

ADA Compliant VoIP Tower Phones with Blue LED Beacon and Strobe Light now with Multicast Paging

The **E-1600-BLT1PEWP**, **E-1600BLT21PEWP**, **E-1600-AST1PEWP** and **E-1600AST21PEWP** ADA Compliant VoIP Emergency Tower Phones are designed to provide quick and reliable handsfree communication for SIP VoIP phone systems with PoE. The tower phones meet ADA requirements for emergency telephones, and can be programmed from any touch tone phone or PC on the same LAN or remotely using a static IP address. The phones can dial up to 5 programmable emergency numbers. In addition, the **E-1600BLT21PEWP** and **E-1600AST21PEWP** feature a second "INFO" button that will dial up to 5 non-emergency numbers.

At the simple push of a button, the Tower Phone will initiate a call to your emergency personnel and send a digital announcement to identify the location of the emergency call. In addition, the tower phone's bright LED strobe light will instantly begin flashing to deter further activity and make it fast and easy for Police or Security personnel to locate the site of the emergency. The strobe light can also be programmed to provide a continuous-on lower intensity beacon when the emergency phone is not in use.

All four Tower Phone Models are equipped with Enhanced Weather Protection (EWP) for outdoor installations where the unit is exposed to precipitation or condensation. EWP products are designed to meet IP66 standards and may feature foam rubber gaskets, sealed connections, gel-filled butt connectors, as well as potted circuit boards with internally sealed, field-adjustable trim pots and DIP switches for easy on-site programming. For more information on EWP.

 **Installation requires the assistance of a Network Administrator / IT Technician.**



E-1600-BLT1PEWP E-1600BLT21PEWP E-1600-AST1PEWP E-1600AST21PEWP

Features

- Automatic polling and programming software included
- SIP compliant (see page 2 for more information)
- **Paging prioritization**
- **SIP endpoint or multicast group member**
- **Compatible with Polycom multicast paging**
- **SIP/Multicast: SIP page, SIP page and zoned multicast receive**
- PoE powered (class 1, <4 Watts)
- VoIP eliminates the need for "Push to Talk" mode
- Network downloadable firmware
- Meets ADA requirements for Emergency Phones:
 - Automatically lights the "Call Connected" LED
 - Transmits a location I.D. or voice announcement
 - Grade 2 Braille label for the visually impaired
- Non-volatile digital voice announcer with 28 seconds of voice memory
- Handsfree operation
- Marine grade 316 stainless steel prevents faceplate / push button corrosion
- Dials up to 5 emergency numbers
- **E-1600BLT21PEWP** and **E-1600AST21PEWP** can also dial up to 5 additional non-emergency "INFO" numbers
- Cycles through backup phone numbers on busy or no-answer
- Comes standard with Enhanced Weather Protection (EWP), EWP products are designed to meet IP66 Ingress Protection Rating
- Hangs up on busy signal, time-out or touch tone command
- Remotely programmable
- Extended temperature range of -40°F to 140°F
- Optional **PB-100** Polling System available

Applications

- Campus Security Sites
- Area of Refuge sites
- Parking Ramps/Lots
- Automated Tellers (ATM)
- Entryways
- Roadside Emergency Sites
- Stairwells in Public Buildings

www.VikingElectronics.com
Information: 715-386-8861

Specifications

Phone Power: PoE class 1 (<4 Watts)
Beacon/Strobe Power: 120VAC / 12VDC power adapter (included)
Maximum Sound Pressure: 90 dB SPL @ 1m
Dimensions: See Installation and Specifications
Operating Temperature: -40°F to 140°F (-40° C to 60° C)
Humidity: Up to 100%
Audio Codecs: G711u, G711a, G722
Network Compliance: IEEE 802.3 af PoE, SIP 2.0 RFC3261, 100BASE-TX with auto cross over
Connections: (1) RJ45 10/100 Base-T, (3) gel-filled butt connectors

VoIP SIP System Compatibility

For compatibility and vendor specific detailed configuration instructions, see the **Viking VoIP SIP System Compatibility List**, DOD 944. To open and download this PDF file:

Scan the QR code below to open and download the **Viking VoIP SIP System Compatibility List**



- OR -

1. Go to **www.vikingelectronics.com** and enter **944** in the search box
2. Click **Application Note (DOD 944)** to open and download the PDF

Important: *Exclusion from this list means only that compatibility has not been verified, it does not mean incompatibility. If you have questions, please call Viking Electronics at 715-386-8861.*

Definitions

Client: A computer or device that makes use of a server. As an example, the client might request a particular file from the server.

DHCP: Dynamic Host Configuration Protocol. In this procedure the network server or router takes note of a client's MAC address and assigns an IP address to allow the client to communicate with other devices on the network.

DNS Server: A DNS (Domain Name System) server translates domain names (ie: www.vikingelectronics.com) into an IP address.

Ethernet: Ethernet is the most commonly used LAN technology. An Ethernet Local Area Network typically uses twisted pair wires to achieve transmission speeds up to 1Gbps.

Host: A computer or device connected to a network.

Host Name: A host name is a label assigned to a device connected to a computer network that is used to identify the device in various forms of network communication.

Hosts File: A file stored in a computer that lists host names and their corresponding IP addresses with the purpose of mapping addresses to hosts or vice versa.

Internet: A worldwide system of computer networks running on IP protocol which can be accessed by individual computers or networks.

IP: Internet Protocol is the set of communications conventions that govern the way computers communicate on networks and on the Internet.

IP Address: This is the address that uniquely identifies a host on a network.

LAN: Local Area Network. A LAN is a network connecting computers and other devices within an office or building.

Lease: The amount of time a DHCP server reserves an address it has assigned. If the address isn't used by the host for a period of time, the lease can expire and the address can be assigned to another host.

MAC Address: MAC stands for Media Access Control. A MAC address, also called a hardware address or physical address, is a unique address assigned to a device at the factory. It resides in the device's memory and is used by routers to send network traffic to the correct IP address. You can find the MAC address of your **VoIP Tower** phone printed on a white label on the top surface of the PoE LAN port.

Router: A device that forwards data from one network to another. In order to send information to the right location, routers look at IP Address, MAC Address and Subnet Mask.

RTP: Real-Time Transport Protocol is an Internet protocol standard that specifies a way for programs to manage the real-time transmission of multimedia data over either unicast or multicast network services.

Server: A computer or device that fulfills requests from a client. This could involve the server sending a particular file requested by the client.

Session Initiation Protocol (SIP): Is a signaling communications protocol, widely used for controlling multimedia communication sessions such as voice and video calls over Internet Protocol (IP) networks. The protocol defines the messages that are sent between endpoints, which govern establishment, termination and other essential elements of a call.

Static IP Address: A static IP Address has been assigned manually and is permanent until it is manually removed. It is not subject to the Lease limitations of a Dynamic IP Address assigned by the DHCP Server. The default static IP Address is: **192.168.154.1**

Subnet: A portion of a network that shares a common address component. On TCP/IP networks, subnets are defined as all devices whose IP addresses have the same prefix. For example, all devices with IP addresses that start with 100.100.100. would be part of the same subnet. Dividing a network into subnets is useful for both security and performance reasons. IP networks are divided using a subnet mask.

TCP/IP: Transmission Control Protocol/Internet Protocol is the suite of communications protocols used to connect hosts on the Internet. TCP/IP uses several protocols, the two main ones being TCP and IP. TCP/IP is built into the UNIX operating system and is used by the Internet, making it the de facto standard for transmitting data over networks.

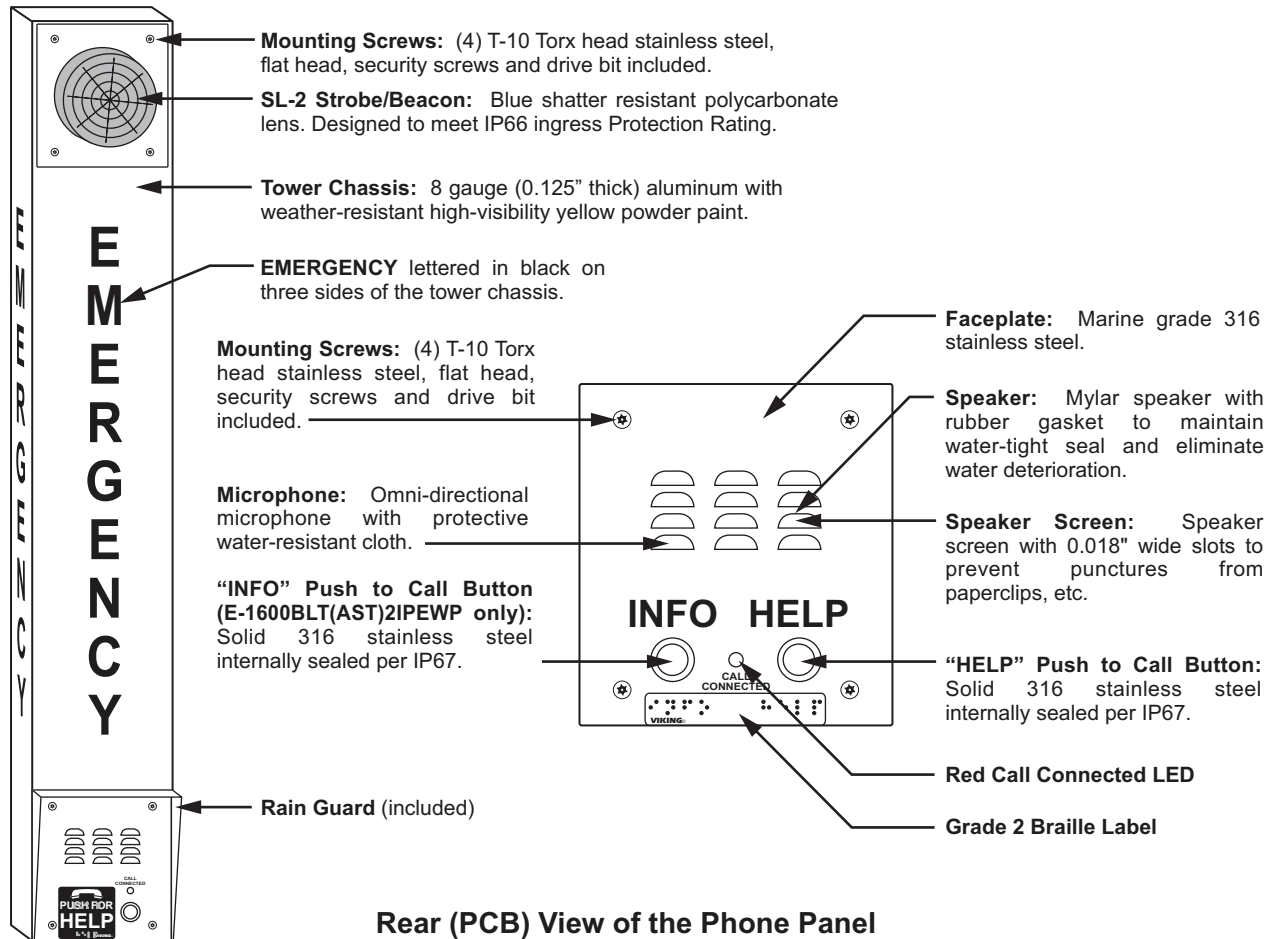
TISP: Telephone Internet Service Provider

WAN: Wide Area Network. A WAN is a network comprising a large geographical area like a state or country. The largest WAN is the Internet.

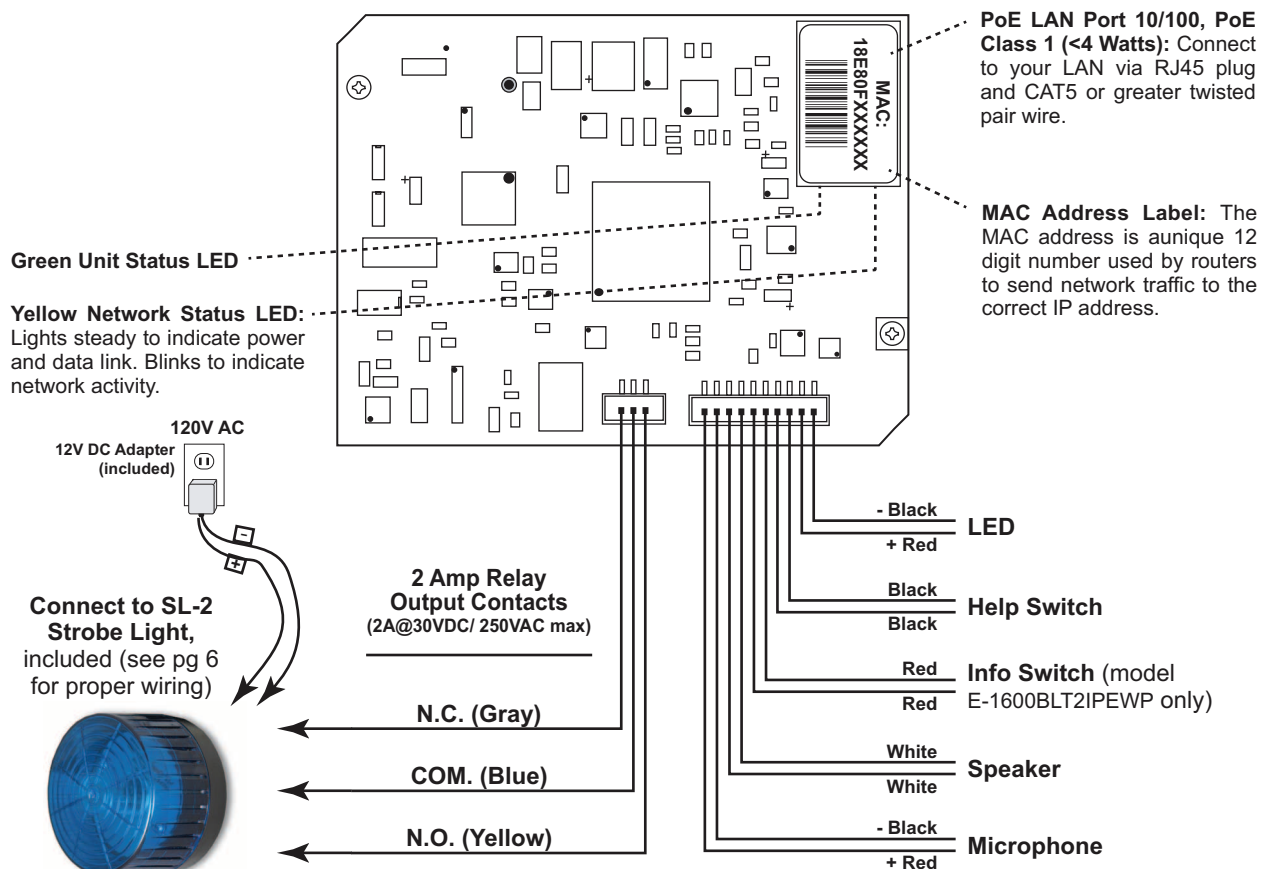
Wireless Access Point (AP): A device that allows wireless devices to connect to a wired network using Wi-Fi, or related standards. The AP usually connects to a router (via a wired network) as a standalone device, but it can also be an integral component of the router itself.

Wireless Repeater (Wireless Range Extender): takes an existing signal from a wireless router or access point and rebroadcasts it to create a second network. When two or more hosts have to be connected with one another over the IEEE 802.11 protocol and the distance is too long for a direct connection to be established, a wireless repeater is used to bridge the gap.

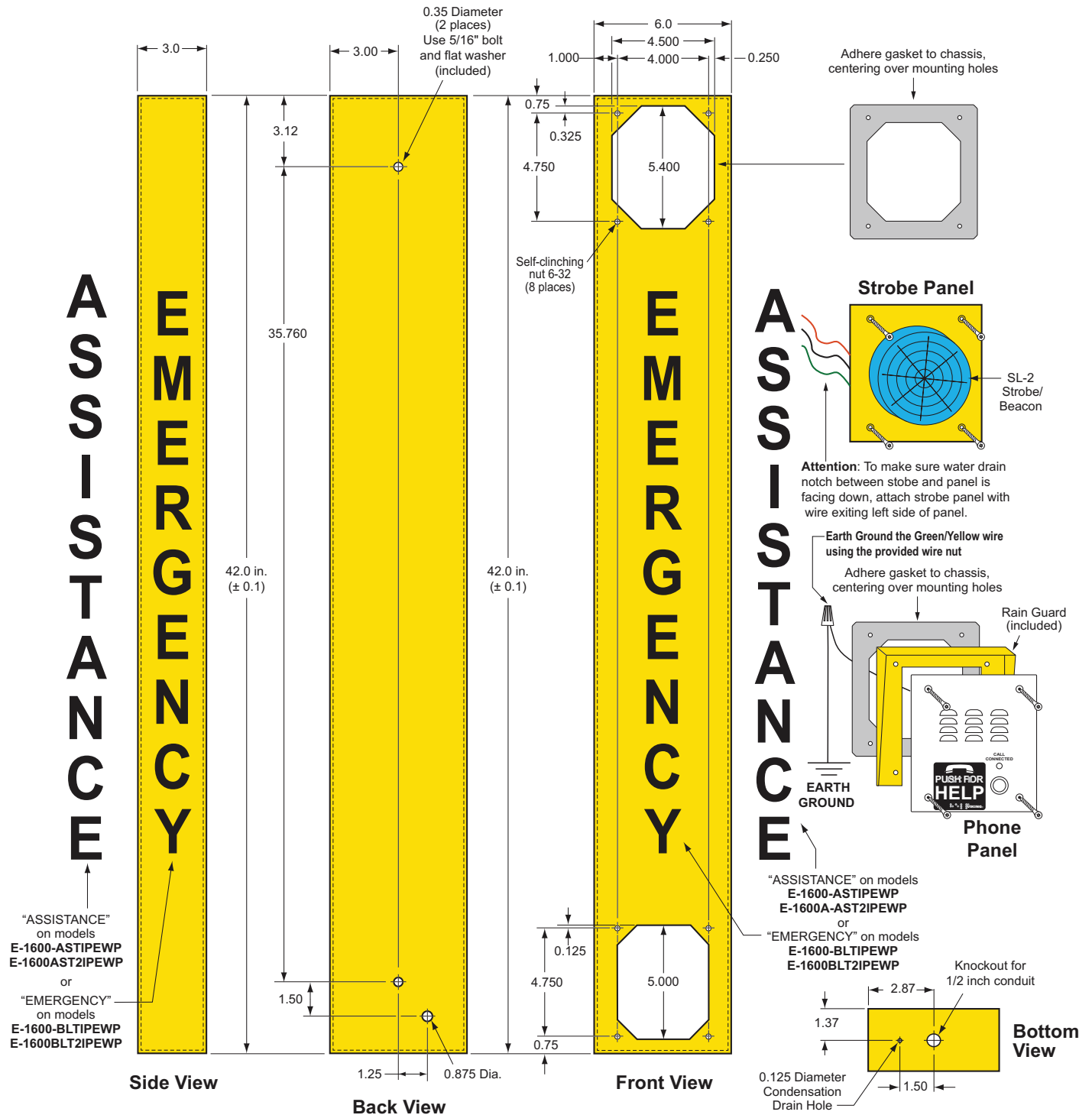
Features Overview



Rear (PCB) View of the Phone Panel



Installation and Specifications



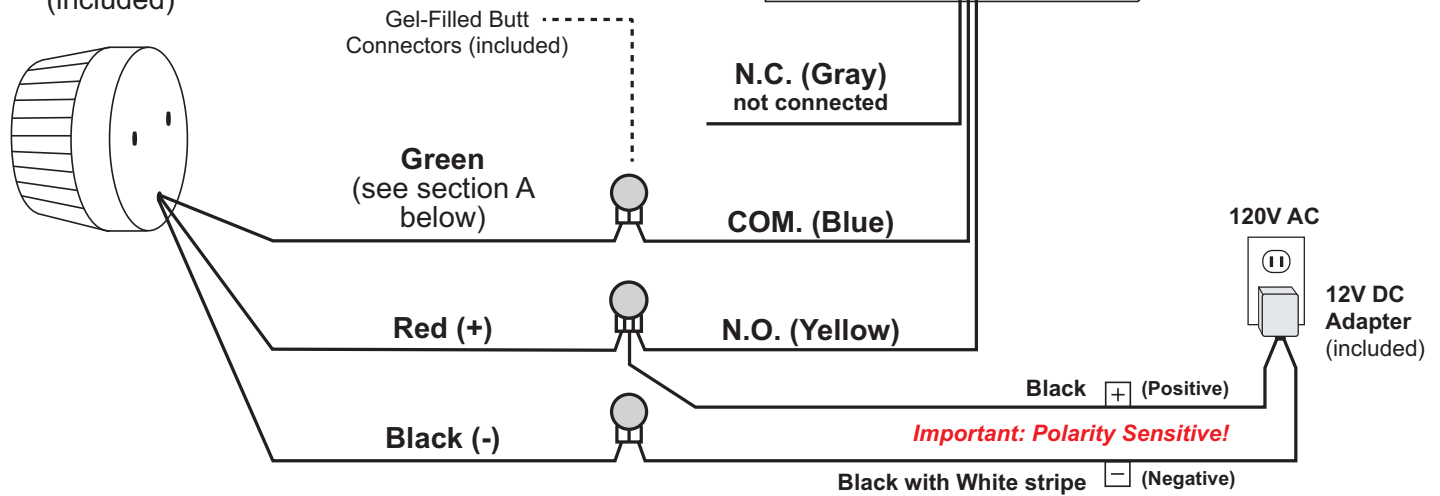
Step 1.	Mount the tower phone approximately 42" above the floor to a flat, sturdy surface using 5/16 hardware. Note: Flat washers should be used on the main mounting bolts for additional strength.
Step 2.	Locate the strobe light panel and attach red, black and green 42" wire extensions with butt connectors then pass the red, black and green wires from the strobe panel through the gasket and the upper hole in the tower.
Step 3.	Mount the strobe panel to the tower using the four security screws provided. Attention: To make sure water drain notch between stobe and panel is facing down, attach strobe panel with wires exiting left side of panel.
Step 4.	Locate the phone panel. Using the gel-filled butt connectors, wire the Strobe Light to the Phone panel as shown on Page 6.
Step 5.	Connect CAT 5 or greater Ethernet cable to RJ45. (See Page 8)
Step 6.	Mount the phone panel to the tower using the remaining four security screws.

Wiring



IMPORTANT: Electronic devices are susceptible to lightning and power station electrical surges from both the AC outlet and the telephone line. It is recommended that a surge protector be installed to protect against such surges.

Connect to SL-2 Strobe Light (included)



Programming the SL-2 Strobe / Beacon

The **SL-2** can be user programmed as a Beacon only, Strobe only or Beacon/Strobe. The brightness setting can be programmed separately for the Strobe or Beacon and one of 4 different Flash Patterns can be programmed for the strobe (see **Strobe Flash Patterns**, on page 7). **Note:** The **SL-2** is factory default programmed as a steady on beacon with a single flash strobe when activated. The beacon and strobe are set to their brightest settings. All programming should be done prior to connecting to the 2 amp relay contacts on the **1600-IP** PCB.

A. Accessing the Programming Mode

(Optional, the **SL-2** is factory programmed to the Beacon/Strobe Mode and brightest Beacon/Strobe settings).

Step 1.	Apply 10-15 VDC power to the Red (+) and Black (-) wires.
Step 2.	Touch and hold the Green (Control) wire to the Black (-) wire for 3 seconds. The strobe should flash twice. You are now in Programming mode. Note: Once in the programming mode, if a programming command has not been entered for 20 seconds the strobe will flash 3 times indicating the unit has exited programming and returned to the Run Mode.

B. Programming Desired Features (after accessing the Programming Mode as shown on page 6)

Select the Feature: Momentarily touch the Green (Control) wire to the Black (-) wire 1 to 10 times to select which feature to program, see Programming Features List to the right. The strobe should momentarily flash each time the Green (Control) wire has touched the Black (-) wire.

Features 1-3 & 6-10: After selecting Programming Features 1-3 or 6-10, wait 3 seconds and the strobe should flash 2 times indicating that feature has been programmed. You can now exit programming or move on to programming the Beacon or Strobe brightness settings.

Setting Strobe Brightness (factory set to 6/Brightest): After selecting Programming Feature 4 (Strobe Brightness), wait 3 seconds and the strobe should begin flashing in the preprogrammed flash pattern. Touch and hold the Green (Control) wire to the Black (-) wire. The **SL-2** will begin flashing the strobe cycling through 6 different brightness settings from lowest to highest. When the **SL-2** flashes at the desired brightness level immediately remove the Green wire from the Black. The **SL-2** should flash twice indicating the selected brightness level has been set.

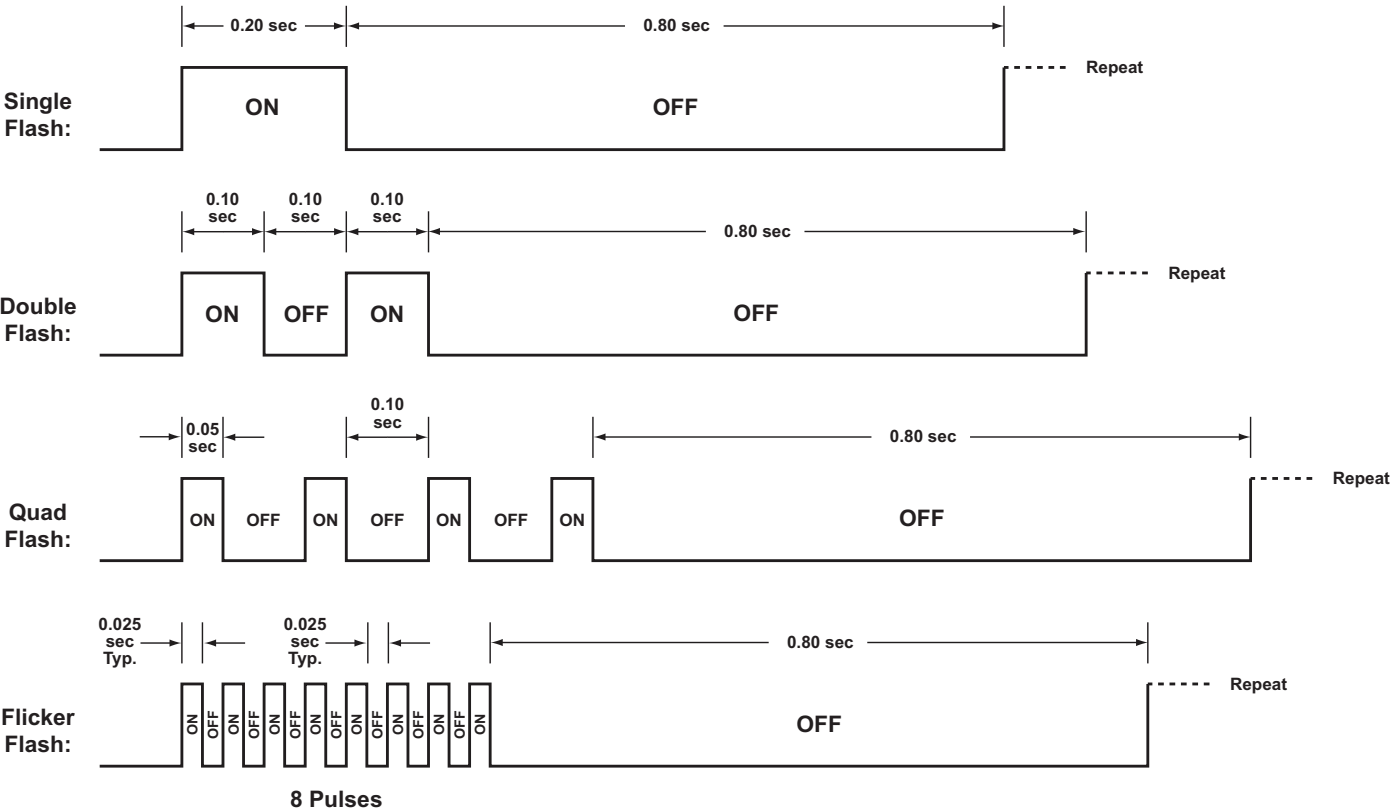
Setting Beacon Brightness (Factory set to 6/Brightest): After selecting Programming Feature 5 (Beacon Brightness), wait 3 seconds and the beacon will light at its preprogrammed beacon brightness setting. Touch and hold the Green (Control) wire to the Black (-) wire. The **SL-2** will begin stepping through 6 different beacon brightness settings from lowest to highest. When the **SL-2** lights at the desired beacon brightness level immediately remove the Green wire from the Black. The **SL-2** should flash twice indicating the selected brightness level has been set.

Exiting Programming: To exit programming simply wait 20 seconds from the last programming command and the unit will flash 3 times. This indicates the **SL-2** has exited the programming mode and is now in the Run mode. **Note:** To eliminate waiting 20 seconds, you can also exit programming after the last programming command by touching and holding the Green (Control) wire to the Black (-) for 3 seconds. The strobe will flash 3 times indicating the unit has exited the programming mode and is now in the Run mode.

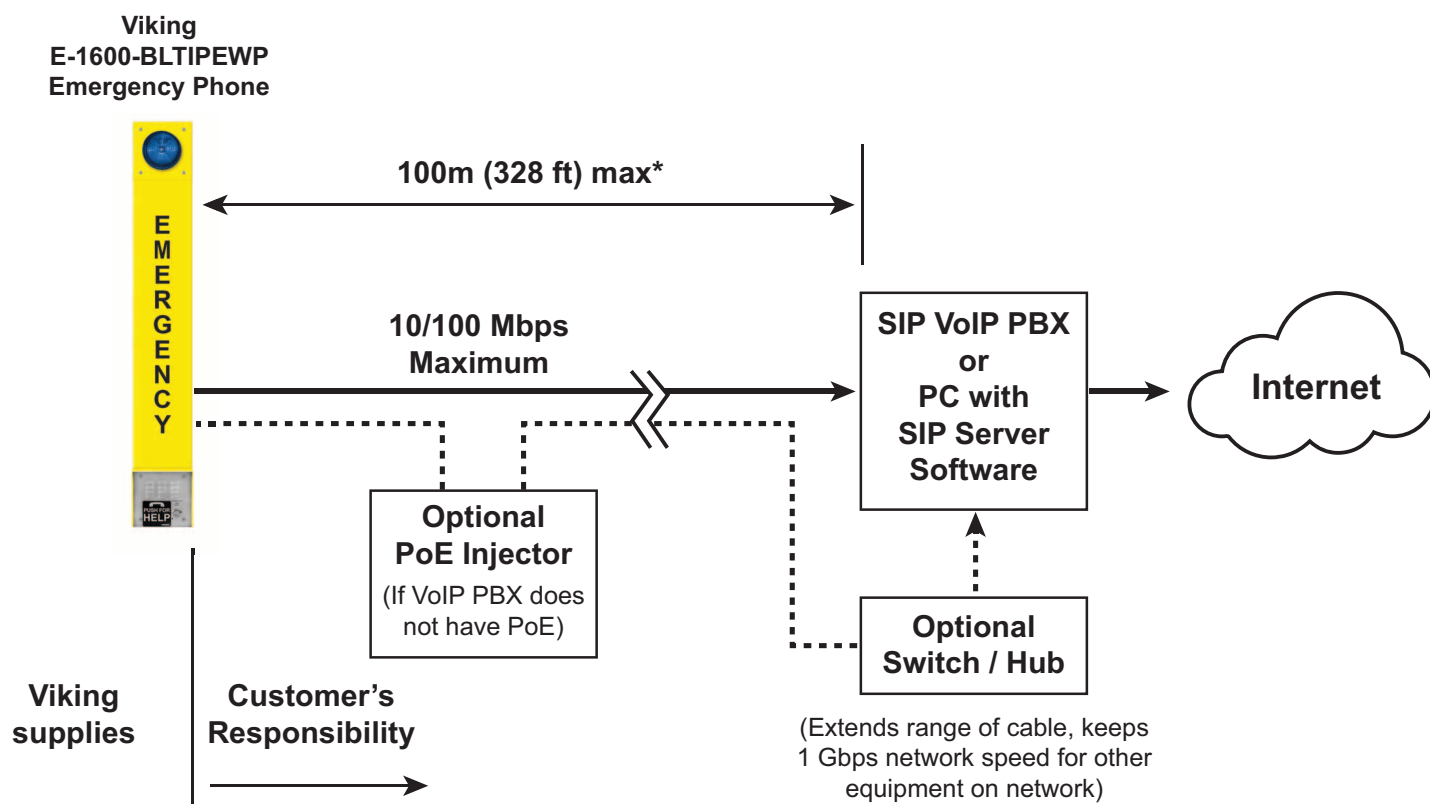
Programming Features List

Feature Number	Description
1	Beacon/Strobe (factory default)
2	Strobe Only
3	Beacon Only
4	Strobe Brightness 1-6 (factory default = 6, brightest setting)
5	Beacon Brightness 1-6 (factory default = 6, brightest setting)
6	Single Flash Strobe (factory default)
7	Double Flash Strobe
8	Quad Flash Strobe
9	Flicker Flash Strobe
10	Reset to Factory Default Settings

C. Strobe Flash Patterns



Typical Installation on SIP Based VoIP Phone System



*** Note:** A PoE extender can be used for an additional 100 meters per extender. For longer runs (up to 2 km / 1.2 miles) a ethernet to fiber media converter can be used.

PC Requirements

- **IBM** compatible personal computer with: Windows 7, 8, 10 or 11
- Adobe Acrobat Reader 8 or higher
- **VoIP Tower Phone** hardware
- Available LAN with PoE (class 1, < 4 watts)
- Ethernet cable (CAT5 min.)
- 1 MB minimum free hard drive space for installation
- 16MB of free physical RAM

PC Programming

Download and install the programming software

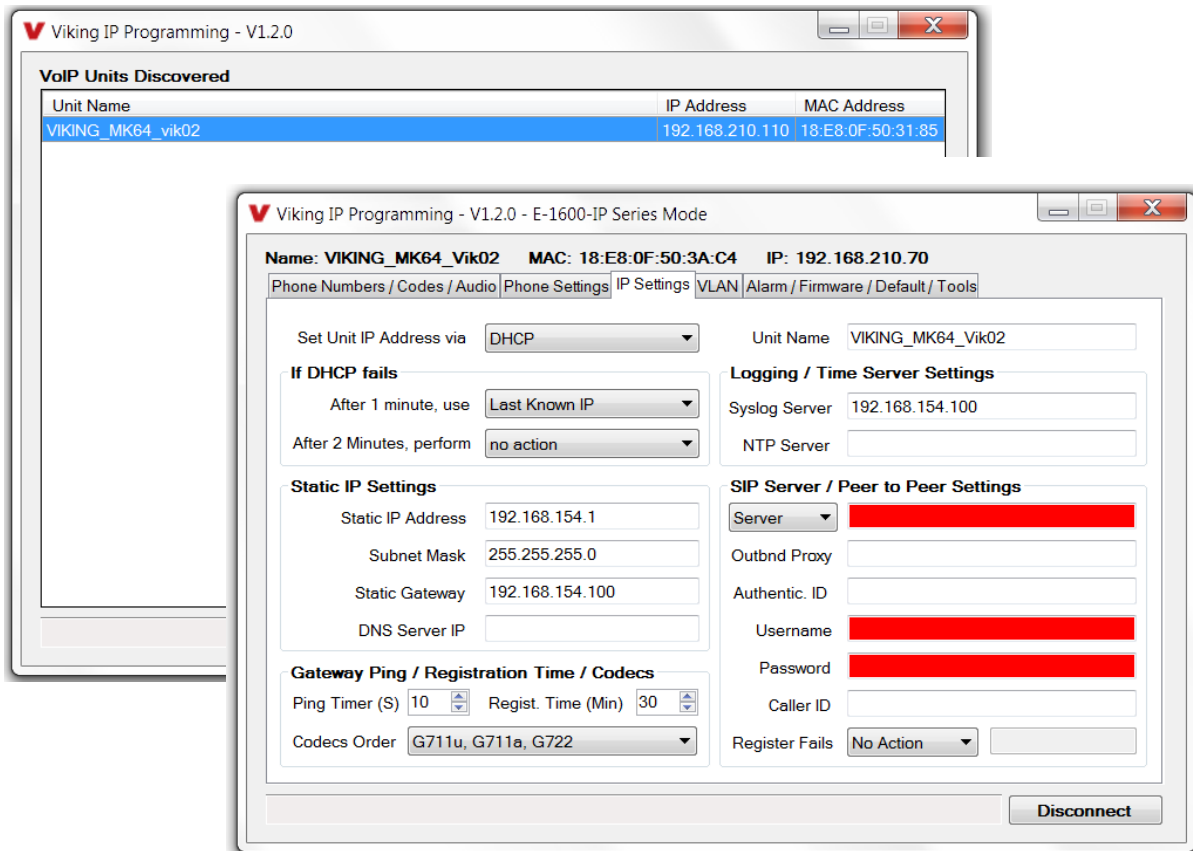
1. Go to www.vikingelectronics.com and enter **E-1600-BLTIPEWP** in the search box
2. Click **E-1600-BLTIPEWP** in the search results
3. Scroll down the page to Downloads, click **IP Programming Software**
4. Install the programming software by saving or opening the file and then clicking on **setup Viking IP Programming.exe**
5. Follow the prompts on your screen to complete software installation
6. To start the Viking IP Programming application, click on the Viking IP Programming icon on your desktop. The Main screen will appear, allowing the user to program any **VoIP Tower Phone** connected to that LAN.

Note: PC must be connected to the same LAN as the **VoIP Tower Phone**.

A. Connect / Disconnect

Open the “Viking IP Programming” software on the PC and the start screen shown below will appear. Any Viking IP phones that are connected to the network will appear on the list. Simply select the **VoIP Tower Phone** on the list and click on the “Connect” button at the bottom or double click the selected phone. If the security code of the selected phone is still set to default (845464), the PC software will not require entering a security code to connect to the phone. **VoIP Tower Phones** have a default name of “VIKING_MK64_Vik02”, so if many phones are connected to the same network that all have the default name, MAC addresses must be used to identify each phone.

When finished programming, click on the “Disconnect” button at the bottom. Closing the program will also automatically disconnect the unit.



B. Manually Muting SIP/Network Failure Alarm Beeps (3 beeps repeated every 30 seconds)

With the unit connected and powered (Green LED on and Yellow LED off or blinking) it will output 3 beeps every 30 seconds and turn the Call/Call Connected LED on and off once per second indicating a SIP registration failure, failure to receive an echo reply from a pinged gateway or Ethernet connection failure. You can manually disable the beeps by pressing and holding the Call button for 5 seconds (2 beeps will then be heard) or by clicking the “Mute Alarm Until Next Failure” tab in the Viking IP programming software. The LED will continue to flash allowing you to trouble shoot the failure.

C. Configuring the VoIP Tower Phone Network Settings

Step 1.	Open the “Viking IP Programming” software on a windows PC that is connected to the same LAN as the VoIP Tower Phone to be programmed.
Step 2.	The window in the upper left corner of the menu will show you each VoIP Tower Phone that is connected to that LAN. Select the unit with the same MAC address shown on the label located on the top of the Ethernet connector on the VoIP Tower Phone .
Step 3.	Click on the “connect” button. A default VoIP Tower Phone has no username or password programmed, the security code is set to “845464” and when configured this way, you are instantly connected to the VoIP Tower Phone . If the VoIP Tower Phone has been programmed with a username and password, a pop up window will ask for the username followed by the password. If the unit’s security code has been changed from “845464”, it will then prompt for the correct security code, then click “retry”.
Step 4.	The program will then read and display the VoIP Tower Phone ’s IP and programming settings.
Step 5.	Click on the “IP Settings” tab.
Step 6.	Select the appropriate value Static IP Settings or DHCP for “Set Unit IP Address via”. Note: changing the IP address will cause you to have to reconnect to the unit. Enter the values for the fields in “if DHCP fails” or “Static IP Settings” as needed.
Step 7.	Set the “Unit Name”, “Logging / Time Server Settings” as needed.
Step 8.	Select Peer-Peer in the “SIP Server / Peer to Peer Settings” to use the unit in Peer to Peer mode. Select Server to register with a SIP registrar server and fill in the “Outbnd Proxy” (SIP Outbound Proxy Server Address, “ip:port”), “Authentic. ID” (SIP Authentication ID), “Username” (SIP Username, <string>), “Password” (SIP Password), and “Caller ID” (SIP Caller ID) with values from your VoIP provider.

Viking IP Programming - V1.2.0 - E-1600-IP Series Mode

Name: VIKING_MK64_Vik02 MAC: 18:E8:0F:50:3A:C4 IP: 192.168.210.70

Phone Numbers / Codes / Audio | Phone Settings | IP Settings | VLAN | Alarm / Firmware / Default / Tools

Set Unit IP Address via: DHCP Unit Name: VIKING_MK64_Vik02

If DHCP fails

After 1 minute, use: Last Known IP

After 2 Minutes, perform: no action

Static IP Settings

Static IP Address: 192.168.154.1

Subnet Mask: 255.255.255.0

Static Gateway: 192.168.154.100

DNS Server IP:

Logging / Time Server Settings

Syslog Server: 192.168.154.100

NTP Server:

SIP Server / Peer to Peer Settings

Server: [Redacted]

Outbnd Proxy:

Authentic. ID:

Username: [Redacted]

Password: [Redacted]

Caller ID:

Register Fails: No Action

Gateway Ping / Registration Time / Codecs

Ping Timer (S): 10 Regist. Time (Min): 30

Codecs Order: G711u, G711a, G722

Disconnect

Example 1: On-Premise SIP Phone System (Panasonic TDE 100/200)

Example 2: Cloud Based Service Provider (Voip.ms)

Example 3: Cloud Based Service Provider requiring Outbound Proxy and Authentication ID (Ring Central)

SIP Server / Peer to Peer Settings

Server: 192.168.0.101

Outbnd Proxy:

Authentic. ID:

Username: 117

Password: 9140

Caller ID: Door 1

Register Fails: Re-Resolve

SIP Server / Peer to Peer Settings

Server: chicago4.voip.ms

Outbnd Proxy:

Authentic. ID:

Username: 190106

Password: Bear987654!

Caller ID: Door 1

Register Fails: Re-Resolve

SIP Server / Peer to Peer Settings

Server: sip.ringcentral.com:5060

Outbnd Proxy: sip20.ringcentral.com:5090

Authentic. ID: 16572241020

Username: 17159644561

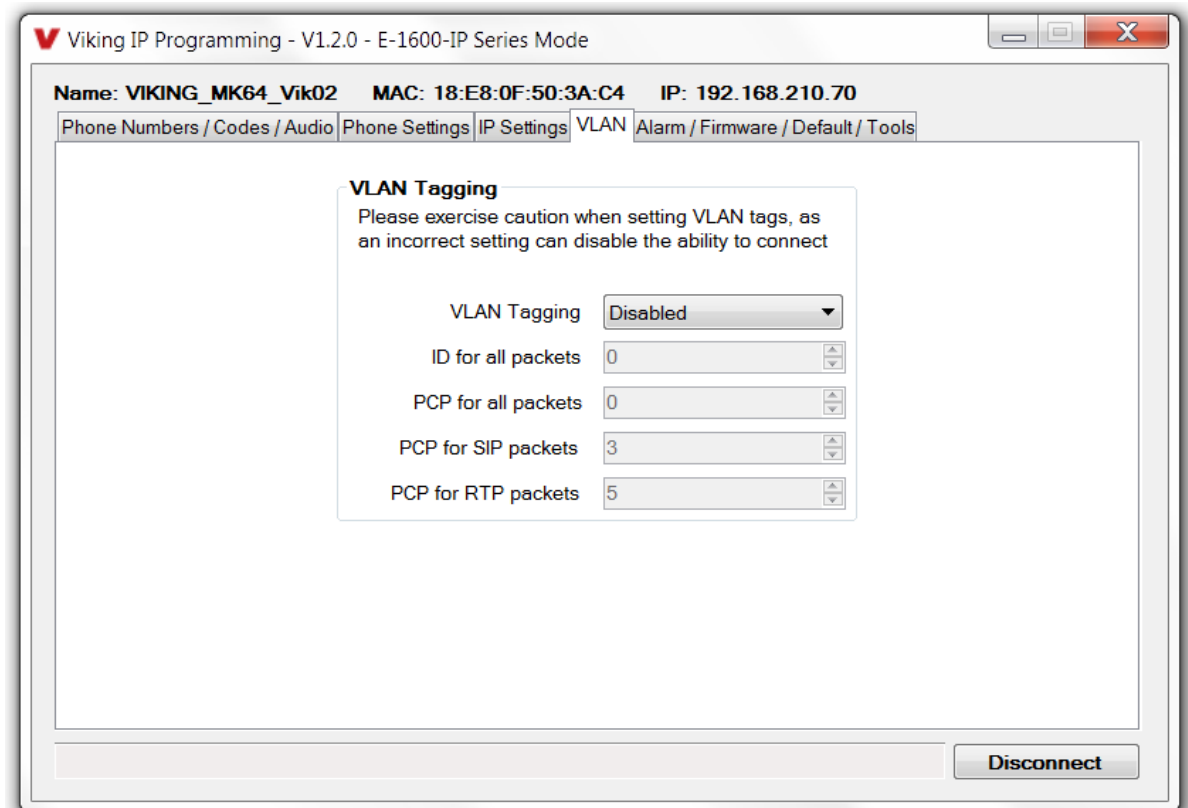
Password: 5j8QA28Lp

Caller ID: Door 1

Register Fails: Re-Resolve

D. Configuring VoIP Tower Phone VLAN Settings

Step 1.	Click on the "VLAN" tab
Step 2.	Disable or enable VLAN tagging by setting the value of "VLAN Tagging".
Step 3.	Set the VLAN tag ID by selecting an integer (1 to 4094) in "ID for all packets".
Step 4.	Set the Priority Code Point (PCP) value for all not SIP and RTP packets in the "PCP for all packets" input (0 is default, priorities are from low to high: 0, 1, 2, 3, 4, 5, 6, 7). Set the "PCP for SIP packets" (3 is default). Set the "PCP for RTP packets" (5 is default).



E. Manually Resetting the Security Code to Enter Programming

Step 1.	Power down the VoIP Tower Phone by disconnecting the LAN Cable (RJ45 plug).
Step 2.	Press and hold the call button, then reconnect the LAN Cable (RJ45 plug).
Step 3.	Continue to hold Call button until you hear 2 beeps, (approximately 6 seconds). Then release the button. The LED will remain off for the first 3 seconds, flash slowly for 3 seconds then fast flash (after 2 beeps), indicating when to release button.
Step 4.	The security code is now reset to 845464 (factory default).
Step 5.	You can now enter programming by following the steps in section A.

F. Manually Resetting All Parameters to Factory Default (IP and phone settings back to default. Phone numbers, paging sources and audio files erased.)

Step 1.	Power down the VoIP Tower Phone by disconnecting the LAN Cable (RJ45 plug).
Step 2.	Press and hold the Call button, then reconnect the LAN Cable (RJ45 plug).
Step 3.	Continue to hold the Call button until you hear 2 beeps, (approximately 6 seconds). Continue to hold reset button until you hear 4 more beeps, approximately 6 seconds later, then release the button. The LED will remain off for the first 3 seconds, flash slowly for 3 seconds (2 beeps), fast flash for 6 seconds (4 beeps), then light steady indicating when to release button.
Step 4.	You can now enter programming by following the steps in section A.

Programming Features Index

DESCRIPTION	Section	Page
Connect/Disconnect	A	9
VLAN Settings	D	11
Unit Name	1	13
SIP Server	2	13
Peer to Peer Settings	3	13
Outbound Proxy	4	13
Authentication ID	5	13
Register Fails	6	13
Pre Page Alert Tone	7	14
Multicast Page Type	8	14
SIP Page / Call Priority VS Multicast	9	15
Multicast Paging	10	16
Multicast Paging Volume (0 - 19, factory set to 1)	11	16
Speed Dial Numbers	12	17
Security code (factory set to 845464)	13	17
ID Number	14	17
Access Code (1 - 6 digits, blank = disabled, factory set to 123456)	15	17
Audio File	16	18
Recording Emergency Announcement from a phone	17	18
Internal / External Relay (factory set to Internal)	18	18
Relay Mode (Door Strike, Outbound Call, Inbound / Outbound Call, Doorbell, LV-1K Control, Ring, Ring Flash, factory set to Outbound Call)	19	19
Relay Activation Command (1 or 2 digits, factory set to **) NOTE: Relay Mode must be set to Door Strike	20	19
Relay Activation Time (0.5 - 99 sec, factory set to 5 sec)	21	19
Relay Buzz Volume (1 - 3 or Disabled, factory set to 2)	22	19
Relay Latch Commands (Enabled or Disabled, factory set to Enabled) NOTE: Relay Mode must be set to Door Strike	23	19
Alternating Switch Action (factory set to Enabled)	24	20
Speaker Mode (ON, OFF / Silent Monitor or OFF until Answered, factory set to ON)	25	20
Speaker Volume (0 - 19, factory set to 7)	26	20
Ring Volume (0 - 19, factory set to 5)	27	20
Microphone Volume (Auto, 1 - 9, factory set to 5)	28	20
Talk/Listen Delay (VOX) (0.1 - 0.9 seconds, factory set to 0.5 seconds)	29	20
In-Band Audio Call Progress (Enabled, Disabled, factory set to Enabled)	30	20
In-Band Audio Detect Sensitivity (1 - 9, 1 = minimum, 9 = maximum, factory set to 5, power cycle unit after setting)	31	20
Repeat Announcement Option (factory set to 1)	32	20
Lap Counter (1 - 9 or Disabled, factory set to Disabled)	33	21
SIP Call Length Time Out (disabled or 1 - 9 minutes, factory set to 3 minutes)	34	21
Multicast Page Length Time Out (1 - 255 seconds, factory set to 180 seconds)	35	21
Inbound Call Mode (Disabled, Auto Answer, Auto Answer-Secure, Ring, Ring with AGC, Factory set to Auto Answer)	36	21
Ring Cadence (factory set to Normal Ring, 2 seconds on 4 seconds off)	37	21
Dial Next Number on RNA (Ring No Answer) (disabled, 1 - 9 = number of rings, factory set to 7)	38	22
Dial Next Number on Busy (disabled or enabled, factory set to enabled)	39	22
Send ID Numbers as (RFC 2833 or In Band DTMF, factory set to RFC 2833)	40	22
Call LED Mode (OFF, Entry Phone, Emergency Phone or Emergency Outbound only, factory set to Emergency Phone)	41	22
Call LED Control (Automatic or enter * to light, factory set to Automatic)	42	22
Mute Current / Next Alarm	43	23
Permanent Alarm Mute (factory set to Alarm Tones Enabled)	44	23
Programming a Username and Password to Restrict Access to Programming	45	23

DESCRIPTION	Section	Page
IP Firmware	46	23
Unit Firmware	47	23
Import/Export	48	23
Clear Phone Settings	49	23
Clear IP Settings	50	23
Diagnostics (used to check mic, speaker and relay operations)	51	23
Resetting Username and Password	52	23

Programming Features

1. Unit Name

Up to a 31 character unit name can be assigned to the **VoIP Tower Phone** being programmed.

2. SIP Server

Enter the IP address or URL of your SIP server or service provider in this field. The SIP server IP address is limited to 74 characters. **Note:** *If an alternate SIP server IP address is programmed, the IP address for the SIP server and alternate SIP server will be limited to 31 characters.* **Note:** *If outbound proxy is not required, enter the SIP server IP address into the Outbnd Proxy field.*

3. Peer to Peer Settings

When set to Peer to Peer mode, a SIP server is not used. The unit should be programmed with a Static IP Address and Username, a password is not used. Caller ID can be programmed if needed. Simply call the unit by entering the programmed "username@192.168...(Static IP address for the unit)". The static IP address is normally programmed into a page button on the VoIP telephones. peer to Peer will only work when both devices are on the sam LAN. An IPV4 address is used in the dialing string, and must be reachable for the **VoIP Tower Phone**.

4. Outbound Proxy

If your SIP provider requires an outbound proxy IP address enter it in the Outbnd Proxy field. If outbound proxy is not required enter the SIP sever IP address into the Outbnd Proxy field. **NOTE:** *If not required, this field must match your SIP server IP address.*

5. Authentication ID

If your SIP provider requires Authentication ID, enter it in the Authentic. ID field. If Authentication ID is not required, leave this field blank.

6. Register Fails (Re-Resolve or Alternate Server)

When registered to a SIP server in the event that registration is lost you can program the unit to re-resolve using the current SIP server IP address or route pages through an alternate SIP server. With Alternate Server selected enter the IP address of the alternate SIP server in the field next to the Register Fails drop down box. Add an Alternate Proxy address as well if Proxy is used. When Registration Fails is set to Re-Resolve or Alternate Server, the unit will reboot after two Minutes and Fifty seconds of SIP / Network Alarm. Alternate Server mode alternates between primary and alternate servers each reboot. **Note:** *With an alternate SIP server IP address programmed, the IP address for the SIP server and alternate SIP server will be limited to 31 characters.*

Standard

Viking IP Programming - V1.4.0 - E-1600-IP(PG) Series Mode

Name: E16PG MAC: 18:E8:0F:50:55:4A IP: 192.168.210.242

Paging Phone Numbers/ Codes/ Audio Phone Settings IP Settings VLAN Alarm / Firmware / Info / Tools

Multicast Settings

Pre Page Tone Enabled

Multicast Page Type Standard

SIP Call Priority vs Multicast

SIP In Above 0 SIP Out Above 0

Paging Sources (Multicast IP and Port) - Volume Level - Timeout Enable/Disable

0		1	Vol 1	TO	<input checked="" type="checkbox"/>	5		1	Vol 1	TO	<input checked="" type="checkbox"/>
1		1	Vol 1	TO	<input checked="" type="checkbox"/>	6		1	Vol 1	TO	<input checked="" type="checkbox"/>
2		1	Vol 1	TO	<input checked="" type="checkbox"/>	7		1	Vol 1	TO	<input checked="" type="checkbox"/>
3		1	Vol 1	TO	<input checked="" type="checkbox"/>	8		1	Vol 1	TO	<input checked="" type="checkbox"/>
4		1	Vol 1	TO	<input checked="" type="checkbox"/>	9		1	Vol 1	TO	<input type="checkbox"/>

Apply Disconnect

7. Pre Page Tone

When enabled, a short beep will be heard prior to SIP or Standard Multicast paging audio. The volume of the Pre Page Alert Tone will match the volume setting of the SIP or Standard Multicast page. **Factory Setting:** Enabled

8. Multicast Page Type

Select Standard or Polycom as the source of multicast paging. If Polycom is selected as the source, the screen will change to show its options (see below). **Factory Setting:** Standard (shown above)

Polycom

Viking IP Programming - V1.4.0 - E-1600-IP(PG) Series Mode

Name: E16PG MAC: 18:E8:0F:50:55:4A IP: 192.168.210.242

Paging Phone Numbers/ Codes/ Audio Phone Settings IP Settings VLAN Alarm / Firmware / Info / Tools

Multicast Settings

Pre Page Tone Enabled

Multicast Page Type Standard

SIP Call Priority vs Multicast

SIP In Above 0 SIP Out Above 0

Paging Sources (Multicast IP and Port) - Volume Level - Timeout Enable/Disable

0		1	Vol 1	TO	<input checked="" type="checkbox"/>	5		1	Vol 1	TO	<input checked="" type="checkbox"/>
1		1	Vol 1	TO	<input checked="" type="checkbox"/>	6		1	Vol 1	TO	<input checked="" type="checkbox"/>
2		1	Vol 1	TO	<input checked="" type="checkbox"/>	7		1	Vol 1	TO	<input checked="" type="checkbox"/>
3		1	Vol 1	TO	<input checked="" type="checkbox"/>	8		1	Vol 1	TO	<input checked="" type="checkbox"/>
4		1	Vol 1	TO	<input checked="" type="checkbox"/>	9		1	Vol 1	TO	<input type="checkbox"/>

9. SIP Page / Call Priority vs Multicast

Incoming and outgoing non-emergency SIP call audio streams can be programmed to have priority over any of the 10 standard multicast paging groups. Use the “SIP In Above” and “SIP Out Above” drop down menus to select which multicast groups you want SIP calls to have priority over. “SIP In Above” is to set the priority of incoming SIP pages and “SIP Out Above” is to set the priority of outgoing non-emergency SIP calls. Regardless of the priorities selected, outgoing SIP emergency calls have the highest priority and will interrupt any multicast source.

Factory Setting for both: 0 (SIP is highest priority)

When Polycom is selected as the Multicast Page Type, the “SIP In Above” and “SIP Out Above” priorities can be set to “Poly” or “None”. See the following tables for the order of priorities with each setting. As shown, outgoing SIP emergency calls have the highest priority and Polycom Group 25 multicast has the second highest priority. The priority options allow you to determine the priorities for Polycom Groups 1 through 24 multicast signals, SIP paging and non-emergency outbound SIP calls. **Factory Setting for both:** Poly

Paging Priorities (SIP In Above and SIP Out Above both set to “Poly”)	Paging Priorities (SIP In Above and SIP Out Above both set to “None”)
Emergency Outbound SIP Calls	Emergency Outbound SIP Calls
Polycom Group 25	Polycom Group 25
Non-emergency Outbound SIP Calls and SIP paging	Polycom Groups 1-24
Polycom Groups 1-24	Non-emergency Outbound SIP Calls and SIP paging
Line In Audio Source	Line In Audio Source

Paging Priorities (SIP In Above set to “None” and SIP Out Above set to “Poly”)	Paging Priorities (SIP In Above set to “Poly” and SIP Out Above set to “None”)
Emergency Outbound SIP Calls	Emergency Outbound SIP Calls
Polycom Group 25	Polycom Group 25
Non-emergency Outbound SIP Calls	SIP Paging
Polycom Groups 1-24	Polycom Groups 1-24
SIP Paging	Non-emergency Outbound SIP Calls
Line In Audio Source	Line In Audio Source

10. Standard or Polycom Multicast Paging

Standard

When using standard Multicast Paging, up to 10 multicast paging groups can be programmed into each **VoIP Tower Phone**. Each multicast group is defined by a multicast address and port number. Each multicast group is assigned a priority, allowing simultaneously arriving pages to be serviced based on importance. A timeout check box can be selected for each paging group. See Multicast Page Length Time Out under Speaker Settings to adjust the timeout from 1 - 255 seconds. Only one timeout time can be programmed for all groups. The volume for each multicast group can be programmed separately.

Assigning Priority to standard Multicast Paging

VoIP Tower Phone will prioritize simultaneous audio streams according to their priority in the Paging Sources list. Group 0 will have the highest priority while group 9 will have the lowest priority. Group 9 is useful for a low priority stream such as background music. Group 0 is useful for high priority streams such as emergency messages.

The multicast paging groups can also be used for up to ten different paging zones for receiving audio streams. A paging zone can consist of one or many **VoIP Tower** multicast phones. There is no limit to how many speakers can be in a given paging zone. Each multicast group is defined by a multicast address and port number. Each multicast group is assigned a priority, allowing simultaneously arriving pages to be serviced based on importance.

Polycom

Polycom IP phones send multicast audio to a specific IP address and port. A total of 25 groups can be used, with group 25 generally reserved for an emergency broadcast. Set the Multicast Page Type to "Polycom". The Viking IP programming page will then show "Polycom Paging Source and Group Subscriptions". Polycom phones will default to IP address 224.0.1.116 and port 5001 for multicast. Under "Polycom Paging Source and Group Subscriptions", enter the IP address and port you chose in the Polycom IP phone as "Multicast IP address" and "port" (Paging/PTT Configuration page). Select the paging groups the **VoIP Tower Phone** should subscribe to. Any multicast audio sent to the correct IP address, port number and a subscribed group will be heard from both the Speaker and Line Out terminals. Group 1 is the lowest priority and group 25 is the highest. The Volume and Timeout can be selected globally, although the timeout does not affect group 25. **Factory Setting:** all Groups selected

11. Multicast Paging Volume

The volume of each individual standard multicast page can be adjusted from 0 - 19. When polycom paging is used, the paging volume is set globally for all 25 groups. **Factory Setting:** 1

Note: A majority of the features below can also be Touch Tone (In-Band DTMF) programmed, see DOD 949.

12. Speed Dial Phone Numbers

Note: Up to 90 digits can be stored in each of the 5 speed dial phone number positions.

The number programmed in the first location under “Speed Dial Numbers” is the telephone or extension number that is dialed when the call button is pressed. Additional numbers (if programmed) will be dialed when there is no answer and the Dial Next No. on Ring No Answer feature is enabled. The **VoIP Tower Phone** will also detect busy and move on (Dial Next Number on Busy Enabled). The **VoIP Tower Phone** will cycle through the programmed Speed Dial numbers until answered.

13. Security Code

The security code allows the user/installer to program the **VoIP Tower Phone**. It is recommended that the factory set security code be changed. **Factory Setting:** 845464

Note: The security code must be 6 digits and cannot include a * or a #.

14. ID Number:

The ID Number (1 - 6 digits) is used by emergency personnel to identify the location of the caller and is given out when the receiving party presses a *. This ID number is transmitted as In-Band DTMF. This can be cleared out by leaving the field blank. **Factory Setting:** 987654

15. Access Code

The Access Code is used for remotely operating the relay (Doorstrike, Mag-Lock, etc) by calling into the unit. This code provides basic security and only allows operation of the relays and not the ability to change any of the programming parameters. Once entered, any of the “Remote Access Operation Commands” can be used. The code can be 1 - 6 digits in length and cannot contain a “*” or “#” or match the numbers used for the security code. Simply call the **VoIP Tower Phone** (set to auto-answer/auto-answer secure), the unit will automatically answer the line and output one beep. You then enter the programmed 1 - 6 digit access code, 2 beeps should be heard. You can now enter any “Remote Access Operation Commands” (see page 25). This code will also enable audio to/from the speaker and the caller. The access code can be cleared (by leaving the field blank) if this additional level of security is not required.

Factory Setting: 123456

16. Audio File (optional)

The **VoIP Tower Phone** Emergency Announcement can be uploaded with any user edited WAV file. See “WAV File Guidelines” below. Alternatively the announcement can be recorded via a telephone call (See section 17 below).

WAV File Guidelines:

1.	The WAV file should be 8 or 16 bit PCM mono or stereo.
2.	Sampling rates of 8k,16k or 32kHz are acceptable.
3.	The WAV file may be stereo or multi-channel, but only the Left Channel will be loaded.
4.	When saved, wave files will be converted to 8KHz, mono, 16 bit, PCM.
5.	The internal flash memory can hold up to 28 seconds of recorded audio.

Note: Applications such as Sound Forge, Audacity, and Windows Sound Recorder can be used to create the WAV files.

17. Recording Emergency Announcement from a Phone

Step 1.	Call into the 1600-IP Series with a Touch Tone phone and access programming by entering the security code (845464 is the factory default).
Step 2.	Enter *4, wait for the tone and then begin recording (28 seconds of record time is available).
Step 3.	Enter # to stop the recording. Playback (for reviewing) is automatic.
Step 4.	Enter *5 to review the announcement again.
Step 5.	If you choose to not use a voice announcement, enter *3 to clear the recording.

Example: “Hudson Elementary School, classroom number 216 needs assistance. Press the star (*) key on your telephone to hear this announcement again.”

Viking IP Programming - V1.4.0 - E-1600-IP(PG) Series Mode - Custom Audio U...

Name: E16PG MAC: 18:E8:0F:50:55:4A IP: 192.168.210.62

Paging Phone Numbers / Codes / Audio Phone Settings IP Settings VLAN Alarm / Firmware / Info / Tools

Internal / External Relay: Internal In-Band Audio Call Progress: Enabled

Relay Mode: Outbound Call In-Band Audio/Vox Detect Sens.: 5

Relay Activation Command: ** Repeat Announcement Option: 1

Relay Activation Time: 5 sec Lap Counter: Disabled

Relay Buzz Volume: 2 SIP Call Length Timeout: 3 min

Relay Latch Commands: Enabled Multicast Page Length Time Out: 180 sec

Alternating Switch Action: Enabled Inbound Call Mode: Auto Answer

Speaker Mode: On Ring Cadence: Normal Ring

Speaker Volume: 7 Dial Next No. on Ring No Answer: 7

Ring Volume: 5 Dial Next Number on Busy: Enabled

Microphone Volume: 5 Send ID Number as: RFC 2833

Talk / Listen Delay (VOX): .5 sec LED Mode: Emergency Phone

"Call" LED Control: Automatic

Apply Disconnect

18. Relay Internal / External

With the relay set to “Internal” the **VoIP Tower Phone** will activate its on board relay for door strike / gate control. The Relay should be set to “External” for higher security installations when using a Viking remote model **RC-4A** relay controller to activate the door strike / gate controller. **Factory Setting:** Internal

19. Relay Mode

Door Strike Mode. When programmed for Door Strike Mode the relay will momentarily activate for the preprogrammed relay activation time after detecting the correct relay activation command (one or two digit touch tone) from the called party.

Outbound Call Mode. When programmed for Outbound Call Mode the relay will activate continuously for the duration of any outbound call from the Emergency/Entry phone. This mode is useful for activating strobe lights for Emergency VoIP phones.

Phone / Paging Mode. When programmed for Phone / Paging Mode the relay will activate continuously for the duration of any SIP or multicast page or any inbound or outbound call to or from the Emergency/Entry phone. This mode is useful for turning on IR flood lights, VoIP phones with cameras, etc.

Doorbell Mode. When programmed for Doorbell Mode the relay will momentarily activate the relay for the preprogrammed relay activation time on any outbound call from the Emergency / Entry phone. This mode is useful for activating a door chime, etc. When activating door chimes, a 0.5 - 1 second relay activation time is recommended.

Alarm Mode. When programmed for Alarm Mode the relay will activate continuously while the Emergency / Entry phone is powered and registered to the SIP server. In the event the unit loses power and/or SIP registration the relay will turn off. The relay can be used to activate the **LV-1K's** flashing LED and audible beep signals.

Ring Mode. When programmed for Ring Mode the relay will continuously activate while the ringing extension is called. This mode is useful for activating a Viking model **SL-2** strobe light, etc.

Ring Flash Mode. When programmed for Ring Flash Mode the relay will momentarily turn on and off in a 400ms on/off cadence while the ringing extension is called. This mode is useful for activating a Viking **LPL-1** Remote Visual Indicator, etc.

Factory Setting: Outbound Call

20. Relay Activation Command

The one or two digit code stored in the Relay Activation Command is the touch tone command that the person being called must enter on their phone in order to momentarily activate the relay to control a doorstrike, mag-lock, gate controller, or other device. The code can contain the characters 0 - 9, # or *. The code cannot match a relay latching command (*1, *0). The code must be entered while the remote phone is communicating with the Speaker phone.

Factory Setting: **

21. Relay Activation Time

The value stored in the Relay Activation Time is the amount of time the relay will be energized after a correct momentary touch tone command is entered. This number can range from 0.5 - 99 seconds. This also affects timing in Doorbell Mode. **Factory Setting:** 5 seconds

22. Relay Buzz Volume

The relay activation tone is a buzzing sound that is heard from the speaker when the door strike relay is activated. After the called party enters the correct relay activation command, the called party will hear 2 short confirmation beeps and the entry phone will output a buzzing sound (relay activation tone) while the door strike relay is activated. The tone (buzz) length will match the relay activation time up to a maximum of 5 seconds. The tone (buzz) can be programmed to three different volume settings 1 = Low, 2 = Medium, 3 = High or it can be disabled. **Factory Setting:** 2

23. Relay Latch Commands

When set to "Enabled" the Remote Access Operation Commands (*0 to *1) to Un-Latch or Latch the relay are enabled. These can be entered on a Inbound call after the access code is dialed (if programmed).

When set to "Disabled" the Remote Access Operation Commands (*0 to *1) to Un-Latch or Latch the relay are disabled. Disabling the Latch commands can be useful in applications where you want to eliminate the possibility of inadvertently entering a latch command leaving a gate open/closed, etc. **Factory Setting:** Enabled

24. Alternating Switch Action (Panic Button Mode)

With Alternate Switch Action Enabled the CALL button alternately connects and disconnects calls. With Alternate Switch Action Disabled the CALL button connects calls only. Pressing the button again after the call has been initiated will not terminate the call. **Factory Setting:** Enabled

25. Speaker Mode

The Speaker Mode can be set to one of the following three modes. **Factory Setting:** ON

OFF/Silent Monitoring Mode: In the “OFF” mode the speaker is disabled at all times. However, the speaker can be enabled after communication has been established by entering touch tone command “9#”. The speaker will remain on for the duration of the call.

ON (factory setting): In the “ON” mode the speaker is enabled during In-bound and Out-bound calls.

OFF Until Answered: In the “OFF Until Answered” mode the speaker will remain silent during dialing and will not turn on until the called party has answered.

26. Speaker Volume

The Speaker volume can be set from 0 - 19. 0 = lowest volume setting, 19= highest volume setting. Adjusting this will set the Speaker volume level for SIP Paging and incoming/outgoing phone calls. This setting does not affect multicast volume settings. **Factory Setting:** 7

The Speaker volume can alternatively be set to Auto 0 – 10, 15 or 19. 0 = the lowest automatic volume setting and 10, 15 or 19 = the highest automatic volume setting. With volume set to auto the **VoIP Tower Phone** will use its built-in microphone to monitor background room noise and automatically increase or decrease the Speaker and Line Out volume for SIP paging, incoming/outgoing phone calls and Multicast paging. **Note:** *The minimum volume setting for multicast paging is set separately per multicast IP address and port. See “Paging” menu in Viking IP Programming.*

27. Ring Volume

When set to Ring or Ring with AGC, The **VoIP Tower Phone** will output a loud ring when it is called. The level can be adjusted from 0 - 19. **Factory Setting:** 5

28. Microphone Volume / Automatic Noise Cancelling Mode

The microphone volume can be set from 1 to 9, 1 = lowest volume setting, 9 = highest volume setting. Alternatively the microphone can be placed in the “Auto” Automatic Noise Cancelling mode. With the mic in the Auto mode, when background noise increases, the mic gain will automatically decrease. When background noise decreases the mic gain will automatically increase. The Auto mode is useful in applications where the background noise level can change drastically. **Factory Setting:** 5

29. Talk / Listen Delay (VOX)

This feature selects switching time between talk and listen modes (VOX switching time). The Talk/Listen Delay can be programmed from 0.1 - 0.9 seconds. **Factory Setting:** 0.5 seconds

30. In-Band Audio Call Progress

The In-Band Audio Call Progress Detection can be set to enabled or disabled. In-Band Audio Call Progress detection should be enabled in applications where you are making an outbound call through your VoIP phone system and are relying on In-Band analog audio for ringback or busy detection. **Factory Setting:** Enabled.

31. In-Band Audio Detect Sensitivity

The In-Band Audio Detection level (Sensitivity) can be set from 1 - 9, 1 = minimum setting, 9 = highest setting. Increasing or decreasing the sensitivity may be required in applications where you are making an outbound call through your VoIP phone system and are relying on In-Band analog audio detection. **Factory Setting:** 5

32. Repeat Announcement Option

The **VoIP Tower Phone** can be programmed to play the announcement from 1 - 9 times, or to continuously repeat the announcement every 6 seconds until a Touch Tone * is detected from the distant party. The call connected LED will turn on automatically after the announcement has stopped repeating. **Factory Setting:** 1 (play the announcement 1 time)

33. Lap Counter

With the lap counter disabled, if the **VoIP Tower Phone** is programmed to dial the next number on ring-no-answer and/or busy signal, the **VoIP Tower Phone** will continuously call its programmed phone numbers forever until the call is answered.

The lap counter is a programmable counter that determines how many times the **VoIP Tower Phone** will cycle through its list of up to 5 Speed Dial phone numbers, before it stops the dialing process and hangs up. When all of the programmed phone numbers have been dialed, the lap counter is incremented and the dialing process repeats. When the lap counter has been met, the dialing process stops and the **VoIP Tower Phone** hangs up.

Factory Setting: Disabled

34. SIP Call Length Timeout

This feature selects the maximum length of time that calls can be connected. Programmable in increments of 1 minute up to a maximum of 9 minutes or disabled. With the call length disabled, the **VoIP Tower Phone** must rely on a call ended signal, busy signal, Ring No Answer limit, or touch tone # to hang-up. **Factory Setting:** 3 minutes

35. Multicast Page Length Time Out

The Multicast Page Length Timeout can be programmed from 1 - 255 seconds in one second increments. The Time Out can be disabled, allowing any length of page or continuous background music. Only one timeout time can be programmed for all multicast paging groups. When using standard Multicast Paging, a timeout checkbox can be selected for each paging group. When using Polycom Multicast Paging, the timeout checkbox enables the timeout globally on groups 1 through 24. There is no timeout for a page to the emergency group 25.

Factory Setting: 180 seconds

36. Inbound Call Mode

The Inbound Call Mode determines how the **VoIP Tower Phone** handles incoming SIP calls. One option is to generate a loud ring sound through the speaker. The **VoIP Tower Phone** can also auto answer a SIP call to transmit a page, control the relay or listen to transmit audio from the microphone. The last option is the silent monitor mode, which allows callers to listen to the transmit audio from the microphone. The “secure” options for auto answer require the callers to dial the access code in order to transmit a page, activate the relay or activate the optional **RC-4A** relays.

Factory Setting: Auto Answer

Disabled – Inbound SIP calls are not allowed.

Auto Answer – Inbound SIP calls are auto answered on the first ring. This can also be used for Silent Monitoring by changing the Speaker Mode to ‘OFF/Silent Monitor’, See page 20. For more security use the Auto Answer Secure Mode.

Auto Answer Secure – In the “Auto Answer - Secure” mode, the phone will automatically answer an incoming call on the first ring. The unit will then output 2 beeps but will not go into two-way voice mode. After the 2 beeps are heard, you will have 10 seconds to enter the pre-programmed 1 - 6 digit Access Code (see section 15 on page 17). When the correct code is entered, 2 beeps will be heard and the unit will enable the Two-Way Mode (mic and speaker audio). After the correct Access Code is entered, any of the “Remote Access Operation Commands” can be used (see Operation, section D on page 25). **Note:** *If the wrong Access Code is entered or more than 10 seconds have elapsed, the phone will output 3 beeps and disconnect.*

Ring: In the “Ring” mode the speaker phone will not automatically answer an incoming call but will output a loud ring signal out of the speaker in a 2 seconds on, 4 seconds off ring pattern. There are four available ring cadences.

Ring with AGC: In the “Ring with AGC” mode the speaker phone will not automatically answer an incoming call but will output a loud ring signal out of the speaker in a 2 seconds on, 4 seconds off ring pattern. The phone will automatically increase or decrease the ring volume based on background ambient noise. The call can then be answered by momentarily pressing the call button.

37. Ring Cadence

The Ring cadence can be programmed to one of 4 different cadences. **Factory Setting:** Normal Ring

Normal Ring (single ring, 2 seconds on 4 seconds off) factory setting

Double Ring (double ring, 1 second on 0.5 seconds off 1 second on 3.5 seconds off)

Short-Short-Long (triple ring, 0.5 seconds on 0.5 seconds off 0.5 seconds on 0.5 seconds off 1 second on 3 seconds off)

Short-Long-Short (triple ring, 0.5 seconds on 0.5 seconds off 1 second on 0.5 seconds off 0.5 seconds on 3 seconds off)

38. Dial Next Number on Ring No Answer

If enabled and a ring-no-answer is detected, the **VoIP Tower Phone** will dial the next programmed Speed Dial number after the programmed amount of rings. A momentary press of the call button will dial the first programmed Speed Dial number. **Factory Setting:** 7 (will redial after 7 rings)

39. Dial Next Number on Busy

If enabled and a busy is detected, the speaker phone will dial the next programmed Speed Dial number. A momentary press of the call button will dial the first programmed phone number. **Factory Setting:** Enabled

40. Send ID Number as

The I.D. number can be transmitted as RFC 2833 or as In-Band DTMF. **Factory Setting:** RFC 2833

41. Call LED Mode

The “Call” LED on the **VoIP Tower Phone** can be programmed to one of four different modes. **Factory Setting:** Emergency Phone

OFF Mode: Useful for silent monitoring applications. In this mode the LED will not light during normal operation. It will only light (blink) if it cannot register with the programmed SIP server or while manually resetting all network parameters to factory default.

Entry Phone Mode: The LED will remain ON in the idle state, turn off while button is pressed, blink during dialing, light steady when the call is answered, then turn OFF momentarily when the call is completed.

Emergency Phone Mode: The LED will remain OFF in the idle state, blink during dialing, light steady when the call is connected, then turn OFF when the call is completed.

Emergency Phone Outbound Only: On outbound calls, the LED will remain OFF in the idle state, blink during dialing, light steady when the call is connected, then turn OFF when the call is completed. On in-bound calls, the LED will remain off. This is useful for silent monitoring on inbound calls.

42. “Call” LED Control

During outbound SIP calls the “CALL” LED can be programmed to light automatically when the called party has answered or only light after the called party has entered a touch tone “*”. **Factory Setting:** Automatic

Viking IP Programming - V1.2.0 - E-1600-IP Series Mode

Name: **VIKING_MK64_Vik02** MAC: **18:E8:0F:50:3A:C4** IP: **192.168.210.70**

Phone Numbers / Codes / Audio | Phone Settings | IP Settings | VLAN | Alarm / Firmware / Default / Tools

Dynamic IP Settings

Unit IP Address: 192.168.210.70
Subnet Mask: 255.255.255.0
Gateway IP Address: 192.168.210.18
These are the current / last known good IP addresses

Firmware Versions

IP: R6.33.1826 [Update IP]
Unit: V3.8 [Update Unit]

Import / Export

[Import Data] [Export Data]

SIP / Network Failure Alarm

SIP Registration Status: Not Registered
Current Alarm Status: **Activated - SIP**
☐ Alarm Muted [Mute Current / Next Alarm]
Permanent Alarm Mute: Alarm Tones Enabled
Alarm can be triggered by unit failing to register with SIP server, Gateway ping failure, or IP address conflict (if unit forced to use static settings).

Default Unit Settings

[Clear Phone Settings] [Clear IP Settings]

Diagnostics

[Test Relay Activation] [Read Relay Status]
Last Read Relay Status: [Text Box]
[Test Mic / Speaker] Result: [Text Box]

[Disconnect]

43. Mute Current / Next Alarm

A network failure alarm will be indicated by providing 3 beeps every 30 seconds. A network failure indicates the unit is not registered to the SIP server or there is a communication failure with the gateway. The three beeps can be muted by clicking on “Mute Current / Next Alarm”. The Status LED will continue to flash to assist troubleshooting. The alarm beeps can also be permanently disabled. See Permanent Alarm Mute.

44. Permanent Alarm Mute

Selecting “Alarm Tones Disabled” will mute all alarm tones indefinitely. To re-enable alarm tones select “Alarm Tones Enabled”. **Factory Setting:** Alarm Tones Enabled

45. Programming a Username and Password to Restrict Access to Programming

To increase security, a username and password can be programmed to limit access to the **VoIP Tower Phone** using Viking IP Programming software. When no username and password are programmed and the security code is still set to default (845464), the PC software will not require a username, password or security code when connecting to the **VoIP Tower Phone**. If the **VoIP Tower Phone** has been programmed with a username and password, a pop up window will ask for the username followed by the password. If the unit's security code has been changed from default (845464), it will then prompt for the correct security code.

If the username and password are unknown, they can only be erased by resetting all network parameters to default with the HELP button (see section F on page 11). If the username and password are known but you wish to erase them, that can be accomplished by exporting the data from the **VoIP Tower Phone**, resetting all network parameters to default with the HELP button (see section F on page 11) or clicking on “Clear IP Settings” and then importing the data back into the **VoIP Tower Phone**.

46. IP Firmware

If new **VoIP Tower Phone** firmware is available, after opening the programming software a pop window will come up asking you if you would like to update firmware. An alternative method of updating can be done by clicking the IP firmware “Update IP” button. You can then browse to the folder that contains the PIP file for updating the unit's IP firmware. This method is typically only used when Viking Technical Support has sent you updated IP firmware.

47. Unit Firmware

If new **VoIP Tower Phone** firmware is available, after opening the programming software a pop up window will ask if you would like to update firmware. Another way to update is accomplished by clicking the Unit firmware “Update Unit” button. You can then browse to the folder that contains the HEX file for updating the unit's firmware. This method is typically only used when Viking Technical Support has sent you updated firmware.

48. Import / Export

The Import / Export feature is useful for backing up all the **VoIP Tower Phone**'s programming or for importing programming when installing multiple units with a majority of the same programming.

49. Clear Unit Settings

Clicking on the “Clear Unit Settings” button in programming will reset all of the Programming Features back to their factory default settings. **Note:** *This command will not change or reset your IP settings.*

50. Clear IP Settings

Clicking on the “Clear IP Settings” will reset all of the IP settings back to their factory default settings. This also clears paging Group settings and Addresses. **Note:** *This will not effect any speaker or paging settings.*

51. Diagnostics

The Diagnostics section in the Viking IP Programming can be used to test the functionality of the mic, speaker and the relay. **Note:** *Using the “Test Relay Activation” button is not recommended when Relay Mode is set to LV-1K Control.*

52. Resetting Username and Password

Export your settings with the Export Data button. Click on Clear IP Settings. The unit will reset and reboot (IP address may change). Connect to the unit and import the Settings file with the Import Data Button.

A. “HELP” Button

When the “**HELP**” button is pressed, the **VoIP Tower Phone** dials a pre-programmed telephone number. The Call Connected LED momentarily flashes during dialing. In the event the line is busy or there is a ring-no-answer, the unit can be programmed to call additional phone numbers.

The phone then cycles through up to 5 pre-programmed emergency numbers until the call is answered. When the call is answered, the digital voice announcer will automatically play to identify the location of the emergency call. The phones are factory programmed to play the announcement once, and then automatically light the “Call Connected” LED to show that handsfree communication to emergency personnel is established. The * key will send the I.D. number (if programmed), and play the announcement again. The distant party will know the location of the emergency call by either the voice announcement or by decoding the touch tone I.D. number. Once the “Call Connected” LED is on, the # key can be used to force the phone to hang-up.

B. “INFO” Button (E-1600BLT2IPEWP and E-1600AST2IPEWP)

When the “**INFO**” button is pressed (**E-1600BLT2IPEWP** and **E-1600AST2IPEWP** only), the phone goes off-hook and dials the first “**INFO**” phone number programmed. If a busy signal is detected or the call goes unanswered, the phone will cycle through all five “**INFO**” phone numbers until the call is answered. When answered, handsfree communication is established. **Note:** *The voice announcement is for **Emergency/Help** calls only and will not play on a call initiated from the “**INFO**” button.*

C. Multicast

The **VoIP Tower Phone** can be set to one of two modes for multicasting, Standard or Polycom. Polycom is only used when Polycom IP phones are used. To change to Polycom multicast, go to the “Paging” tab, “More Speaker Settings” and change the “Multicast Page Type” to Polycom.

Standard Multicast Paging (Factory Setting)

Up to 10 multicast paging groups can be programmed into each unit and each group is assigned a priority level allowing simultaneous arriving multicast pages to be handled based on importance. Low priority groups are often used for low priority streams like background music and high priority groups are used for high priority streams such as emergency messages. Group 0 is the highest priority, and Group 9 is the lowest. A timeout limit can be set and either enabled or disabled for each multicast group. The volume of each group can be programmed and non-emergency SIP calls can be assigned either higher or lower priority than all or some of the multicast groups (see Programming section 11).

Polycom Multicast Paging

Polycom phones send multicast audio to a specific IP address and port. A total of 25 groups can be used and groups 1 through 24 can individually be enabled/disabled in each **VoIP Tower Phone**. Group 25 is always enabled, as group 25 is generally used for emergency broadcasts. Any multicast page made from a Polycom phone (on the same LAN) to the programmed IP address, port and to an enabled group will play through the **VoIP Tower Phone** speaker. The priority of Polycom sources ranges from group 1 (lowest) to group 25 (highest). Non-emergency SIP calls can be assigned either higher or lower priority than Polycom group 1 through 24 multicast paging (see the table in programming section 11 for Polycom paging priorities). Multicast page timeout can be enabled/disabled globally for groups 1 through 24 (there is no timeout for group 25) and the length of the Multicast page timeout is adjustable. The volume for Polycom multicast is set globally for all groups.

D. Remote Access Operation Commands

The following commands can be entered after answering an inbound call from the emergency phone. The commands can also be entered on an outbound call to the emergency phone. After the emergency phone auto answers the call, two beeps will be heard. If the access code has been disabled, you can now enter the Remote Access Operation Commands below. If an Access code has been programmed, enter the Access code digits. With the correct code entered, two beeps will be heard and you can now enter the Remote Access Operation Commands below.

Feature	Tone Tone Command	Description
Activate Relay	** or — —	Momentarily activate relay (1 or 2 digits, factory set to **).
Un-Latch Relay	*0	Un-latch* (deactivate) the relay.
Latch Relay	*1	Latch* (continuously activate) the relay.
Disconnect	#	Disconnects or forces the emergency phone to hang up.
Send ID and Play Message	*	Send I.D. number (if programmed) and plays the announcement.

* **Note:** Latching commands must be enabled (*6) in programming.

Troubleshooting

If the unit cannot register with the programmed SIP server, the LED will blink on and off every two seconds, and three error beeps will be heard every 30 seconds until communication is restored. This alerts a potential user of a problem with the device that will prevent an emergency phone call from being made.

You may silence the error beeps, per instance, by pressing and holding the HELP button for 5 seconds or by clicking the “Mute Alarm Until Next Failure” button in the Viking VoIP Programming Software (see section **B** on page 9). The error beeps automatically re-enable once the unit is registered, to alert of any new problems that arise.

Related Products

ADA Compliant VoIP Emergency Phones with Built-In Dialer and Digital Voice Announcer

The **1600-IP Series** of ADA Compliant VoIP Emergency Phones are designed to provide quick and reliable handsfree communication for SIP VoIP phone systems with PoE and also allows for standard paging via multicast. All **1600-IP Series** phones meet ADA requirements for elevator or emergency telephones, and can be programmed from any touch tone phone or PC on the same LAN or remotely using a static IP address. The phones can dial up to 5 programmable emergency numbers. In addition, the **E-1600-20-IP**, **E-1600-22-IP**, **E-1600-52-IP** and **E-1600-TP2IPEWP** feature a second “INFO” button that will dial up to 5 non-emergency numbers. Two amp relay contacts are provided for strobe light, camera, door/gate control, etc.

The **1600-IP Series** phones can be programmed to automatically deliver a digital announcement to identify the location of the emergency call. Alternatively, a DTMF touch tone code may also be delivered. A “Call Connected” LED can be initiated manually or automatically. All programming parameters, including phone numbers and location numbers, are stored in non-volatile memory. All units are PoE powered.

For outdoor installations where the unit is exposed to precipitation or condensation, select **1600-IP Series** phones are available with Enhanced Weather Protection (**EWP**). EWP products are designed to meet IP66 standards and may feature foam rubber gaskets, sealed connections, gel-filled butt connectors, as well as potted circuit boards with internally sealed, field-adjustable trim pots and DIP switches for easy onsite programming. For more information go to www.vikingelectronics.com/ewp



Warranty

IF YOU HAVE A PROBLEM WITH A VIKING PRODUCT, CONTACT VIKING TECHNICAL SUPPORT: 715-386-8666

Our Technical Support Department is available for assistance Monday through Friday 8:00am to 5:00pm central time. So that we can give you better service, before you call please:

1. Know the model number, the serial number and what software version you have (see serial label).
2. Have the Product Manual in front of you.
3. It is best if you are on site.

RETURNING PRODUCT FOR REPAIR

The following procedure is for equipment that needs repair:

1. Customer must contact Viking's Technical Support Department at 715-386-8666 to obtain a Return Authorization (RA) number. The customer MUST have a complete description of the problem, with all pertinent information regarding the defect, such as options set, conditions, symptoms, methods to duplicate problem, frequency of failure, etc.
2. Packing: Return equipment in original box or in proper packing so that damage will not occur while in transit. The original product boxes are not designed for shipping - an overpack box is required to prevent damage in transit. Static sensitive equipment such as a circuit board should be in an anti-static bag, sandwiched between foam and individually boxed. All equipment should be wrapped to avoid packing material lodging in or sticking to the equipment. Include ALL parts of the equipment. C.O.D. or freight collect shipments cannot be accepted. Ship cartons prepaid to:

**VIKING ELECTRONICS
1531 INDUSTRIAL STREET
HUDSON, WI 54016**

3. Return shipping address: Be sure to include your return shipping address inside the box. We cannot ship to a PO Box.
4. RA number on carton: In large printing, write the RA number on the outside of each carton being returned.

RETURNING PRODUCT FOR EXCHANGE

The following procedure is for equipment that has failed out-of-box (within 10 days of purchase):

1. Customer must contact Viking's Technical Support at 715-386-8666 to determine possible causes for the problem. The customer MUST be able to step through recommended tests for diagnosis.
2. If the Technical Support Product Specialist determines that the equipment is defective based on the customer's input and troubleshooting, a Return Authorization (RA) number will be issued. This number is valid for fourteen (14) calendar days from the date of issue.
3. After obtaining the RA number, return the approved equipment to your distributor. Please reference the RA number on the paperwork being shipped back with the unit(s), and also the outside of the shipping box. The original product boxes are not designed for shipping - an overpack box is required to prevent damage in transit. Once your distributor receives the package, they will replace the product over the counter at no charge. The distributor will then return the product to Viking using the same RA number.
4. **The distributor will NOT exchange this product without first obtaining the RA number from you. If you haven't followed the steps listed in 1, 2 and 3, be aware that you will have to pay a restocking charge.**

TWO YEAR LIMITED WARRANTY

Viking warrants its products to be free from defects in the workmanship or materials, under normal use and service, for a period of two years from the date of purchase from any authorized Viking distributor. If at any time during the warranty period, the product is deemed defective or malfunctions, return the product to Viking Electronics, Inc., 1531 Industrial Street, Hudson, WI., 54016. Customer must contact Viking's Technical Support Department at 715-386-8666 to obtain a Return Authorization (R.A.) number.

This warranty does not cover any damage to the product due to lightning, over voltage, under voltage, accident, misuse, abuse, negligence or any damage caused by use of the product by the purchaser or others. This warranty does not cover non-EWP products that have been exposed to wet or corrosive environments. This warranty does not cover stainless steel surfaces that have not been properly maintained.

NO OTHER WARRANTIES. VIKING MAKES NO WARRANTIES RELATING TO ITS PRODUCTS OTHER THAN AS DESCRIBED ABOVE AND DISCLAIMS ANY EXPRESS OR IMPLIED WARRANTIES OR MERCHANTABILITY OR FITNESS FOR ANY PARTICULAR PURPOSE.

EXCLUSION OF CONSEQUENTIAL DAMAGES. VIKING SHALL NOT, UNDER ANY CIRCUMSTANCES, BE LIABLE TO PURCHASER, OR ANY OTHER PARTY, FOR CONSEQUENTIAL, INCIDENTAL, SPECIAL OR EXEMPLARY DAMAGES ARISING OUT OF OR RELATED TO THE SALE OR USE OF THE PRODUCT SOLD HEREUNDER.

EXCLUSIVE REMEDY AND LIMITATION OF LIABILITY. WHETHER IN AN ACTION BASED ON CONTRACT, TORT (INCLUDING NEGLIGENCE OR STRICT LIABILITY) OR ANY OTHER LEGAL THEORY, ANY LIABILITY OF VIKING SHALL BE LIMITED TO REPAIR OR REPLACEMENT OF THE PRODUCT, OR AT VIKING'S OPTION, REFUND OF THE PURCHASE PRICE AS THE EXCLUSIVE REMEDY AND ANY LIABILITY OF VIKING SHALL BE SO LIMITED.

IT IS EXPRESSLY UNDERSTOOD AND AGREED THAT EACH AND EVERY PROVISION OF THIS AGREEMENT WHICH PROVIDES FOR DISCLAIMER OF WARRANTIES, EXCLUSION OF CONSEQUENTIAL DAMAGES, AND EXCLUSIVE REMEDY AND LIMITATION OF LIABILITY, ARE SEVERABLE FROM ANY OTHER PROVISION AND EACH PROVISION IS A SEPARABLE AND INDEPENDENT ELEMENT OF RISK ALLOCATION AND IS INTENDED TO BE ENFORCED AS SUCH.

If trouble is experienced with the **E-1600-BLTIPEWP / E-1600BLT2IPEWP / E-1600-ASTIPEWP / E-1600AST2IPEWP** phone, for repair or warranty information, please contact:
Viking Electronics, Inc., 1531 Industrial Street, Hudson, WI 54016 715-386-8666

WHEN PROGRAMMING EMERGENCY NUMBERS AND (OR) MAKING TEST CALLS TO EMERGENCY NUMBERS:

Remain on the line and briefly explain to the dispatcher the reason for the call. Perform such tests in off-peak hours, such as early morning or late evenings.

PART 15 LIMITATIONS

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

CANADA

This class A digital apparatus complies with Canadian ICES-003. Cet appareil numérique de la classe A est conforme à la norme NMB-003 du Canada.

Product Support: 715-386-8666

Due to the dynamic nature of the product design, the information contained in this document is subject to change without notice. Viking Electronics, and its affiliates and/or subsidiaries assume no responsibility for errors and omissions contained in this information. Revisions of this document or new editions of it may be issued to incorporate such changes.