



92 N. Main Street, Building 18A
Windsor, NJ 08561
www.emcomsys.com

(609) 585-5500
Fax: (609) 585-7723
info@emcomsys.com

IP6030 Family Rugged, Handset VoIP Phones Installation, Operation and Programming V3

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Product Description

The IP6030 family members are rugged, handset phones that range from simple, handset only, auto-connect units to units with a keypad, a display, auxiliary headset jacks and external speaker beacon capability. The phones are typically located in public areas and areas with harsh environments, to provide connection to emergency assistance resources or for day-to-day use. The phones utilize VoIP (Voice over Internet Protocol) and communicate through the Ethernet with other devices using SIP (Session Initiation Protocol). The various IP6030 family members are designated with a dash number as shown in table 1.

Each IP6030 has its own phonebook which may be programmed to autodial or, when its keypad is pressed, connect to other devices such as

- VoIP telephones
- Analog telephones (through gateways)
- The Public Switched Telephone Network
- PBXs
- The EmVista Central call management and conferencing system

Programming of the phone is accomplished through a simple web browser interface or through Emcom Systems' Emvista Maintenance Station application, which is typically used for large installations.

The IP6030 family utilizes DSP (Digital Signal Processing) to manage audio levels. The audio levels are remotely programmable and the handset contains a press bar to set 3 levels of earpiece volume. The handset is hearing aid compatible.

Table 1, Model Number and Function

Model Number	Function	Circuit Boards
IP6030-0	Auto-connect, no keypad	1 board
IP6030-1	Keypad	1 board
IP6030-3	Keypad and external speaker/beacon	2 boards
IP6030-15	Keypad, display, external speaker/beacon, headset jacks	2 boards

2 Installation

2.1 Mechanical

Figure 1 depicts the outline of the IP6030 with overall dimensions and spacing between the 6 mounting holes on the front panel. The unit is typically secured with either 10-32 flat head or 10-32 pan head screws and is supplied with 6 pan head, bit-spanner screws. If flat head screws are used, the threaded devices that the screws are fastened to should be floating to ensure that the screws seat properly in order to prevent easy removal.

An IP6030 requires 4 inches of space behind the front panel. If the unit is mounted in a wall, an in-wall enclosure should be used to ensure greater environmental integrity. A bezel is available to hide the rough-in of in-wall mounting. Emcom Systems can provide the bezel and both in-wall and surface mount enclosures. It is recommended that where possible, all access to enclosures be from the bottom.

2.2 Electrical

The opening at the bottom of an IP6030 back box provides access to the unit's external connections. For IP6030 units that do not use an external speaker beacon, there is one circuit board. There are two circuit boards for units that use an external speaker-beacon.

2.2.1 Single board units

For the single board unit, there are 9 connectors, but in a typical application only 2 connectors are used. These are for power and for the Ethernet connection. Additional connectors provide for auxiliary contact closures, for a second Ethernet connection, a diagnostic port, an external trigger, an external audio output and an external audio input. The function of each connector is marked on the case of the unit.

2.2.1.1 Power

J2 – 12VDC to 48VDC non-polarized

J3 – POE

The IP6030 may be operated from a 6 watt, ripple-free source of from 12 – 48VDC on connector J2. The connector is labeled EXT PWRIN on the case. When POE is supplied on Ethernet connector J3 no additional power is required. Connector J3 is labeled PoE Port on the case. The maximum rated current of the unit is 1 amp at 12VDC. The electronics are protected by a F1, a self-resettable fuse rated at 1 amp at 60 C.

2.2.1.2 Ethernet Connections

J3, J4 – The IP6030 contains a 100 Mbit 3-port Ethernet switch. One of the switch ports is the IP phone itself. The other 2 switch ports, RJ45 connectors J3 and J4, allow for connection to the unit, for adding another Ethernet device or for daisy-chaining IP6030 units. Connector J3 is labeled PoE Port and J4 is labeled Ether Port.

2.2.1.3 Auxiliary Contact Closures

J28 – There are two auxiliary contact closures available on the unit. These are opto-isolated, normally open, connections capable of operating at 120 volts DC with 120 milliamps maximum current. The contact closures may be used for gate release, beacon activation, etc. These contact closures are labeled AUX 1 and AUX 2 on the case of the unit. In the standard configuration AUX 1 opens and closes at a 1 second rate when a phone is active and AUX 2 is continually closed when a phone is active.

2.2.1.4 Diagnostic Connector

J17 – This 8-pin connector allows a serial port connection and is normally used for factory troubleshooting. This connector is labeled SERIAL on the case of the unit.

2.2.1.5 External Trigger

J26 – This connector provides for an external means of activating the unit and is used for special circumstances. Please consult the factory should there be a need for a function of this nature. This connector is labeled CTRL on the case.

2.2.1.6 External Audio Output

J24 – This connector provides output for an external amplifier and speaker that may be attached to the unit. This 600 ohm output is AC coupled. It is labeled AUDOUT on the case of the unit.

2.2.1.7 External Audio Input

J27 – This connector provides for an external line-level audio source that may be placed in parallel with the mouthpiece of the handset. This 600 ohm input is AC coupled and should not exceed 2 volts peak-to-peak. This connector is labeled AUDIN on the case of the unit.

2.2.1.8 Ancillary Power

J1 – This connector provides redirected input power to drive subsidiary boards that may be used in various versions of the IP6030. When not used, this connector is covered over. This connector is labeled PWR OUT on the case of the unit.

2.2.2 Two Board Units

For IP6030 units with external speaker beacon functionality, there is a second circuit board with connections to the first board. External connections are then made to this second board and to the Ethernet connectors of the first board. The speaker beacon board provides amplification to the AUDOUT signal from the first board. The speaker beacon board also provides regulated current and on/off control to an external high intensity beacon.

2.2.2.1 Power

For an IP6030 with two boards to fully function, the unit must be powered from 12 to 24VDC only. The unit is marked to indicate this and power must be connected to J2 of the speaker beacon board (labeled PWR IN on the case). POE may be applied to J3 of the main board and the phone portion of the unit will function, however the speaker beacon board will not function as it is protected from operating from the higher POE input voltage. The maximum rated current of the unit is 1 amp at 12VDC. The electronics are protected by F1, a self-resettable fuse rated at 1 amp at 60 C.

2.2.2.2 Speaker

Output to an external speaker is provided on the second board and is labeled SPKR. The external speaker impedance must be 4 ohms or higher.

2.2.2.3 Beacon

Connection to an Emcom Systems blue, high intensity, LED beacon is provided at the connector labeled BEACON. This output provides a constant current to ensure LED long life.

2.3 Operating Temperature Range

The operating temperature range of the IP6030 family is from -30 C to +55 C (-22 F to 131 F).

3 Operation

The IP6030 communicates with other VoIP networked devices using SIP. Voice connections can be made to other SIP devices directly or to other non-SIP devices such as analog telephones through SIP intermediaries called gateways.

3.1 Calls From An IP6030

When the handset of a non-keypad IP6030 is lifted, the unit, using SIP, places a call via the Ethernet port to the far end communication device whose access information has been previously programmed into the IP6030 phone book. If the IP6030 is a keypad unit, when the handset is lifted dial-tone is heard. The user may they dial any number or group of numbers that have been previously programmed in the phone book as allowable. In either instance, keypad or non-keypad, if the far end communication device responds, a SIP dialog is held with the IP6030 and ringback tone is heard. When the far end device is answered, ringback tone ceases and a two way conversation may be held. During the conversation, the audio level of the earpiece may be adjusted with the press bar in the handset. When either the IP6030 or the far end communication device hangs up, another SIP dialog takes place, the call is terminated.

The phone book of the IP6030 can be programmed with the access information of up to 100 far end devices that can be arranged in a fall through list. That is, if the first device does not respond or is busy, the IP6030 tries the next one, and so on, until an available device is reached or until the complete list has been tried. The phone book can also be programmed so that any number of a defined length can be dialed.

3.2 Calls to An IP6030

Calls are made to the IP6030 from a far end communications device in the reverse manner. The far end device simply dials (or autodialed) the access information of the IP6030 and a SIP dialog is begun. The IP6030 acknowledges the receipt of the call, issues ringing tones until the handset is lifted. Two-way conversation can then take place. When the far end communications device hangs up, another SIP dialog takes place and the call is terminated.

3.3 Additional IP6030 functionality

3.3.1 Speaker Beacon Option

With a speaker beacon board included in the unit, the ability to add an external high power speaker and a high intensity blue beacon is enabled. The board provides up to 20 watts of audio power to an external 4 ohm speaker and provides a current source with flashing capability for the beacon.

When calls are placed to the IP6030, the external speaker functions as the ringer and issues loud alerting tones. The external speaker may also be used as a paging device when the IP6030 is connected in an EMVista Central system. Paging calls may be made from an IP6030 to other IP6030 units singly or in groups.

When a beacon is attached to an IP6030 unit, the beacon is always on to indicate the location of the unit. When a call is placed from the IP6030 or a call is made to the unit, the beacon flashes to indicate the unit making or receiving the call. The beacon continues to flash for the duration of the call and returns to continuous illumination when the call is terminated.

3.3.2 Display Option

The display of an IP6030 unit, when so equipped, indicates the time, the ID number of the unit, and the call status of incoming or outgoing calls such as Idle, Off Hook, Dialed number, Incoming Call Number and Call Terminated, Hang Up.

3.3.3 Headset Jack Option

If the IP6030 is equipped with headset jacks, an aviation type headset may be plugged in. This type of headset has a separate plug for the earpiece and for the microphone. When only the earpiece is plugged in, the headset may be used to monitor the phone conversation and the handset of the unit is still active. When the microphone plug is inserted, the mouthpiece of the handset is disabled and the microphone of the headset is enabled. The handset may be placed on hook and the conversation will not be terminated until the headset microphone plug is removed.

4 Programming

Programming of the functions and phonebook of the IP6030 may be accomplished through a web interface from any current browser or by using the EMVista Maintenance Station.

4.1 Programming Using the Browser Interface

4.1.1 Entering the Browser Interface

The phone setup is reached by placing the IP address of the IP6030 in the address line of the browser and pressing enter.

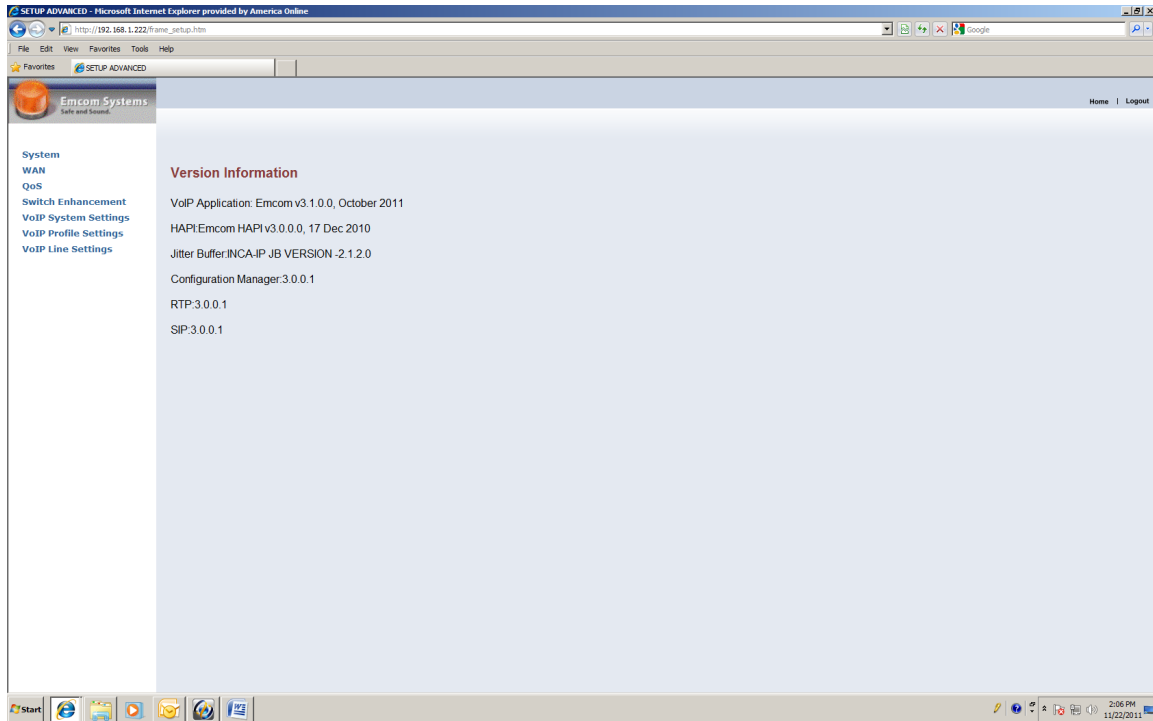
As shipped from the factory, the IP address of the unit is 192.168.1.22X where X is equal to the last digit of the serial number of the unit. The serial number is on the back case of the unit.

Example: Serial number is 653 IP address is 192.168.1.223

Upon pressing enter, the login page will appear.

The default password is: “1234”

Upon entering the password and pressing enter, the following menu screen will appear.

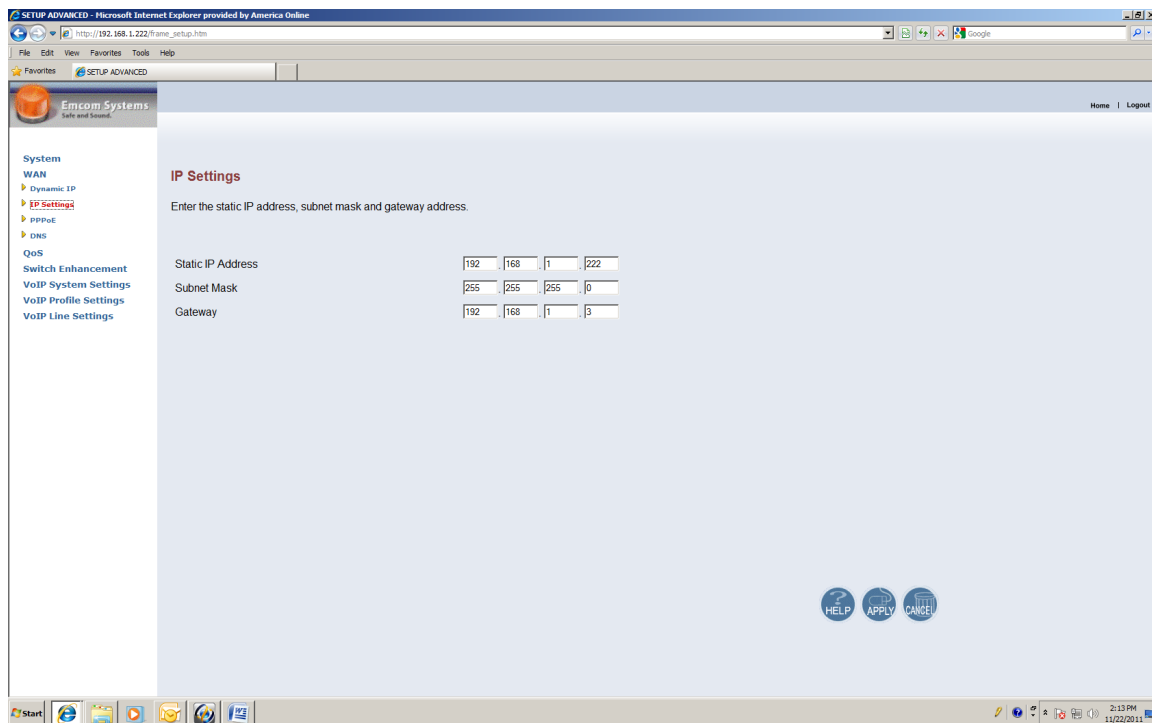


The right pane of this screen provides software version information. The left pane of the screen provides a menu that is always available to navigate to other setup functions.

4.1.2 Changing the IP Address, Subnet Mask and Default Gateway

Using the left pane navigator, click on WAN. The navigation tree will expand and three selection buttons will appear in the right pane. Make sure that the Static IP Address button is selected.

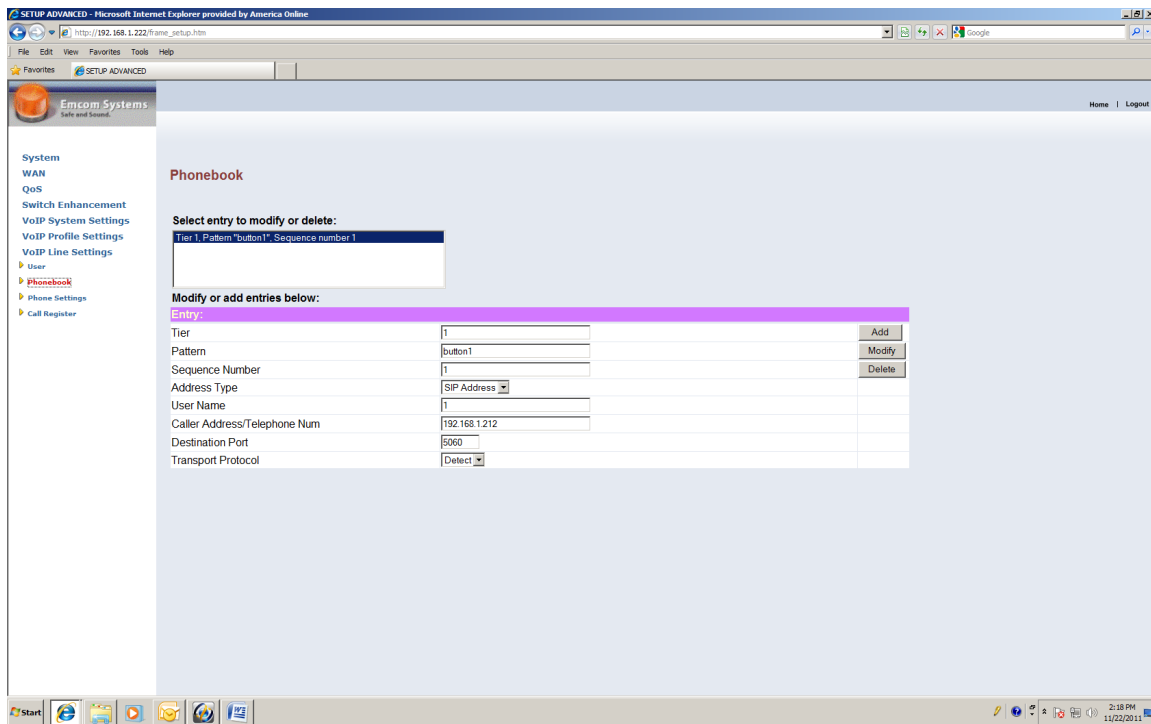
In the navigation tree, click on Static IP Settings. The following screen will appear:



Change the IP Address, the Subnet Mask and the IP Gateway to the desired settings and click on apply. The IP6030 will automatically enter the new values and will restart itself. Since the IP6030 has been restarted, the connection to the browser will have been lost. The connection is re-established by again entering the IP address in the browser access line.

4.1.3 Changing the Phonebook

The phonebook of the IP6000 is changed by clicking on VoIP Line Settings in the navigation tree pane and then by clicking on Phonebook in the submenu. The following window will appear:



The phonebook consists of a list of entries that are created by filling in the fields of the above table. Each phonebook entry maps an input condition (Pattern), such as a number dialed on the IP6030’s keypad, to a target destination called a SIP-URI which is presented to the target as User Name @ Caller Address/Telephone Num. The individual fields are defined as follows:

Tier – Tier is a sequencing mechanism used for complex searches that are sometimes required for large configurations. Normally, all IP6030 calls should be assigned to Tier “1”.

Pattern – The pattern describes the input condition. Valid conditions are:

Offhook – Lifting the handset will cause the IP6030 to begin a call without waiting for digits to be dialed.

Digits and question marks (up to 20) – The phone will compare strings of digits from the keypad to patterns in this form. Question marks serve as wildcard characters but particular digits have precedence over question marks. If the phonebook contains the following patterns: “????”, “1????”, and “1234”, then a dialed string “1234” will always match the third pattern, any other four-digit string beginning with “1” will match the second, and any other four-digit string will

match the first. The length of the dialed string must match the length of the pattern. If the phonebook contains “????” and “9?????????”, then any number beginning with “9” must have eleven additional digits, but any number beginning with a digit other than nine will initiate a call after three more digits are dialed.

Sequence Number – Sequence numbers apply to phonebook entries with identical patterns and different destinations. They specify the order in which succeeding calls are placed if the first call does not go through. Sequence “1” would be the first attempt, Sequence “2” the second attempt, etc.

Address Type – Address Type defines the method of communication. This should be left at SIP Address.

User Name – This specifies the user ID field of the SIP-URI sent to the target destination. If the Destination User Name is filled in with an “.”, then the user ID field is composed of the input condition (Pattern). This method can be used to pass a phone number dialed by the user to a gateway, PBX, or other device that routes the call on the basis of the SIP-URI. If the Destination User Name is anything but “.”, it is copied to the user ID field literally. For direct calls from one IP6030 to another, this field must contain “1”. This field must always be specified, even for SIP devices that ignore it. In the case of such devices, we recommend using “.”.

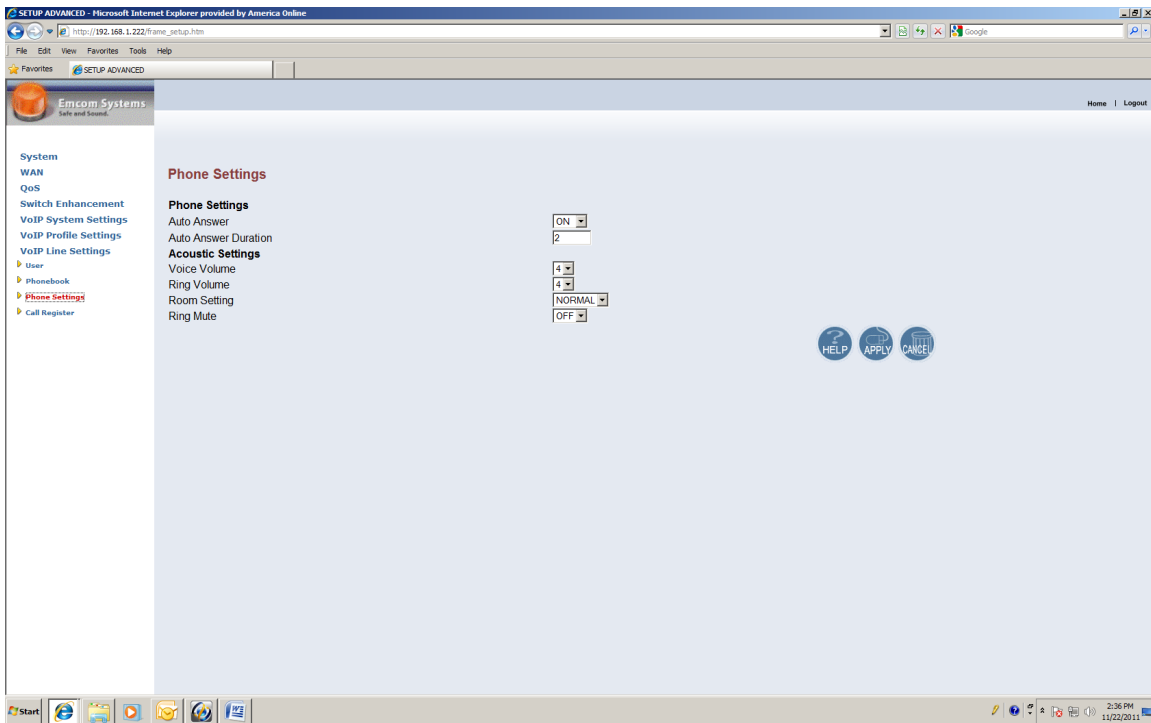
Caller Address/Telephone Num – This is the IP address of the SIP device that is to receive the call.

Destination Port – The Destination Port is a standard port number assigned to IP phones and should be left at 5060.

Transport Protocol – The Transport Protocol should be left at Detect.

4.1.4 Changing the Acoustic Environment and Audio Levels

The acoustic environment and audio level settings are also reached as a subdirectory of the VoIP Line Settings menu. Click on VoIP Line Settings and then click on Phone Settings. The following screen will appear:



Auto Answer – With auto answer “on”, the IP6030 will ring (if there is a speaker attached) and then automatically go off hook on an incoming call. With auto answer “off”, the IP6030 will continue to ring until the handset is lifted.

Auto Answer Duration – This setting establishes how long, from 2 to 5 seconds, that the IP6030 will ring when Auto Answer is set to “on”.

Voice Volume - Voice volume sets the level heard from the IP6030 speaker (if a speaker is attached and it is being used in a paging mode). The adjustment levels are from 0 to 7 with 7 the loudest. The numbers on this screen may be changed when the phone is in idle mode. The next page will reflect the new level.

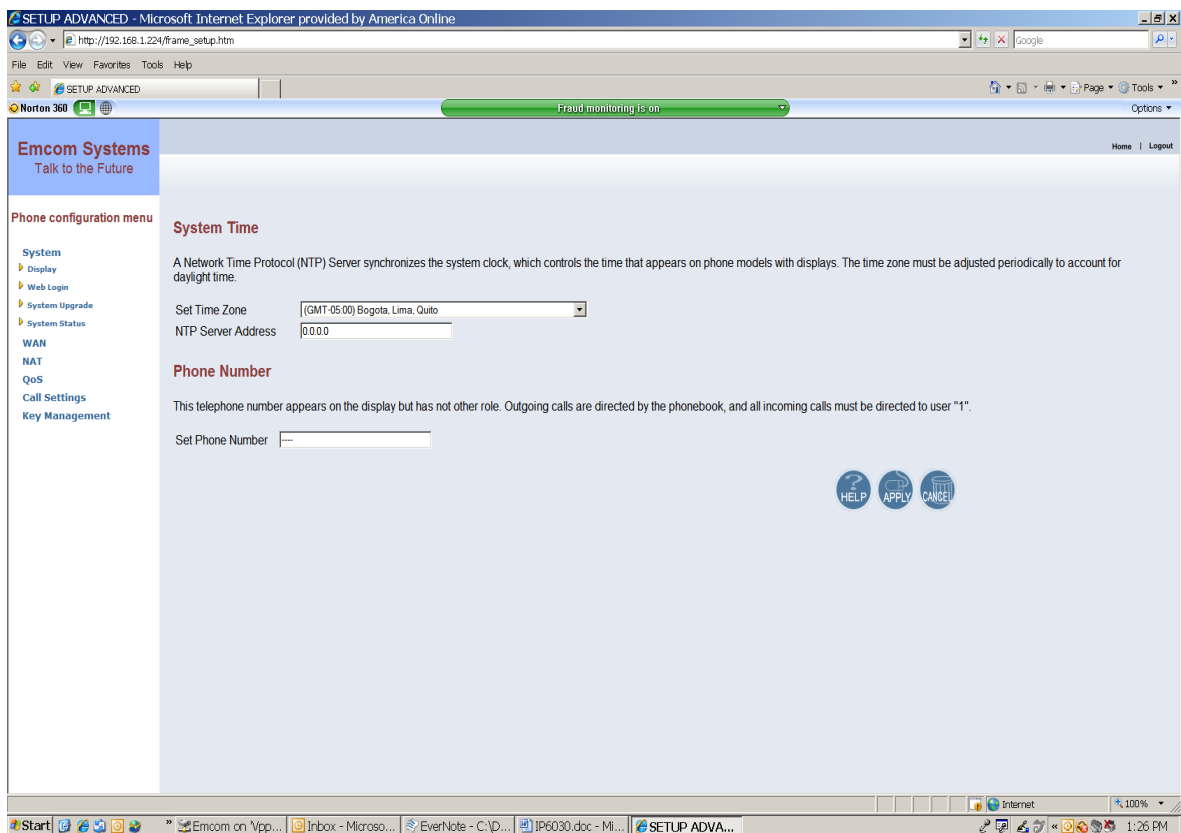
Ring Volume – Ring Volume adjusts the alerting level first heard (if a speaker is attached) from the IP6030 when a call is placed to it. The scale is from 0 to 7.

Room Setting – This setting allows adjusting for Normal, Noisy or Echoic rooms.

4.1.5 Setting the Time and Phone Number on the Display

The phone number can be made to appear on an IP6030 with a display. If the network to which the IP6030 is connected has an NTP (Network Time Protocol) server, the time can also be made to appear on the display.

Using the left pane navigator, click on SYSTEM and under SYSTEM, click on Display. The following window will appear.



For time, set the appropriate time zone and enter the IP address of the NTP server. Enter the phone number that is desired to appear on the display. Then click APPLY.

4.1.6 Additional Programmable Functions

A number of additional programmable functions are available through the various submenus of the left navigation pane. These functions are typically used only for unique applications. It is recommended that the user first communicate with factory support before any setup changes in these areas are attempted.

4.2 Programming Using EMVista Maintenance Station

EMVista Maintenance workstation is an application that resides on a laptop or workstation on the network and provides an automated interface for programming the IP6030. It also maintains a database of all of the phonebook information and programming parameters of each IP6030 and automatically tests each unit.

The use of the EMVista Maintenance workstation is discussed in the document EMVista Maintenance Station User Guide

