

Viking VoIP SIP System Compatibility List

IMPORTANT: Exclusion from this list means only that compatibility has not been verified, **it does not mean incompatibility.**

For detailed configuration instructions, contact Viking at: 715-386-8861

Vendor	Infrastructure Class				
	Softswitch	PBX	Proxy	SBC (session border controller)	Service Provider
3COM VCX		X			
3CX		X			
Aastra		X			
Asterisk		X			
Atcom		X			
Avaya Aura Communication Manager	X				
Avaya IP Office		X			
BlueBox		X			
Brekeke		X			
Callcentric					X
Cisco Unified Communications Manager (CUCM)	X	X			
Cisco Unified Communications Manager Express (CUCME)	X	X			
Elastix		X			
Epygi QX200		X			
Freeswitch		X			
Grandstream		X			
Interactive Intelligence	X	X			
iPECS (Ericsson-LG)		X			
iptel.org					X
Kamailio			X	X	
MetaSwitch				X	X
NEC		X			
OfficeSIP		X			
OpenSIPS		X			
Panasonic** (with SIP Extension Card)		X			
Samsung Communications Manager (SCM)	X	X			
ShoreTel*		X			
Siemens Communications Server (SCS)		X			
SIP Express Router (SER)			X	X	
sip.antisip.com					X
Snom PBX		X			
Sonus				X	
Switchvox		X			X
Teksip			X		
Toshiba		X			
VoIP.ms					X

* **Note:** Not compatible with ShoreTel Ring Group/Hunt Group (unit can be programmed to ring an extension 2 or 3 times then roll to the next number, for a total of 5 numbers).

** **Note:** Relay operation commands are Not compatible with Panasonic Phone Systems (Panasonic does not transmit DTMF between station ports).

Known Incompatible System or Service Provider: Ring Central (Requires Authorization ID and Proxy address).

Callcentric as the SIP provider for Viking IP Products:

An account must be set up with Callcentric. Go to www.callcentric.com to set up an account and upon completion, a SIP username and password will be assigned. With Callcentric service, the SIP server will always be "callcentric.com".

The "In-band audio call progress" feature under phone settings must be enabled (if not already).

When the Viking IP has placed a call and the remote party hangs up, it takes approximately 22 seconds for the SIP server to pass the disconnect signal to the Viking IP phone.

The screenshot shows the 'Viking IP Programming - V1.1.4' window. At the top, it displays 'Name: Entry Phone', 'MAC: 18:E8:0F:50:06:8B', and 'IP: 192.168.210.108'. Below this is a tabbed interface with 'Phone Numbers / Codes / Audio' selected. The 'Emergency Phone Numbers' section contains five input fields; the 'First' field is highlighted in yellow and contains '17153864355'. The 'Information Phone Numbers' section also has five empty input fields. The 'Phone Codes' section includes 'Security Code (6 digits)' with '845464', 'ID Number (0-6 digits)' with '0', and an empty 'Access Code (0-6 digits)' field. The 'Audio File' section shows 'Loaded Audio File Name: Manual Recording' and two buttons: 'Upload Wav File (8KHz, Mono, 8 or 16-bit PCM)' and 'Erase Existing Audio File'. At the bottom, a status bar says 'Checking Alarm Status - Completed' and a 'Disconnect' button is on the right.

In our case, it was programmed to dial 715-386-4355, which is a test phone number here. Note that Callcentric requires a "1" before the telephone number.

Viking IP Programming - V1.1.4

Name: Entry Phone MAC: 18:E8:0F:50:06:8B IP: 192.168.210.108

Phone Numbers / Codes / Audio Phone Settings IP Settings Alarm/ Firmware/ Import/ Export/ Default/ Diagnostics

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

Relay Mode Door Strike In-Band Audio Detect Sensitivity 5

Relay Activation Command 5 Repeat Announcement Option 1

Relay Activation Time 15 sec Lap Counter 1

Relay Buzz Volume 2 Call Length Time Out 1 min

Relay Latch Commands Enabled Inbound Call Mode Auto Answer

Alternating Switch Action Enabled Ring Cadence Normal Ring

Speaker Mode On Dial Next No. on Ring No Answer 7

Speaker Volume 1 Dial Next Number on Busy Enabled

Ring Volume 1 Send ID Number as RFC 2833

Microphone Volume 4 "Call" LED Mode Emergency Phone

Talk / Listen Delay (VOX) .5 sec "Call" LED Control Automatic

Disconnect

Note that "In-Band Audio Call Progress" is enabled.

Viking IP Programming - V1.1.4

Name: Entry Phone MAC: 18:E8:0F:50:06:8B IP: 192.168.210.108

Phone Numbers / Codes / Audio Phone Settings IP Settings Alarm/ Firmware/ Import/ Export/ Default/ Diagnostics

Set Unit IP Address via DHCP Unit Name Entry Phone

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

Gateway Ping / Registration Time

Ping Timer (Sec) 10

Registration Timer (Min) 30

Logging / Time Server Settings

Syslog Server 192.168.154.100

NTP Server 2.viking.pool.ntp.org

SIP Server Settings - Not Registered

SIP Server callcentric.com

Username 17772685129

Password bear845464

Caller ID

Codec Selection and Negotiation Order

Codecs G711a, G711u, G722

Disconnect

Note the SIP server settings. When we set up our account with Callcentric, our SIP username was "17772685129" and our SIP password was "bear845464".

Cisco Unified Communication Manager:

To connect a Viking VoIP Phone to Cisco's Unified Communication Manager, it is important to know that the phone is set up as a third party SIP device without Authentication ID. Cisco has created a write up for connecting most third party phones, which we have reformatted here specifically for Viking equipment.

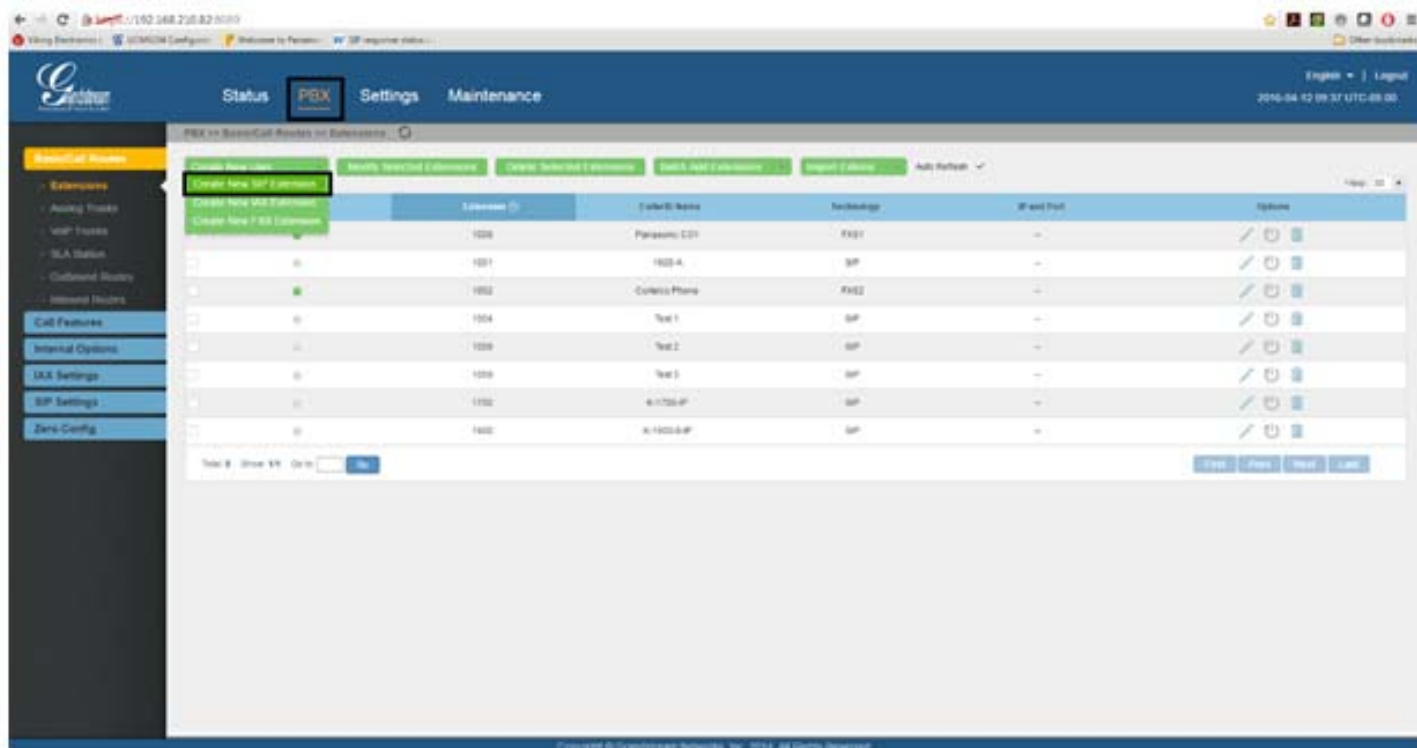
Step 1.	Gather the following information about the phone: • MAC address • Physical location of the phone • Cisco Unified Communications Manager user to associate with the phone • Partition, calling search space, and location information, if used • Line number (DN) to assign to the phone
Step 2.	Determine whether sufficient Device License Units are available. If not, purchase and install additional Device License Units. Third-Party SIP Devices (Basic) consume three Device License Units each.
Step 3.	Configure the end user. Viking VoIP Phones do not support an authorization ID (digest user), so create a user with a user ID that matches the DN of the phone. For example, create an end user named 1000 and create a DN of 1000 for the phone. Assign this user to the phone (see step 9).
Step 4.	Configure the SIP Profile or use the default profile. The SIP Profile gets added to the phone that is running SIP by using the Phone Configuration window.
Step 5.	Configure the Phone Security Profile. To use digest authentication, you must configure a new phone security profile. If you use one of the standard (non-secure) SIP profiles that are provided for auto-registration, you cannot enable digest authentication.
Step 6.	Add and configure the Viking VoIP Phone by choosing Third-party SIP Device (Basic) from the Add a New Phone Configuration window.
Step 7.	Add and configure lines (DNs) on the phone.
Step 8.	In the End User Configuration window, associate the Viking VoIP Phone with the user by using Device Association and choosing the Viking VoIP Phone.
Step 9.	In the Digest User field of the Phone Configuration window, choose the end user that you created in step 3.
Step 10.	Provide power, install, verify network connectivity, and configure network settings for the Viking VoIP Phone. Username should match the user that was created in step 3. Password should match the password created for the digest user.
Step 11.	Make calls with the Viking VoIP Phone.

Configuring Viking IP phones with Grandstream PBX devices:

1. Log into the Grandstream configuration tool in your browser:



2. Click on “PBX” at the top, and then “Create New User” and “New SIP Extension”



3. Put in your preference for “Extension” (which will be your username), “SIP/IAX Password” and “Voicemail Password”, along with “First Name” and “Last Name”. Click on “Save”.

Create New SIP Extension

General

Extension1500

PermissionInternal

Enable Voicemail☒

Call Forward Unconditional

Call Forward No Answer

Call Forward Busy

Ring Timeout

Skip Voicemail Password Verification

Disable This Extension

CallerID Number

SIP/IAX Password1500

Voicemail Password1500

CFU Time ConditionAll Time

CFN Time ConditionAll Time

CFB Time ConditionAll Time

Auto Record

Support Hot-Desking Mode

Music On Holddefault

User Settings

First NameJohn

Last NameSmith

Email Address

LanguageDefault

SIP Settings

NAT☒

DTMF ModeRFC2833

Enable Keep-alive

AuthID

Can ReinviteNo

InsecurePort

Keep-alive Frequency60

TEL URIDisabled

Other Settings

SRTP☐

Skip Trunk AuthNo

StrategyAllow All

Fax Detection☐

Dial Trunk Password

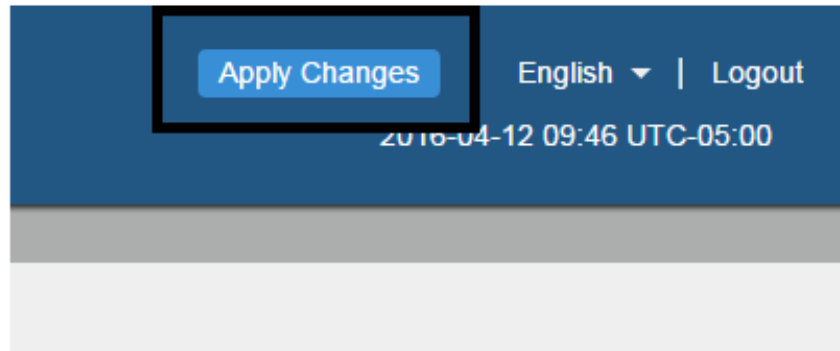
Cancel

Save

040317

DOD# 944

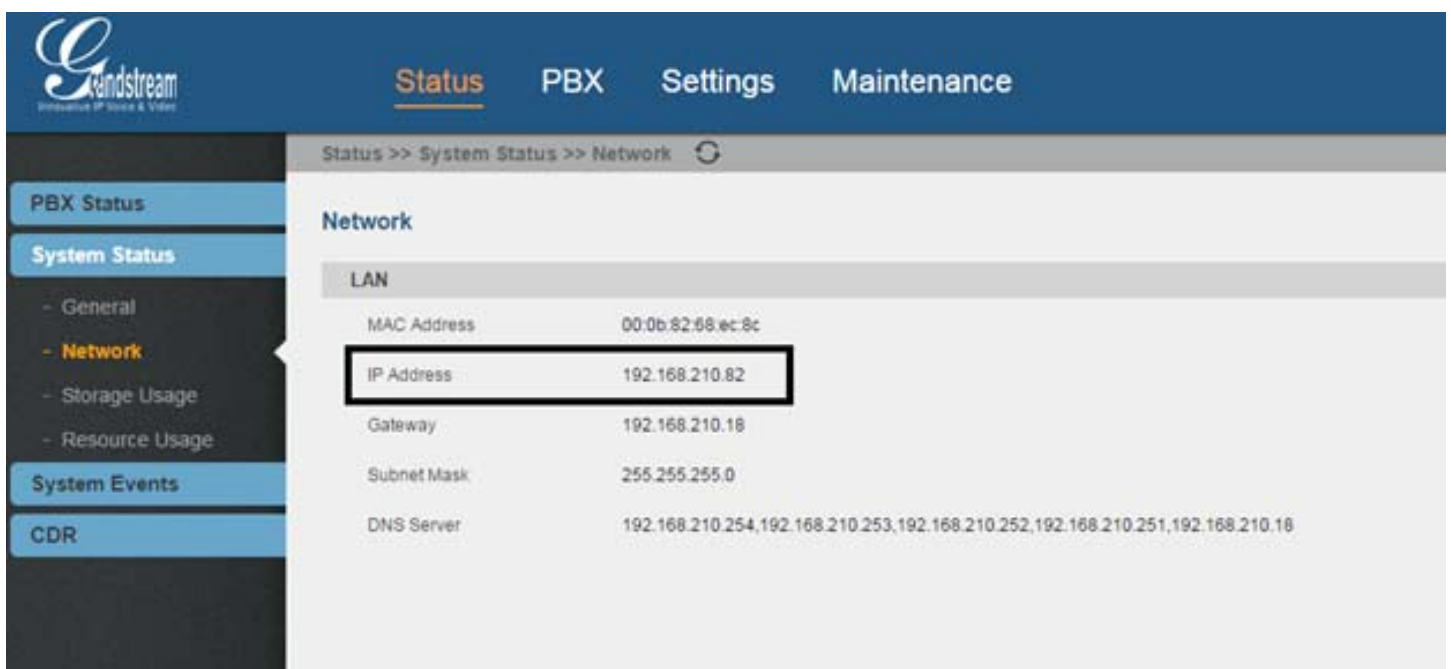
4. After saving, click on the “Apply Changes” button in the top right corner.



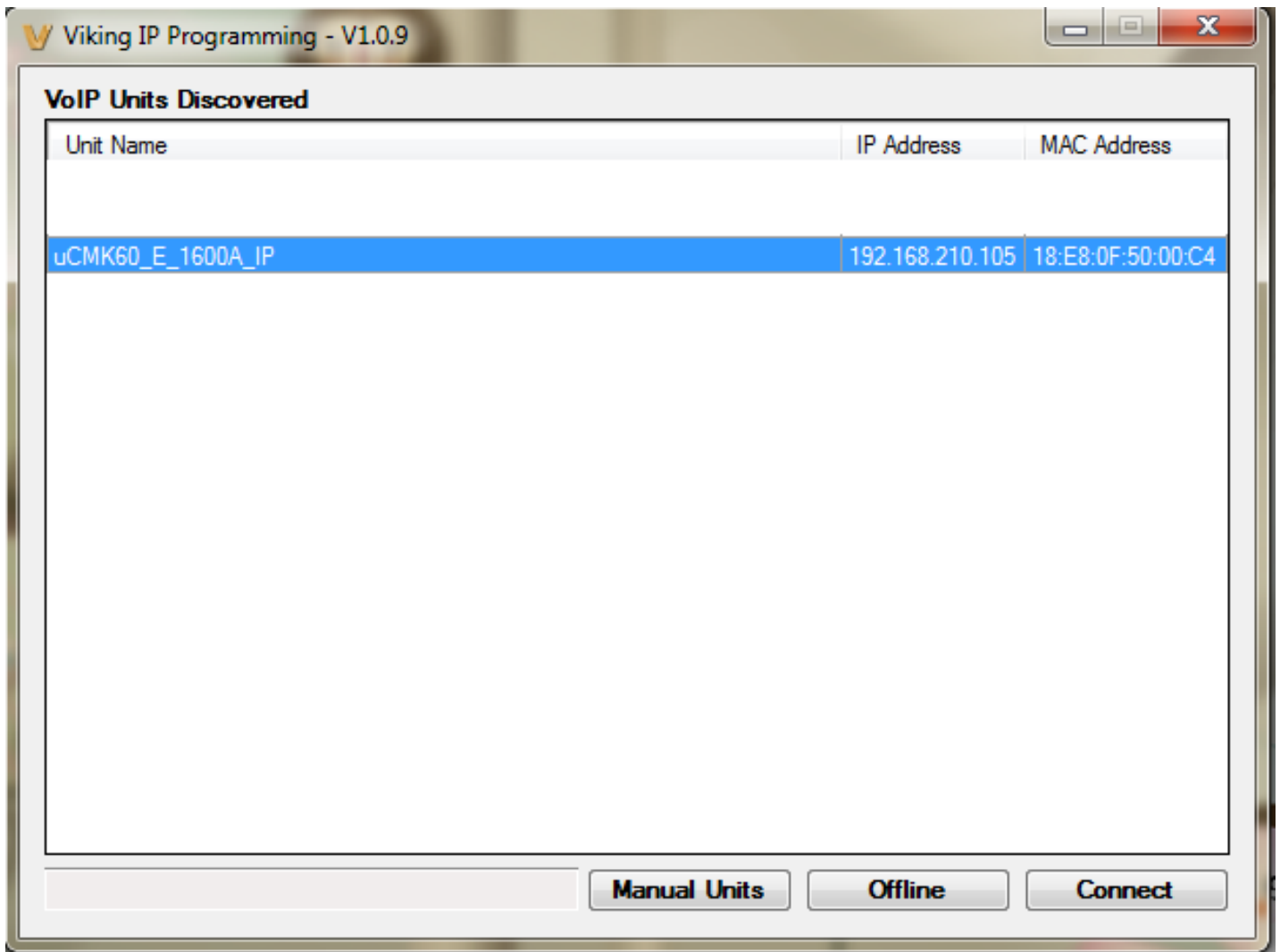
5. You will then see the extension you set up under the list of extensions.



6. Next, click on “Status” on the top, and on the left side, click on “System Status” and “Network”. Look for where it says “IP Address”. This is your SIP Server IP address that you will put in the Viking VoIP programming app.



7. Open the Viking programming app, and find the unit you want to work with, then click “Connect”.



Iwatsu ECS:

To connect a Viking VoIP Phone to the Iwatsu ECS you can register the phone as a SIP extension. As of version 2.6 of the Viking VoIP Phone Firmware inbound calls to the door box are not supported. To connect the phone, please follow the following steps:

Step 1.	Gather the following information about the phone: • MAC address • Physical location of the Viking Voip Phone • CCSU or LAN2 IP Address of the Iwatsu ECS System • Extension Number for the phone or hunt group that will be called when the button is pressed • Extension Number to assign to the Viking VoIP Phone
Step 2.	Determine if the system is provisioned for SIP extensions; licenses are required to support SIP Phones on the Iwatsu ECS and they must be installed prior to deployment.
Step 3.	Configure the SIP Extension in the Iwatsu ECS. The username will be the extension number and the password default is 1234.
Step 4.	Provide power, install, verify network connectivity, and configure network settings for the Viking VoIP Phone. Username should match the user that was created in step 3. Password should match the password created for the extension. A Screen Capture of a working configuration is shown in Figure 1.
Step 5.	Audio Call Progress must be set to 'Enabled.'
Step 6.	Make calls with the Viking VoIP Phone.

The screenshot displays the 'Viking VoIP Phone Programming - v1.1.1' window. It features several panels for configuring the phone:

- VoIP Phones Discovered:** A table showing discovered phones. One phone is listed: 'Call_Box' with IP Address '192.168.1.128' and MAC Address '18 E8 0F 50 00 AE'.
- Phone Settings:** Includes settings for Talk/Listen Delay (5 sec), Call Length Timeout (3 min), Mag. Play Count (1), Lap Counter (1), Dial Next # on RNA (Disabled), Dial Next # on Busy (Disabled), Microphone Volume (7), Speaker Volume (7), Relay Command (**), Relay Act. Time (5 sec), Relay Mode (Door Strike), Relay Buzz Volume (3), Alt. Switch Action (Disabled), Latch Commands (Enabled), Auto Answer/Ring (Auto Answer), Send ID Number as (RFC 2833), Audio Call Progress (Enabled), "Call" LED Control (Automatic), Audio Detect Level (6), Speaker Mode (On), and "Call" LED Mode (Emergency Phc).
- IP Settings:** Includes IP Address via (DHCP), Static IP Settings (IP Address: 10.2.2.202, Subnet Mask: 255.255.255.0, Gateway: 10.2.2.1, DNS Server: 10.2.2.10), and SIP Server Settings (IP Address: 10.2.2.20, Username: 150, Password: 1234, Caller ID: 150).
- Phone Numbers:** Includes Speed Dial Numbers (First: 368) and Information Phone Numbers (First through Fifth).
- Phone Codes:** Includes Security Code (845464), ID Number (987654), and Access Code (123456).
- Audio:** Includes Upload, Read, Erase, and Debug buttons.
- Phone Firmware:** Includes Update, Read, V2.6, and Verify buttons.
- IP Firmware:** Includes Update, R3.28.1448, and Port 0 buttons.
- Import / Export:** Includes Import Data and Export Data buttons.
- Diagnostics:** Includes Mic / Speaker and Activate Relay buttons.
- SIP / Network Failure Alarm:** Includes Alarm Status (Activated), Muted checkbox, and Enable Alarm Beeps button.

Figure 1 - Iwatsu ECS Sample Configuration

epygi:

The Viking VoIP Phone can be configured easily with Epygi QX IP PBXs (herein QX) like other IP phones, to make and receive calls and to support different application scenarios. This guide provides instructions how to configure Viking as an IP extension on QX. Based on this configuration simple emergency call scenario and interconnection with the door strikes are described. Features, settings, applications and connections specific to the operation of Viking are beyond the scope of this document.

The configuration described below is generic for all QX IP PBX models, such as the QX20, QX50, QX200, QX500, QX2000 and QXISDN4+.

Requirements:

- QX connected to the network with all network settings properly configured.
- QX is running firmware version 6.1.2 or higher. Always use the latest available firmware to achieve maximum compatibility with the QX's features and settings.
- At least five IP extensions (phones) connected to QX as destinations for emergency call.
- Viking running VoIP FW version: IP R3.45.1541, Phone V3.3, connected to the LAN interface of QX.
- PC with MS Window and Viking IP programming V.1.1.2 SW installed for Viking configuration, connected to the QX LAN interface.

Note: If Viking VoIP Phone is going to be connected to QX via WAN interface, ensure a filtering rule is enabled on the QX firewall for it (the unit's IP is added into Allowed IP List). Creating a rule is not required if the firewall on the QX is disabled or set to Low.

A. Configuring an IP Extension on QX for Viking VoIP Phone

The following main settings will be used in the example below for configuring Viking VoIP Phone as an IP extension on QX.

Username / User ID	Password	SIP Server, SIP Port	Attached IP Line, Extension	SIP Username
locext115	*****	172.30.0.1:5060	IP Line 15, Ext.115	20230@sip.epygi.com

To configure the QX, login into QX WEB GUI, select and configure an IP Line with extension attached, that will be used for Viking VoIP Phone:

Step 1.	Go to the Interfaces - IP Lines page.
Step 2.	Select a free (inactive) IP line (line # 13 in this example)and configure it as follows: • Enable the IP Phone option. • Select Other from the Phone Model drop down list. • Specify the Username and Password fields (Figure 1). Note: Make a note of the specified Username and Password as they will be needed when configuring the Viking. It is suggested to use a good strong password, or use the system generated one.
Step 3.	Go to the Extensions-Extensions page.
Step 4.	Click the Admin Settings icon for the Extension 115.
Step 5.	Go to the SIP Settings section (Figure 2) and register the extension on a SIP Server (sip.epygi.com in this example) to be able to make remote SIP calls to the unit (if needed).

QX200

Overview

IP Lines

FXS

FXO

E1/T1 Trunk

ISDN Trunk

PSTN Gateways

Dashboard

Setup

Extensions

Interfaces

Telephony

Firewall

Network

Status

Maintenance

IP Lines

IP Line Settings

IP Phone Templates

IP Phones Logo

FXS Gateways

IP Line Settings - IP Line 13

Go Back

IP Line 13

Inactive

IP Phone

Phone Model:

Other

MAC Address:

Line Appearance:

2

Username:

locext115

Password:

Generate Password

Transport:

UDP

Use Template:

<-- use default -->

Use Session Timer

Use Symmetric RTP

Save

Figure 1: IP Line Settings page

QX200

Dashboard

Setup

Extensions

Interfaces

Telephony

Firewall

Network

Status

Maintenance

Overview

Extensions

Dialing Directories

Conferences

Recordings

Receptionist

ACD

Authorized Phones

Add Extension

Add Multiple Extensions

Bulk Import

Extensions Management - Edit Entry

Go Back

General Settings

SIP Settings

SIP Advanced Settings

Remote Settings

Call Queue Settings

Voice Mailbox Settings

Class of Service Settings

SIP Registration Settings

115

Username / DID Number

20230

Password

Confirm Password

SIP Server

sip.epygi.com

SIP Port

5060

SIP Registration Transport

UDP

☒ Registration on SIP Server

Save

Go To User Settings

Go To Line Settings

Go To Codec Settings

Figure 2: SIP Settings section

B. Configuring the Viking VoIP Phone

Power on the unit and connect it to the LAN interface for QX. The settings of the unit will be configured through Viking IP programming SW application installed on PC with MS windows. The following configuration steps for Viking should be done:

Step 1.	Open the Viking VoIP Phone Programming software on the MS Windows PC that is connected to the same LAN as the Viking phone to be programmed.
Step 2.	The window in the upper left corner of the menu will show you the Viking phone(s) 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 Viking phone.
Step 3.	Click the Connect button. If a pop up window appears, enter the unit's security code (845464 by default) then click OK.).
Step 4.	The program will then read and display the settings for Viking phone(s).
Step 5.	After adjusting the IP and other phone's settings, click the "Write" button under each column of settings to send the programming commands to the connected unit.
Step 6.	Press IP Settings menu bar item and set the following parameters: •SIP Server: 172.30.0.1 (the default LAN IP address of QX) •Username: locext115 (the same as configured on QX IP line settings) •Password: ***** (the same as configured on QX IP line settings)
Step 7.	Press Phone Number / Phone Codes menu bar item and define the QX extensions that should be called.
Step 8.	Press Phone Settings menu bar item and set the relay related parameters (Relay mode, Relay activation command, Relay Activation time, etc.).

VOIP.MS as the SIP provider for Viking IP Products

An account must be set up with VOIP.MS. Go to www.voip.ms to set up an account and upon completion, a SIP username and password will be assigned. Their account information page shows a long list of available VOIP servers. You can pick any of these servers to be used as the SIP server for the Viking IP phone. We chose to use the "chicago4.voip.ms" server from that list, so our screenshot shows that particular server as the SIP server.

The "In-band audio call progress" feature under phone settings must be enabled (if not already). When the Viking IP has placed a call and the remote party hangs up, it takes approximately 22 seconds for the SIP server to pass the disconnect signal to the Viking IP phone.

Viking IP Programming - V1.1.4

Name: Entry Phone **MAC:** 18:E8:0F:50:06:8B **IP:** 192.168.210.108

Phone Numbers / Codes / Audio Phone Settings IP Settings Alarm / Firmware / Import / Export / Default / Diagnostics

Emergency Phone Numbers
These numbers are dialed in sequence after pressing the "Call" button on the unit.

First: 7153864355
Second:
Third:
Fourth:
Fifth:

Information Phone Numbers
These numbers are dialed in sequence after pressing the "Info" button on the unit (if present).

First:
Second:
Third:
Fourth:
Fifth:

Phone Codes

Security Code (6 digits): 845464
ID Number (0-6 digits): 0
Access Code (0-6 digits):

Audio File

Loaded Audio File Name: Manual Recording
Upload Wav File (8KHz, Mono, 8 or 16-bit PCM)
Erase Existing Audio File

Checking Alarm Status - Completed Disconnect

In our case, it was programmed to dial 715-386-4355, which is a test phone number here.

Viking IP Programming - V1.1.4

Name: Entry Phone MAC: 18:E8:0F:50:06:8B IP: 192.168.210.108

Phone Numbers / Codes / Audio Phone Settings IP Settings Alarm / Firmware / Import / Export / Default / Diagnostics

Internal / External Relay Ext 1 In-Band Audio Call Progress Enabled

Relay Mode Door Strike In-Band Audio Detect Sensitivity 5

Relay Activation Command 5 Repeat Announcement Option 1

Relay Activation Time 15 sec Lap Counter 1

Relay Buzz Volume 2 Call Length Time Out 1 min

Relay Latch Commands Enabled Inbound Call Mode Auto Answer

Alternating Switch Action Enabled Ring Cadence Normal Ring

Speaker Mode On Dial Next No. on Ring No Answer 7

Speaker Volume 1 Dial Next Number on Busy Enabled

Ring Volume 1 Send ID Number as RFC 2833

Microphone Volume 4 "Call" LED Mode Emergency Phone

Talk / Listen Delay (VOX) .5 sec "Call" LED Control Automatic

Disconnect

Note that "In-Band Audio Call Progress" is enabled.

Viking IP Programming - V1.1.4

Name: Entry Phone MAC: 18:E8:0F:50:06:8B IP: 192.168.210.108

Phone Numbers / Codes / Audio Phone Settings IP Settings Alarm / Firmware / Import / Export / Default / Diagnostics

Set Unit IP Address via DHCP Unit Name Entry Phone

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 2.viking.pool.ntp.org

SIP Server Settings - Not Registered

SIP Server chicago4.voip.ms

Username 190106

Password Bear987654!

Caller ID 190106

Gateway Ping / Registration Time

Ping Timer (Sec) 10

Registration Timer (Min) 30

Codec Selection and Negotiation Order

Codecs G711a, G711u, G722

Disconnect

Note the SIP server settings. When we set up our account with VOIP.MS, our SIP username was "190106" and our SIP password was "Bear987654!".