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# IP6002
# Electronics Only, VoIP Phone
# Installation, Operation and Programming
# V3

## June 03, 2021

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RECORD OF CHANGES

Changes to this IP6002 Electronics Only, VoIP Phone I&O and Programming V3 shall be recorded in the accompanying table.

Record of Changes Table

| Rev # | Date | Brief Description | Change Order # |
|---|---|---|---|
| C | 5/31/2018 | New Logo<br>Changes to support UL60950-1 | |
| D | 6/3/2021 | New address | |
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## 1. Product Description

The IP6002 is the electronics for a VoIP, hands-free phone enclosed in a protective housing. It is used in conjunction with an externally mounted speaker/microphone assembly and external LED assemblies that indicate call progress.

The phone utilizes VoIP (Voice over Internet Protocol) and communicates through the Ethernet with other devices using SIP (Session Initiation Protocol).

The IP6002 has its own phonebook which may be programmed so that when its pushbutton is pressed the unit connects 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 IP6002 utilizes DSP (Digital Signal Processing) to manage audio levels and can be set for normal, echo prone or noisy environments. The environment settings and volume levels are remotely programmable.

The IP6002 can be powered from DC sources of 12 to 48 Volts and is compatible with POE (Power over Ethernet).

## 2. Installation

### 2.1 Mechanical

Figure 1 depicts the outline of the IP6002 with overall dimensions. The unit is secured using the attached DIN rail mounting clip. The housing of the unit has additional holes to allow moving the DIN rail mounting clip to accommodate various mounting situations.

### 2.2 Electrical

The opening in the housing of the IP6002 provides access to the unit's external connections. The connections are indicated in Figure 1 and are described below.

#### 2.2.1 Power

J2 – 12VDC to 48VDC non-polarized
J3 - POE
The IP6002 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 housing. When POE is supplied on Ethernet connector J3 no additional power is required. Connector J3 is labeled PoE Port on the housing. The maximum rated current of the unit is 1 amp at 12VDC. The electronics are protected by F1, a self-resettable fuse rated 1 amp at 60 C.

### 2.2.2 Ethernet Connections

J3, J4 – The IP6002 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 IP6002 units. Connector J3 is labeled PoE Port and J4 is labeled Ether Port.

### 2.2.3 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 SPARE on the housing 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.

### 2.2.4 Diagnostic Connector

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

### 2.2.5 External Trigger

J26 – This connector provides for an external means of activating the unit. It is internally opto-isolated from the core electronics. A momentary closure at J26 provides activation. This connector is labeled BUT 1 IN on the housing.

### 2.2.6 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 SPARE on the housing of the unit.

### 2.2.7 External Audio Input

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

### 2.2.8 Ancillary Power

J1 – This connector provides redirected input power to drive subsidiary boards that may be used in various configurations of the IP6002. When not used, this connector is covered over. This connector is labeled VDC OUT on the back case of the unit.

### 2.2.9 Microphone

MIC – This connector provides an external connection for the microphone used with the IP6002.

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### 2.2.10 Speaker

SPK – This connector provides an external connection for the speaker used with the IP6002.

### 2.2.11 Red LED

RED LED – This connector provides an external connection for the red LED typically used with the IP6002.

### 2.2.12 Green LED

GRN LED – This connector provides an external connection for the green LED typically used with the IP6002.

## 2.3. Operating Temperature

The operating temperature range of the IP6002 is from -30 to +55 C (-22 to 131 F).

# 3. Operation

The IP6002 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 IP6002

When a contact closure is made at the BUT 1 IN connector, 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 IP6002 phone book. At the same time, the red LED connected to the unit flashes to indicate that a call has been placed and a ring back tone is heard from the connected speaker. If the far end communication device responds, a SIP dialog is held with the IP6002. The red LED is extinguished, the ring back tone ceases, the connected green LED begins flashing and a two way conversation may be held between the IP6002 and the far end communication device. When the far end communication device hangs up, another SIP dialog takes place, the call is terminated and the green LED is extinguished.

The phone book of the IP6002 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 IP6002 tries the next one, and so on, until an available device is reached or until the complete list has been tried.

## 3.2 Calls to An IP6002

Calls are made to the IP6002 from a far end communications device in the reverse manner. The far end device simply dials (or autodials) the access information of the IP6002 and a SIP dialog is begun. The IP6002 acknowledges the receipt of the call, issues a shrill ringing tone to the speaker and flashes the green LED. Two-way conversation can then take place. When the far end communications device hangs up, another SIP dialog takes place, the call is terminated and the green LED is extinguished.

## 4. Programming

Programming of the functions and phonebook of the IP6002 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 IP6002 in the address line of the browser and pressing enter.

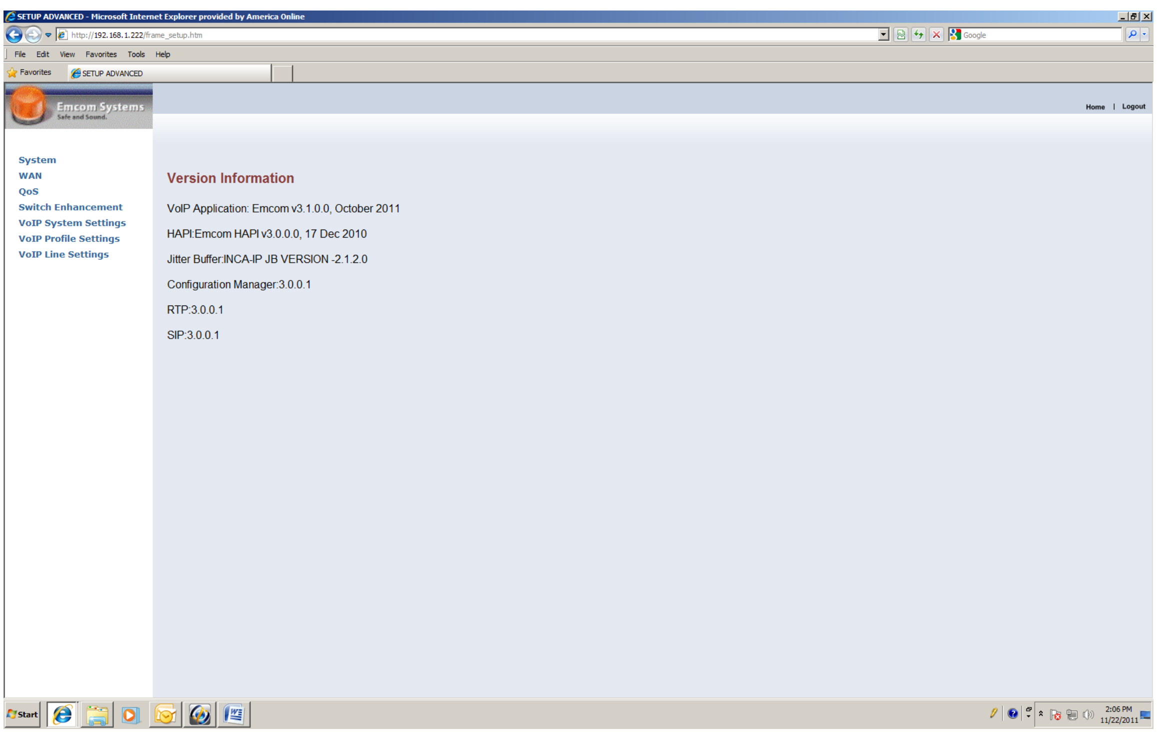
As shipped from the factory, the IP address of the unit is 192.168.1.22X where X is equal to the last digit of the serial number of the unit. The serial number is on the back case of the unit.

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.

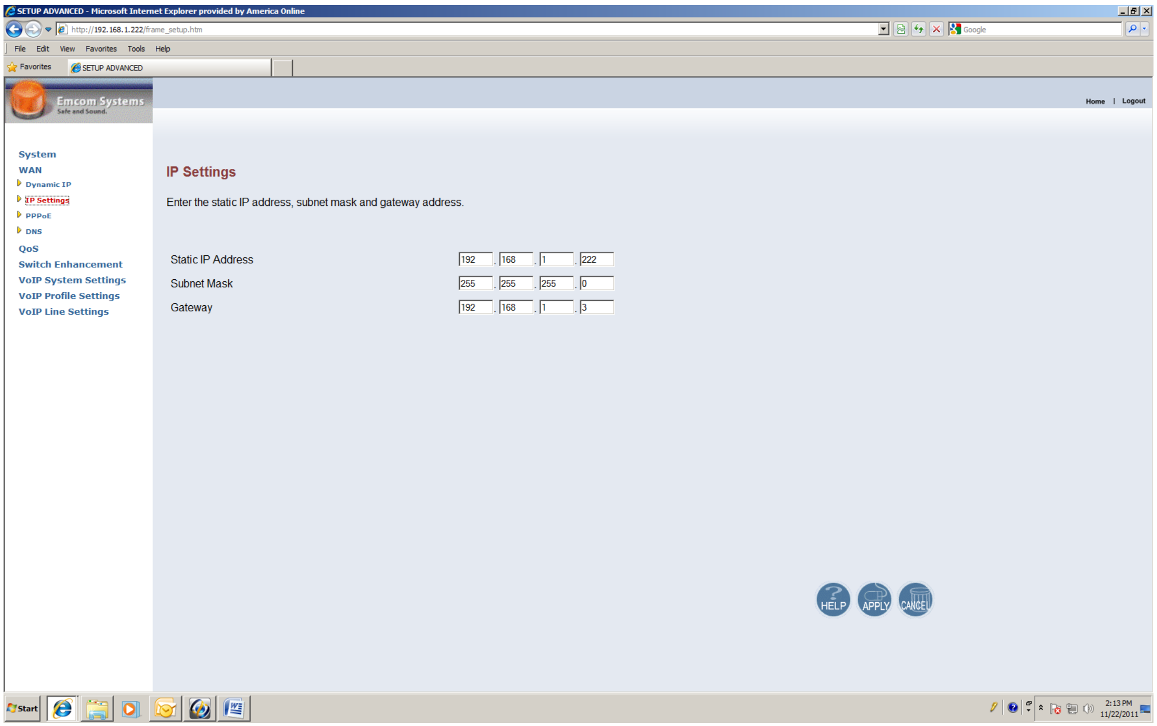



The right pane of this screen provides software version information. The left pane of the screen provides a menu that is always available to navigate to other setup functions.

### 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.

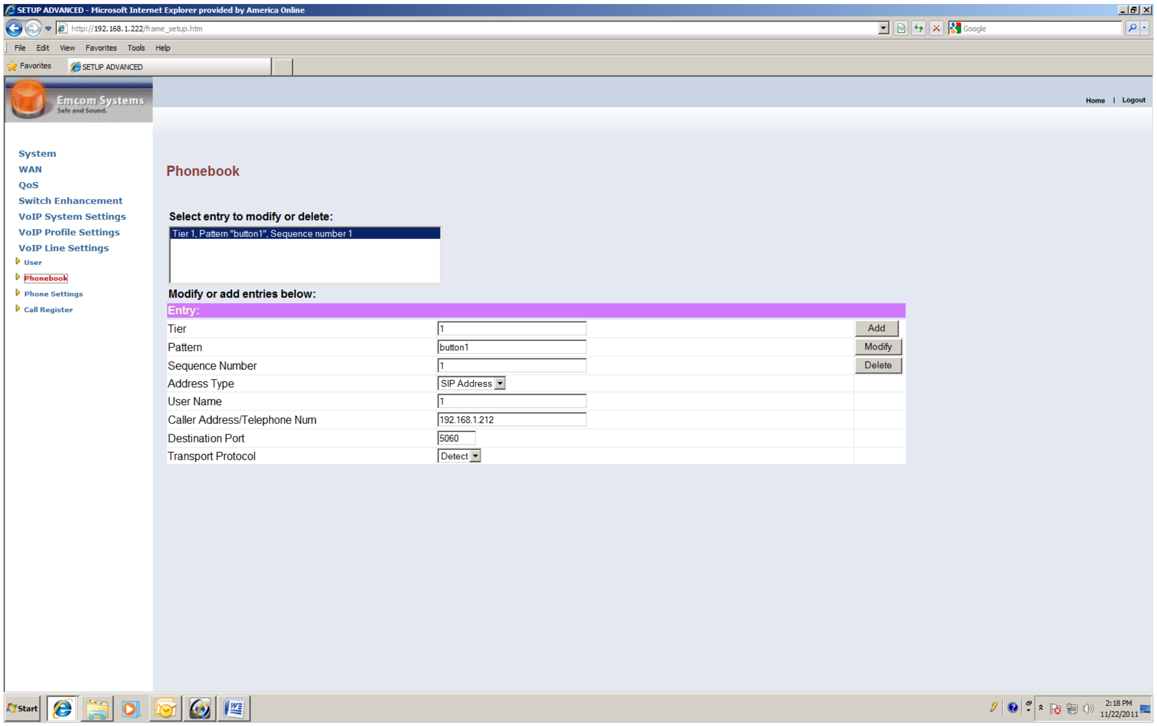
In the navigation tree, click on IP Settings. The following screen will appear:




Change the IP Address, the Subnet Mask and the IP Gateway to the desired settings and click on apply. The IP6002 will automatically enter the new values and will restart itself. Since the IP6002 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 IP6002 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 setup functions in this screen are defined as follows:

**Tier** – Tier is a sequencing mechanism used for complex searches that are sometimes required for other models of Emcom phones. All IP6002 calls should be assigned to Tier "1".

**Pattern** – Pattern defines the event that enables a call. For the IP6002 it is the pressing of "Button1". Note that there is no space in "Button1".

**Sequence Number** – Sequence Number defines the sequence 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 is the local number of the device attached to the receiving SIP device. It is normally designated in the setup of the receiving device. For Example: If the final receiving device is an analog phone connected to one of the ports of a SIP gateway, the User Name would be the port number assigned to the port that the analog phone is connected to. If the SIP receiving device is the final destination (such as another IP phone) nothing need be entered here.

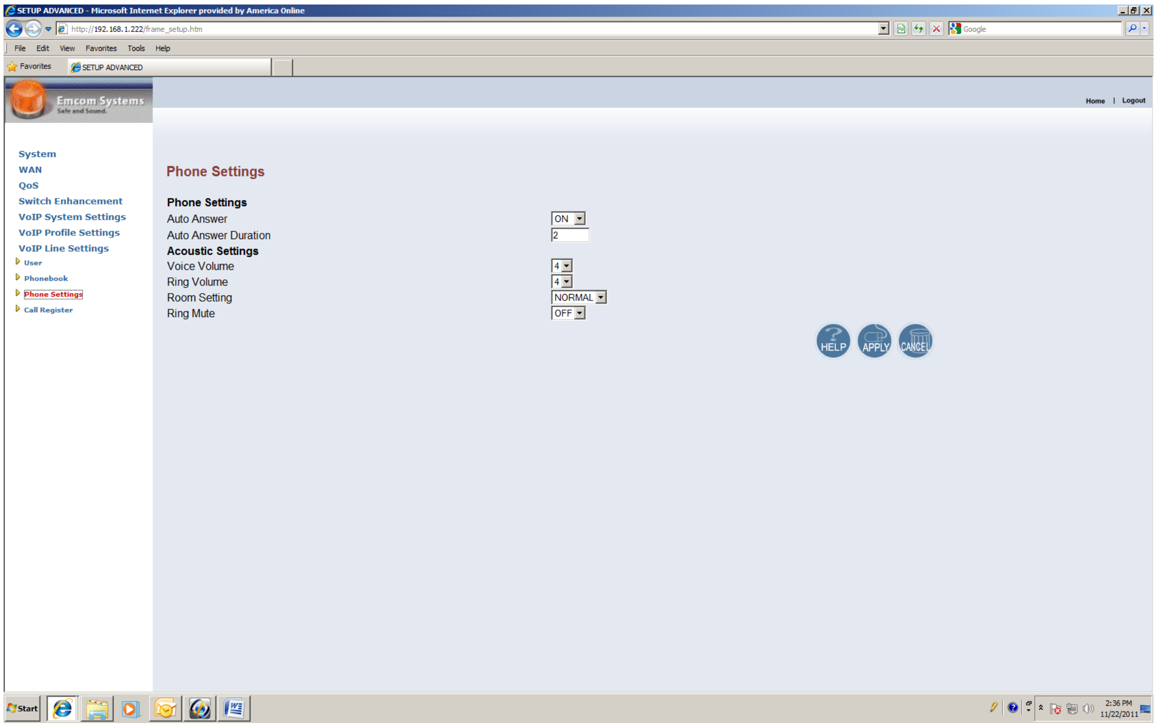
**Destination Port** – The Destination Port is a standard port number assigned to IP phones and should be left at 5060.

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

**Transport Protocol** – The Transport Protocol should be left at Default.

### 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 IP6002 will ring and then automatically go off hook on an incoming call. With auto answer "off", the IP6002 will continue to ring until the activation button is pressed. It will then go off hook.

**Auto Answer Duration** – This setting establishes how long, from 2 to 5 seconds, that the IP6002 will ring when Auto Answer is set to "on".

**Voice Volume** - Voice volume sets the level heard from the IP6002 speaker. 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 conversation through the IP6000 will reflect the new level.

**Ring Volume** – Ring Volume adjusts the alerting level first heard from the IP6002 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.

**Ring Mute** – When Ring Mute is set to "on", the ringer is disabled.

### 4.2 Additional Browser 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.

### 4.3 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 IP6002.  It also maintains a database of all of the phonebook information and programming parameters of each IP6002 and automatically tests each unit.

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

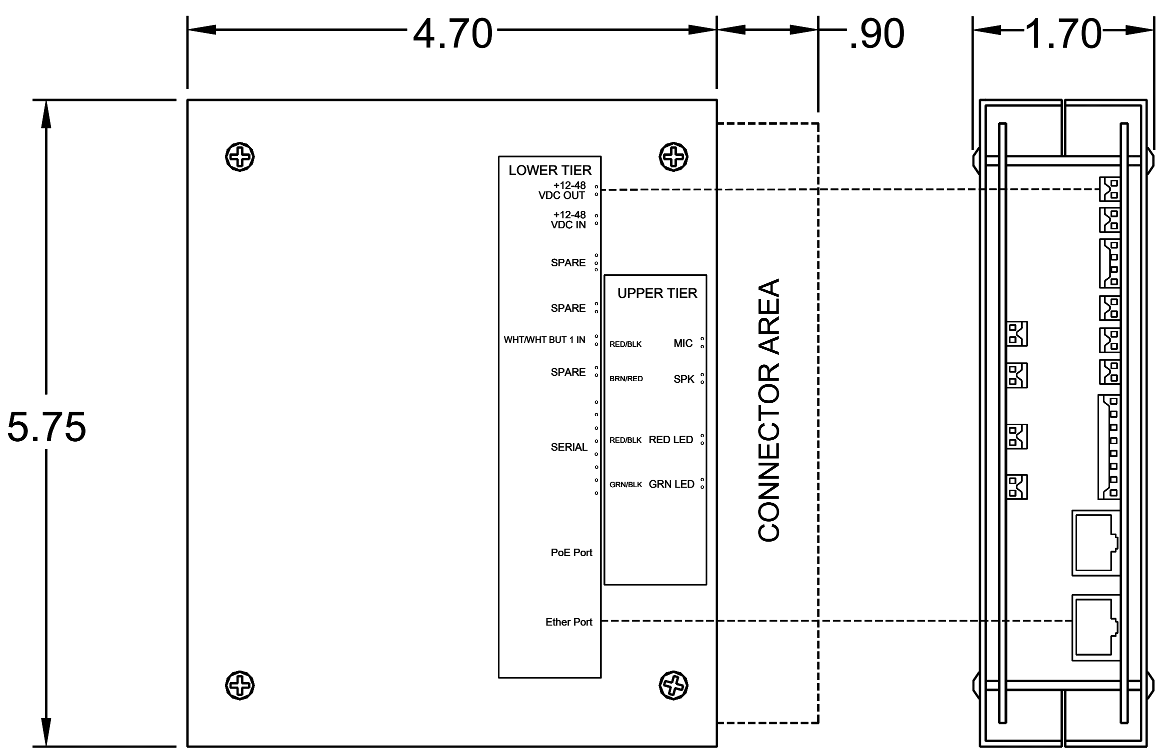



**FIGURE 1**