registered line that you want to use to send the IM.
- 2. Enter a text message.
- 3. Enter a contact's number.
- 4. Press "Send" button to send the instant message out.

### 5. Using Advanced Call Features

### 5.1. Centralized Conferencing (for Broadsoft PBX)

The IP3032 allows you to create multiple conferences with unlimited participants (depending on IP PBX limitations) when your Administrator enables Centralized Conferencing.

The steps to make centralized conferencing available are different for each supported IP PBX or hosted SIP service. On a Broadsoft softswitch, you can initiate a Server-based conference by pressing "Conf" soft key on LCD.

<u>Note:</u> The "Conf " soft-key will show only you have set the "Conference Number" on SIP Web page.

#### To create a Conference:

- 1. Place some calls with the participants.
- 2. Press "NewCall" to enter dial screen, and a "Conf" soft-key will show up.
- 3. Press "Conf" soft-key and a conference will be created.

All participants are added to the conference automatically.

#### To add new participant to the conference:

- 1. Press "NewCall", and dial the participant you want to add.
- 2. There will be a "**Conf**" soft-key when the call is answered by the other participant.
- 3. Press the "Conf" soft-key and the new participant will be added to the conference.

### 5.2. Call Waiting

The call waiting feature notifies the user on an active call of a new incoming call. You can disable the call waiting feature, so that the new incoming call is automatically rejected by the phone with a busy message.

If you disable call waiting on the phone, and the user is on a call, any further incoming calls receive busy unless "Call Forward Busy" is configured on the phone, where it then forwards the call according to the configured destination.

### 5.3. Intercom Call

### 5.3.1. Answering an Intercom Call

By default, the IP phone allows incoming intercom calls to be automatically answered on your phone. If the intercom call comes into the phone while an active call is already present, the phone puts the active call on hold and answers the intercom call.

### 5.3.2. Placing an Intercom Call

The Intercom call feature depends on PBX/SIP Server as there are different intercom call codes for each servers; for example, the Asterisk default intercom code is "\*80".

To place intercom call on an Asterisk server, just make a call to "*80"+ [destination number] ("*801001" for example). For other phone systems or hosted SIP services, please contact your administrator or Service Provider for more information.			

## **6. Phone Firmware Upgrade**

The IP3032 supports several different ways to update its firmware (please refer to the table below). This is for system administrators or service providers. End users normally do not have to upgrade firmware by themselves. Please contact your Administrator or service provider for assistance.

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.	<b>√</b>	✓	<b>✓</b>
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.	×	✓	<b>✓</b>
3	Upgrade Using TFTP/FTP/HTTP/HTTPS Server	Software Patch (update.ip3032) and XML configuration file update via TFTP/FTP/HTTP/HTTPS server.	×	<b>√</b>	<b>✓</b>
4	Engineering Key Sequences on Root Menu	Download Software Patch (update.ip3032) from TFTP server.	×	<b>√</b>	<b>√</b>
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.	×	✓	<b>✓</b>
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.	×	×	<b>√</b>

# 7. Troubleshooting

Symptom	Check & Remedy	
No operation	<ul> <li>Check if the power adapter is properly connected.</li> <li>Check if the Ethernet cable is properly connected.</li> </ul>	
No dial tone	<ul> <li>Check if the handset cord is properly connected.</li> <li>Check if the power adapter is properly connected.</li> </ul>	
LAN connection lost status message	Check if the Ethernet cable is properly connected.	
Cannot make call	<ul> <li>Check the status of your SIP registration status or contact your administrator, supplier or ITSP for more information or assistance.</li> </ul>	
Cannot receive a call	<ul> <li>Check if the Ethernet cable is properly connected.</li> <li>Check the status of your SIP registration status, or contact your administrator, supplier or ITSP for more information or assistance.</li> </ul>	
Cannot connect to the IP3032 configuration pages with a browser	<ul> <li>Check if the Ethernet cable is properly connected.</li> <li>Check the IP address of the IP Phone.</li> <li>Check if your firewall/NAT settings is correct.</li> </ul>	

# 8. Glossary

8.1. Acronyms		
ANC	Anonymous Call	
APS	Auto-Provisioning Server	
B2BUA	Back to Back User Agent	
CA	Certificate Authority	
CID	Caller ID	
CODEC	Coder and Decoder of Voice	
CNG	Comfort Noise Generation	
CPC	Calling Party Control	
CPE	Customer Premises Equipment	
CWCID	Call Waiting Caller ID	
CWI	Call Waiting Indication	
CWT	Call Waiting Tone	
DHCP	Dynamic Host Configuration Protocol	
	Domain Name Server	
DNS		
DTMF	Dual Tone Multiple Frequency	
ETSI	European Telecommunication Standard	
FQDN	Fully Qualified Domain Name	
FSK	Frequency Shift Keying	
HTML	Hypertext Markup Language	
HTTP	Hypertext Transfer Protocol	
ICMP	Internet Control Message Protocol	
IETF	Internet Engineering Task Force	
IP .	Internet Protocol	
ISP	Internet Service Provider	
ITSP	IP Telephony Service Provider	
IVR	Interactive Voice Response	
LAN	Local Area Network	
LBR	Low Bit Rate	
LBRC	Low Bit Rate Codec	
MWI	Message Waiting Indication	
PoE	Power over Ethernet (IEEE802.3af standard)	
PPPoE	Point-to-point protocol over Ethernet (mainly for xDSL modem connection)	
PSTN	Public Switched Telephone Network	
NAT	Network Address Translation	
NTP	Network Time Protocol	
RTP	Real Time Protocol	
RTCP	Real Time Control Protocol	
SDP	Session Description Protocol	
SIP	Session Initiation Protocol	
TFTP	Trivial File Transfer Protocol	
TCP	Transmission Control Protocol	
UDP	User Datagram Protocol	
URL	Uniform Resource Locator	
VLAN	Virtual Local Area Network	
WAN	Wide Area Network	
XDSL	Digital Subscriber Loop, such as ADSL, VDSL, HDSL, etc.,	
XML	Extensible Markup Language	
- · · · · · · · · · · · · · · · · · · ·	1	

### 8.2. Terminology

40/400DAOE T	1
10/100BASE-T	It is a LAN transmission line specification stipulated by IEEE. Transmission speed is 10 or 100 Mbps and the modulation technique is base-band modulation. The cable uses unshielded twisted pair, similar to a telephone wire. 10BaseT is an IEEE standard (802.3) for operating 10 Mbps Ethernet networks (LANs) with twisted pair cabling and a wiring hub.
802.1p	An IEEE standard for providing QoS using three bits (defined in 802.1q) to allow switches to reorder packets based on priority level.
802.1q	An IEEE standard for providing virtual LAN (VLAN) identification and QoS levels. Three bits are used to allow eight priority levels, and 12 bits are used to identify up to 4,096 VLANs.
<u>AGC</u>	Automatic Gain Control is a feature of IP solutions units that allows the units to automatically adjust the incoming voice signal to a user defined level in dBm.
<u>A-law</u>	The PCM coding and compression standard used in Europe and in areas outside of North America influence. A Law Encoding is the method of encoding sampled audio waveforms used in 2.048Mbps, 30 channel PCM primary system known as E-carrier.
Auto answer	In telephone call control: The capability of a machine to answer a ringing telephone without human intervention.
Autodial	In telephone call control: An auxiliary device for a telephone that automatically dials any of a group of prerecorded telephone numbers.
Blocked Calls	Caused by an insufficient network facility that does not have enough lines to allow calls to reach a given destination. May also pertain to a call from an originating number that is blocked by the receiving telephone number.
Call Completion	The point at which a dialed number is answered.
Call Termination	The point at which a call is disconnected.
CDR	In telephone call control: Call Detail Record for billing
0211	This service is offered by the LEC to the end user. The feature-rich
IP Centrex	Centrex line offers the same features and benefits as a PBX to a customer without the capital investment or maintenance charges. The LEC charges a monthly fee to the customer, who must agree to sign a term agreement. And the abive service is thru VoIP called IP centrex.
CO	Switching center for the local exchange carrier.
CODEC	The CODEC (CODER/DECODER) is a standard through which voice information can be encoded into data or decoded back to voice information. Both a Coder and Decoder are necessary on both sides of the telephone call since telephone calls occur simultaneously in both directions. Bandwidth is an extremely important factor in QOS (Quality of Service). MOS (Mean Opinion Score) is an attempt to make a quantifiable benchmark of voice quality. Below are examples of the CODEC, bit rate and mean opinion score: G.711 (toll quality) 64K MOS=4.1, G.726 16K (32K) MOS =3.8, G.729AB (cell phone quality) 8K MOS=3.7.

C ot o mo o m	
Customer	The only part of the telecommunications system that the customer
Premise	comes into direct contact with. Example of such pieces of
Equipment	equipment are: telephones, key systems, PBXs, voicemail systems
	and call accounting systems as well as wiring telephone jacks. The
	standard for this equipment is set by the FCC, and the equipment is
	supplied by an interconnect company.
DHCP	A utility that enables a server to dynamically assign IP addresses
	from a predefined list and limit their time of use so that they can be
	reassigned. Without DHCP, an IT Manager would have to manually
	enter in all the IP addresses of all the computers on the network.
	When DHCP is used, whenever a computer logs onto the network,
	it automatically gets an IP address assigned to it.
DID	Direct Inward Dialing: The ability to make a telephone call directly
	into an internal extension without having to go through the operator.
Diff-Serv	Differentiated Services : The Diff-Serv model divides traffic into a
DIII-36IV	small number of classes to provide quality of service (QoS). One of
	QoS in internet
Divo et involved	-11
Direct inward	A Centrex feature that allows an outside caller to dial a central
dialing	business number, as well as an extension number.
Diversity antenna	A type of antenna system that uses two antennas to maximize
2110	reception and transmission quality and reduce interference.
DNS	Domain Name Service : A server/program that translates URLs to
	IP addresses by accessing a database maintained on a collection of
	Internet servers. The program works behind the scenes to facilitate
	surfing the Web with alpha versus numeric addresses. A DNS
	server converts a name like mywebsite.com to a series of numbers
	like 107.22.55.26. Every website has its own specific IP address on
	the Internet. Typically one or more DNS servers is located in an
	IP network.
DSL	Various technology protocols for high-speed data, voice and video
	transmission over ordinary twisted-pair copper POTS (Plain Old
	Telephone Service) telephone wires.
DSP	Digital Signal Processors (DSP) standardize the different states of a
	digital signal into an organized and understandable signal. DSP
	circuits can differentiate between digital signals and digital noise.
	Signal-to-Noise ratio is one of the most important factors in
	telephony voice communication because if there is too much noise
	in a signal, the DSP will be unable to find a signal standard and lose
	it. DSP circuits always adjust digital signal levels so the can
	maintain a standard without noise.
DTMF	Dual Tone Multi-Frequency: The type of audio signals generated
DTWF	when you press the buttons on a touch-tone telephone. Can be
	used in inbound(after voice channel connected) and
	outbound(before voice channel connected) application.
Ethernet	, , , ,
Luicilici	5 57
	implementations. Basic 10BaseT networks offer a bandwidth of
	about 10 Mbps. Fast Ethernet (100 Mbps) and Gigabit Ethernet
C 744	(1000 Mbps) are becoming popular.
G.711	64 kbps PCM half-duplex codec (high quality, high bandwidth,
	minimum processor load)

G.723.1	6.4/5.3 kbps MP-MLQ codec (low quality, low bandwidth, high
	processor load due to the compression)
G.726	40/32/24/16 ADPCM codec (good quality, medium bandwidth, low
	processor load)
G.729	8 kbps ACELP codec (medium quality, low bandwidth, high
	processor load)
IEEE	Institute of Electrical and Electronics Engineers, New York,
	www.ieee.org. A membership organization that includes engineers,
	scientists and students in electronics and allied fields. It has more
	than 300,000 members. The IEEE is an international organization
	that develops standards for hundreds of electronic and electrical
	technologies for computers and communications. The organization
	uses a series of numbers, like the Dewey Decimal system in
	libraries, to differentiate between the various technology families.
IP (Internet	Internet Protocol is located at 3rd layer of ISO network model. A set
Protocol)	of rules used to send and receive messages at the Internet address
	level. IP protocol is widely used in Internet and LAN networks. The
	purpose is to deliver data between computing equipment over the
	network. The protocol is generally effective but does not guarantee
IP address	complete and accurate data communications.  A 32-bit number that identifies each sender or receiver of
ir address	information that is sent across the Internet. An IP address has two
	parts: an identifier of a particular network on the Internet and an identifier of the particular device (which can be a server or a
	workstation) within that network. A number used to identify the
	location of a host device. It is expressed in numeric dot notation
	(e.g. 202.203.27.31).
	This service is offered by the LEC to the end user. The feature-rich
IP Centrex	Centrex line offers the same features and benefits as a PBX to a
II Controx	customer without the capital investment or maintenance charges.
	The LEC charges a monthly fee to the customer, who must agree to
	sign a term agreement. And the abive service is thru VoIP called IP
	centrex.
Jitter	Jitter refers to fluctuations in transmission delay time. In the case of
	voice data, conversations are packaged into packets (IP
	packetization) and transmitted. Individual packets can take different
	routes through the varied networks that comprise the Internet.
	When the time interval for the arrival of the packets is not constant
	the timing fluctuations that may occur are referred to as "jitter".
MAC	Medium Access Controller (MAC) : The IEEE 802.11 Standard
	encompasses the physical layer (PHY) and the lower portion of the
	data link layer. The lower portion of the data link layer is often
	referred to as the Medium Access Controller (MAC) sublayer. Every
	wireless 802.11 device has its own specific MAC address hard-
	coded into it. This unique identifier can be used to provide security
	for wireless networks. When a network uses a MAC table, only the
	802.11 radios that have had their MAC addresses added to that
	network's MAC table will be able to get onto the network.
<u></u>	<u> </u>

NAT	Network Address Translation: A network capability that enables a houseful of computers to dynamically share a single incoming IP address from a dial-up, cable or xDSL connection. NAT takes the single incoming IP address and creates new IP address for each client computer on the network.
Numbering plan	In a communications network, a numbering plan allocates specific numeric codes to identify each subscriber line, special lines, trunk lines, etc. Numbering plans can be divided into the numbering system and number assignments. The numbering system indicates the number of digits to be assigned and the basic scheme of number assignment. The assignment of specific codes is performed according to the numbering system.
PBX	Private Branch eXchange :An in-house telephone switching system that interconnects telephone extensions to each other as well as to the outside telephone network.
Peer-to-peer network	A wireless or wired computer network that has no server or central hub or router. All the networked PCs are equally able to act as a network server or client, and each client computer can talk to all the other wireless computers without having to go through an access point or hub. However, since there is no central base station to monitor traffic or provide Internet access, the various signals can collide with each other, reducing overall performance.
POTS	Plain Old Telephone Service (POTS), or Pretty Old Telephone System is the standard analog traditional telephone service that most homes use. In contrast, telephone services based on high-speed, digital communications, such as ISDN and DSL, are not POTS. The main distinctions between POTS and non-POTS services are speed and bandwidth.
Proxy server	Used in larger companies and organizations to improve network operations and security, a proxy server is able to prevent direct communication between two or more networks. The proxy server forwards allowable data requests to remote servers and/or responds to data requests directly from stored remote server data.
PSTN	Public Switched Telephone Network: The worldwide voice telephone network.
QoS	Quality of service (QoS)
RJ-45	Standard connectors used in Ethernet networks. Even though they look very similar to standard RJ-11 telephone connectors, RJ-45 connectors can have up to eight wires, whereas telephone connectors have only four.
Router	A device that forwards data packets from one local area network (LAN) or wide area network (WAN) to another. Based on routing tables and routing protocols, routers can read the network address in each transmitted frame and make a decision on how to send it via the most efficient route based on traffic load, line costs, speed, bad connections, etc.

RTP/RTCP	Real-Time Protocol/Real-Time Control Protocol(RTP / RTCP): IETF specifications for audio and video signal management. Allows applications to synchronize and spool audio and video information. RTP (Real Time Protocol) is specifically concerned with the dependable transmission of latency-sensitive traffic across the network and is involved in using time stamping to determine network jitter tolerance and makes sure that voice packets are arriving in order.
SIP	Session Initiation Protocol :A protocol that provides telephony services similar to H.323, but is less complex and uses less resources. SIP (Session Initiation Protocol) is a signaling protocol for Internet conferencing, telephony, presence, events notification and instant messaging. SIP is a text-based protocol, similar to HTTP and SMTP, for initiating interactive communication sessions between users.
Subnetwork or Subnet	Found in larger networks, these smaller networks are used to simplify addressing between numerous computers. Subnets connect to the central network through a router, hub or gateway. Each individual wireless LAN will probably use the same subnet for all the local computers it talks to.
TCP/IP	Internet Standard Protocol: The underlying technology behind the Internet and communications between computers in a network. The first part, TCP, is the transport part, which matches the size of the messages on either end and guarantees that the correct message has been received. The IP part is the user's computer address on a network. Every computer in a TCP/IP network has its own IP address that is either dynamically assigned at startup or permanently assigned. All TCP/IP messages contain the address of the destination network as well as the address of the destination station. This enables TCP/IP messages to be transmitted to multiple networks (subnets) within an organization or worldwide.
TOS	Type of Service: A method of setting precedence for a particular type of traffic for QoS.
UDP	User Datagram Protocol. An unreliable networking layer that sits at the same level of the networking stack at TCP. UDP is a connectionless transport protocol that is part of a suite of protocols (the others being RTP and IP) that allow for the timely and efficient transfer of voice data across and IP network. UDP is the better transport protocol for VoIP data than TCP.
VAD	Voice Activity Detection (VAD) helps save bandwidth during calls. Examples: When your making a VoIP call and your not speaking and your listening, that silence is still taking up bandwidth during the call. When silence is detected by VAD software over a predetermined length of time, is sends silent packets that inform other VAD enabled systems to stop holding the bandwidth for these empty packets.

oIP	Voice Over Internet Protocol: VoIP is based on the principal of transmitting digitized voice packets over networks. Basically, VoIP consists of converting voice signals into streams of digital packets and sending those packets of data through an IP-constructed network environment. VoIP can work in both LAN (local area network) and WAN (wide area network) environments for intranetwork or internetwork communication between VoIP channel users. Routers and switches and other special compression protocols direct the packetized voice data to their destination IP address. VoIP can be less expensive than voice transmission using standard analog packets over POTS (Plain Old Telephone Service). It allows telephone calls, faxes, or overhead paging to be transported over an existing IP data network topology.
VPN	Virtual Private Network(VPN) is a virtual created sub-network based on a public real network. A type of technology designed to increase the security of information transferred over the Internet. VPN can work with either wired or wireless networks, as well as with dial-up connections over POTS. VPN creates a private encrypted tunnel from the end user's computer, through the local wireless network, through the Internet, all the way to the corporate servers and database.
WAN	Wide Area Network: For global or long distance networking access thru the support of telecommunication devices. A communication system of connecting PCs and other computing devices across a large local, regional, national or international geographic area. Also used to distinguish between phone-based data networks and Wi-Fi. Phone networks are considered WANs and Wi-Fi networks are considered Wireless Local Area Networks (WLANs).
WLAN	Also referred to as LAN. A type of local-area network that uses high-frequency radio waves rather than wires to communicate between nodes.