Viking VoIP SIP System Compatibility List

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

On-Premise	Cloud Based Service Provider
3COM VCX	Callcentric*
3CX	iptel.org
Allworx*	MetaSwitch
Aastra	Ring Central*
Asterisk	sip.antisip.com
Atcom	Switchvox
Avaya Aura Platform	unify
Avaya IP Office Platform	Vertical Wave*
BlueBox	Voice Carrier
Brekeke	VoIP.MS*
Cisco Unified Communications Manager (CUCM)*	Wildix*
Cisco Unified Communications Manager Express (CUCME)	
Elastix	
epygi QX200*	
Freeswitch	
Grandstream*	
Interactive Intelligence	
iPECS (Ericsson-LG)*	
Iwatsu ECS*	
Kamailio	
Mitel 3300	
NEC	
OfficeSIP	
OpenSIPS	
Panasonic** (with SIP Extension Card)	
PolyCom*	
Samsung Communications Manager (SCM)	
ShoreTel*	
Siemens Communications Server (SCS)	
SIP Express Router (SER)	
Snom PBX	
Sonus	
Switchvox	
Teksip	
Toshiba	
Vertical Wave*	
Yealink T Series SIP Phones	

*** Note:** *Additional programming information on the following pages.*

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

Allworx:

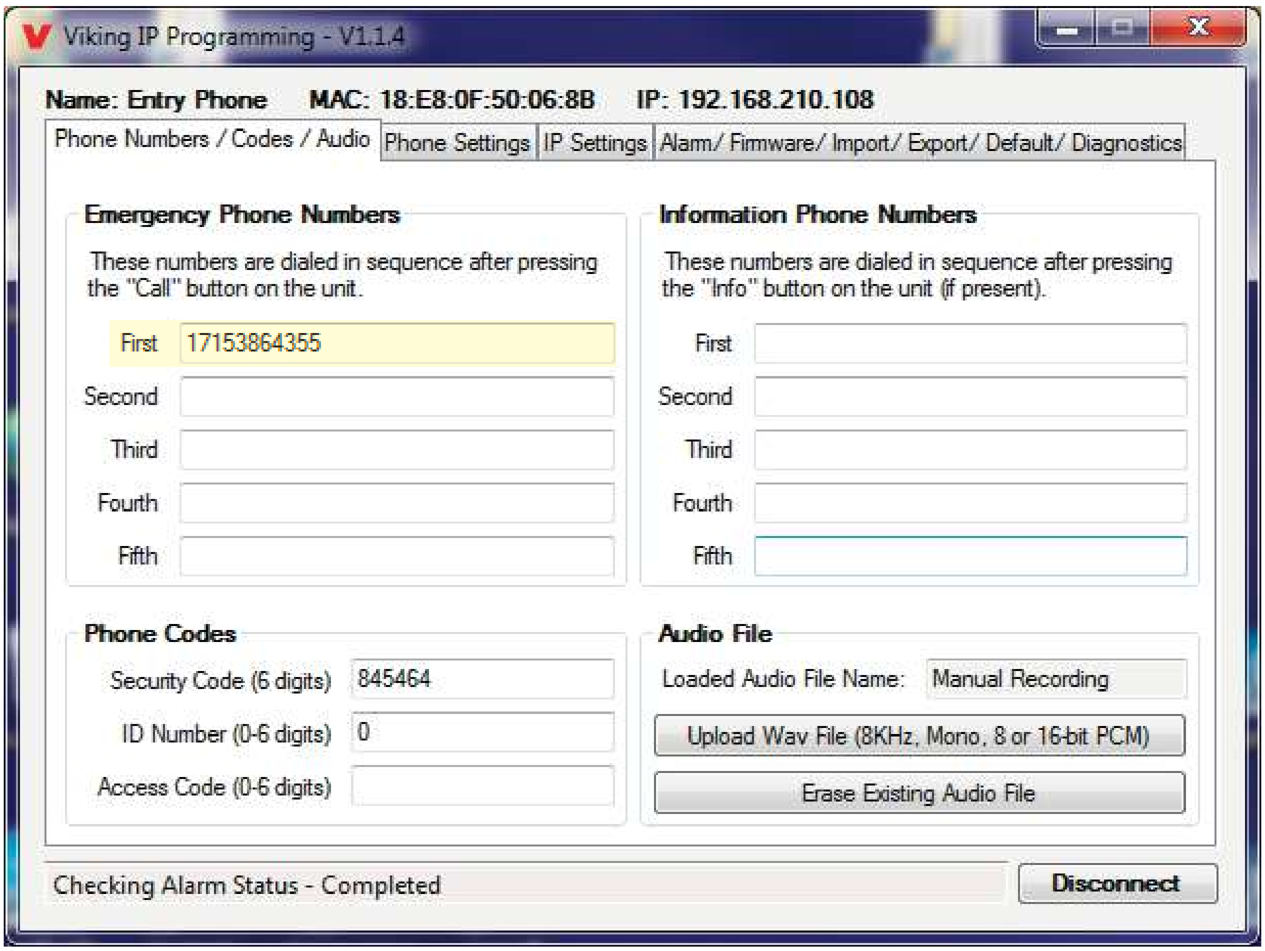
Make “user ID” and “Login ID” the same in the user account.

Callcentric as the SIP provider for Viking IP Products:

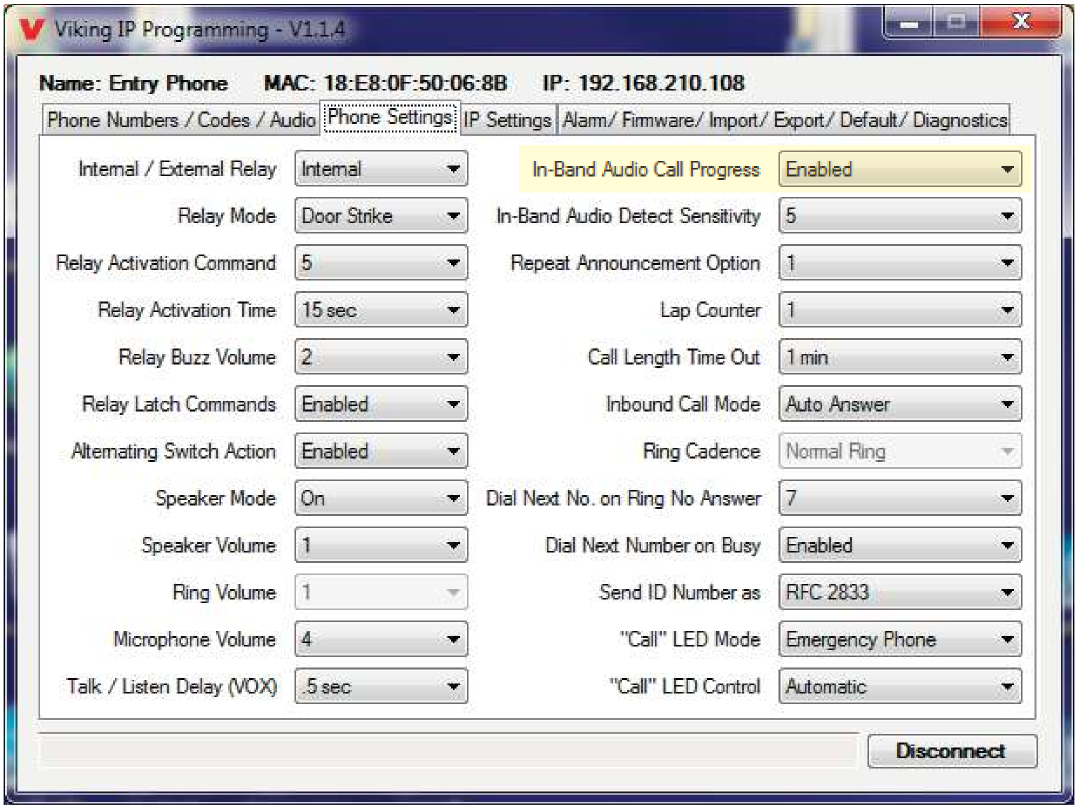
An account must be set up with Callcentric. Go to www.callcentric.com to set up an account and upon completion, a SIP username and password will be assigned. With Callcentric service, the SIP server will always be “callcentric.com”.

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

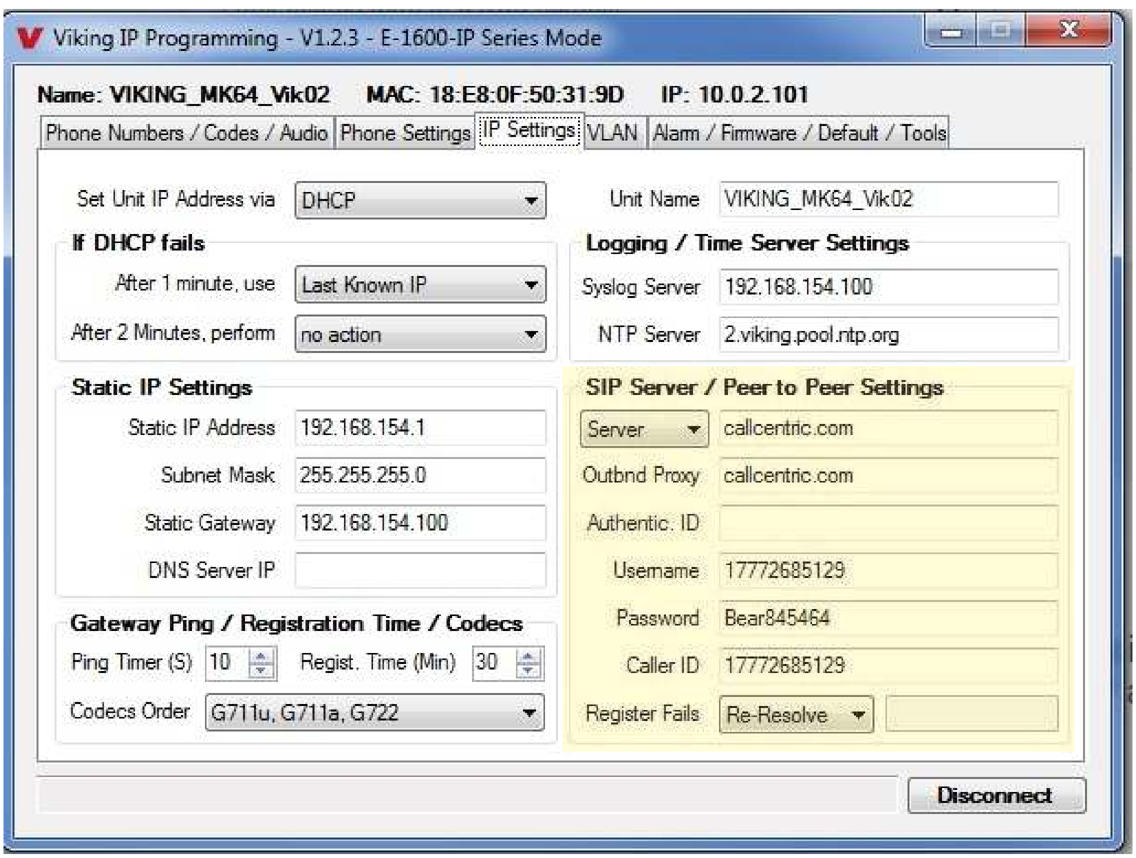
When the Viking IP has placed a call and the remote party hangs up, it takes approximately 22 seconds for the SIP server to pass the disconnect signal to the Viking IP phone.



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.



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



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.

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).

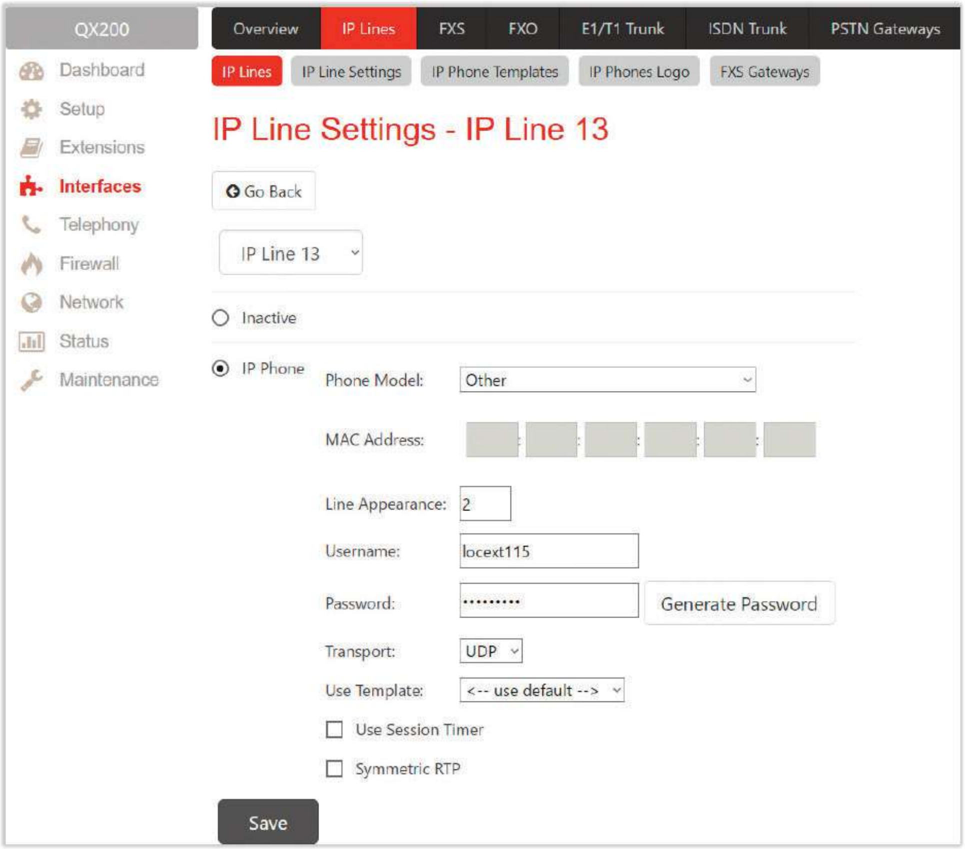


Figure 1: IP Line Settings page

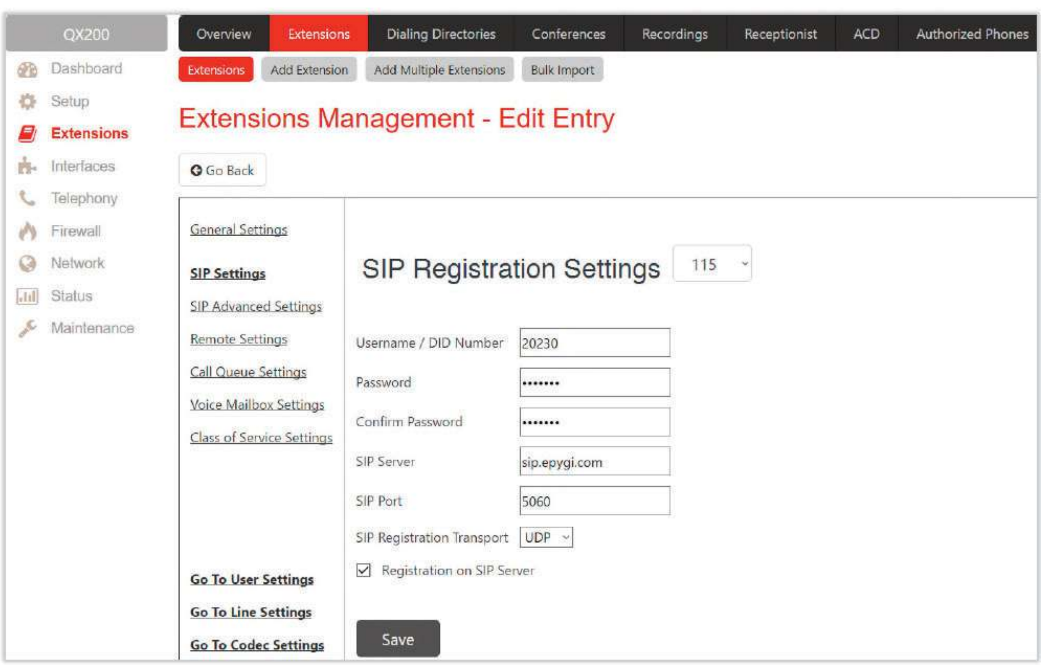


Figure 2: SIP Settings section

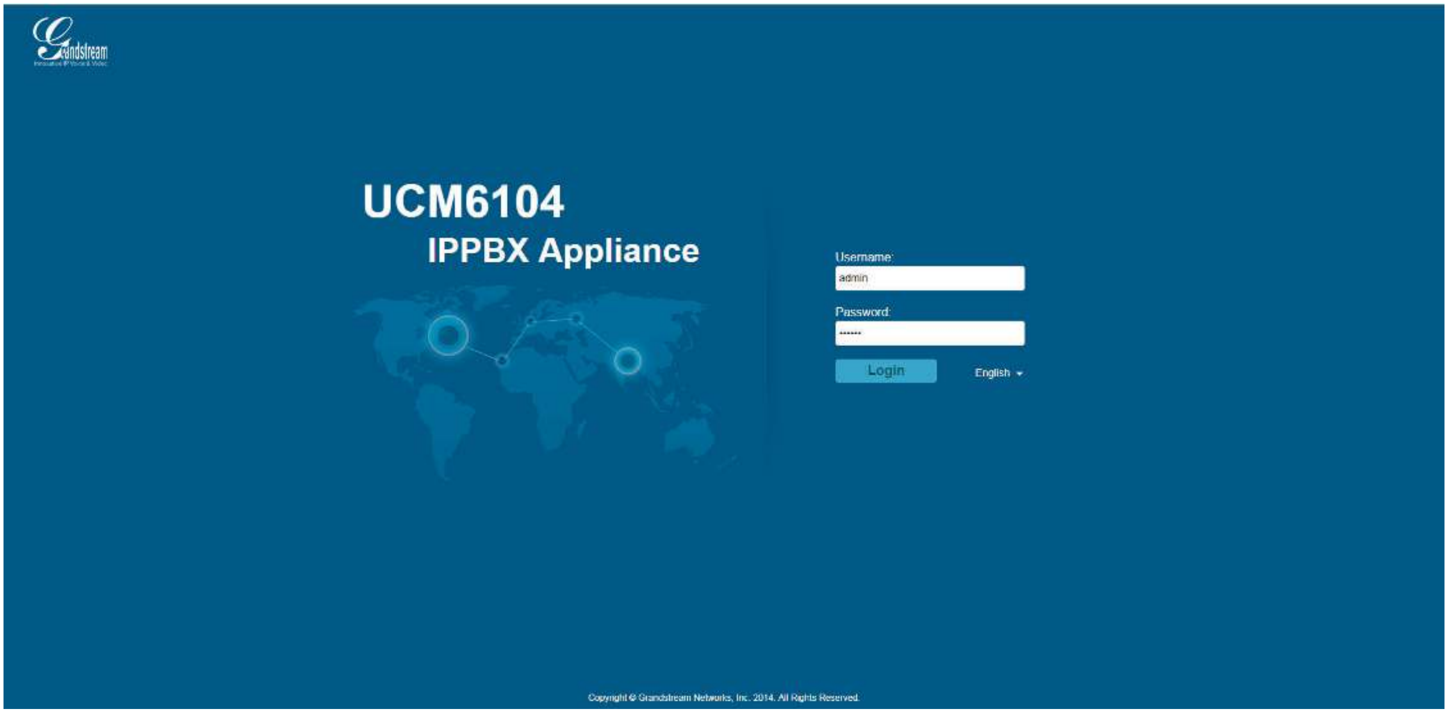
B. Configuring the Viking VoIP Phone

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:

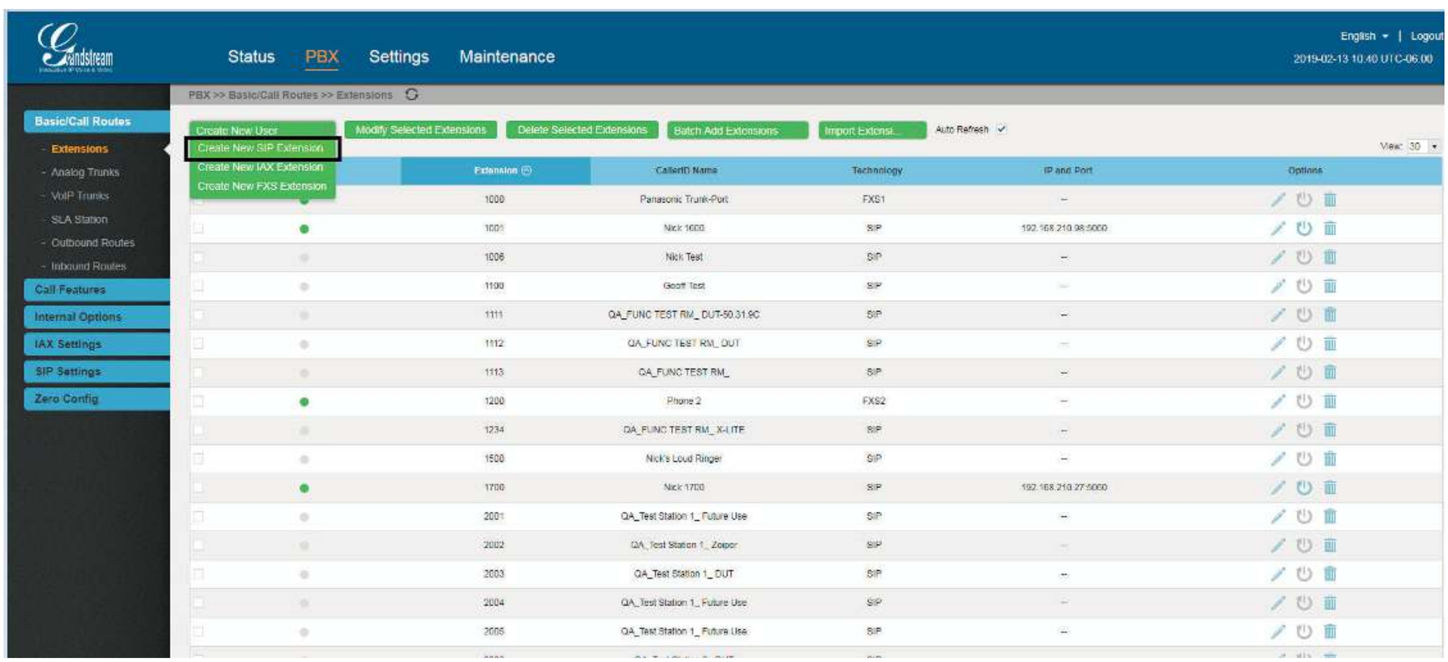
Step 1.	Open the Viking VoIP Phone Programming software on the MS Windows PC that is connected to the same LAN as the Viking phone to be programmed.
Step 2.	The window in the upper left corner of the menu will show you the Viking phone(s) that is connected to that LAN. Select the unit with the same MAC 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.).

Configuring Viking IP phones with Grandstream PBX devices:

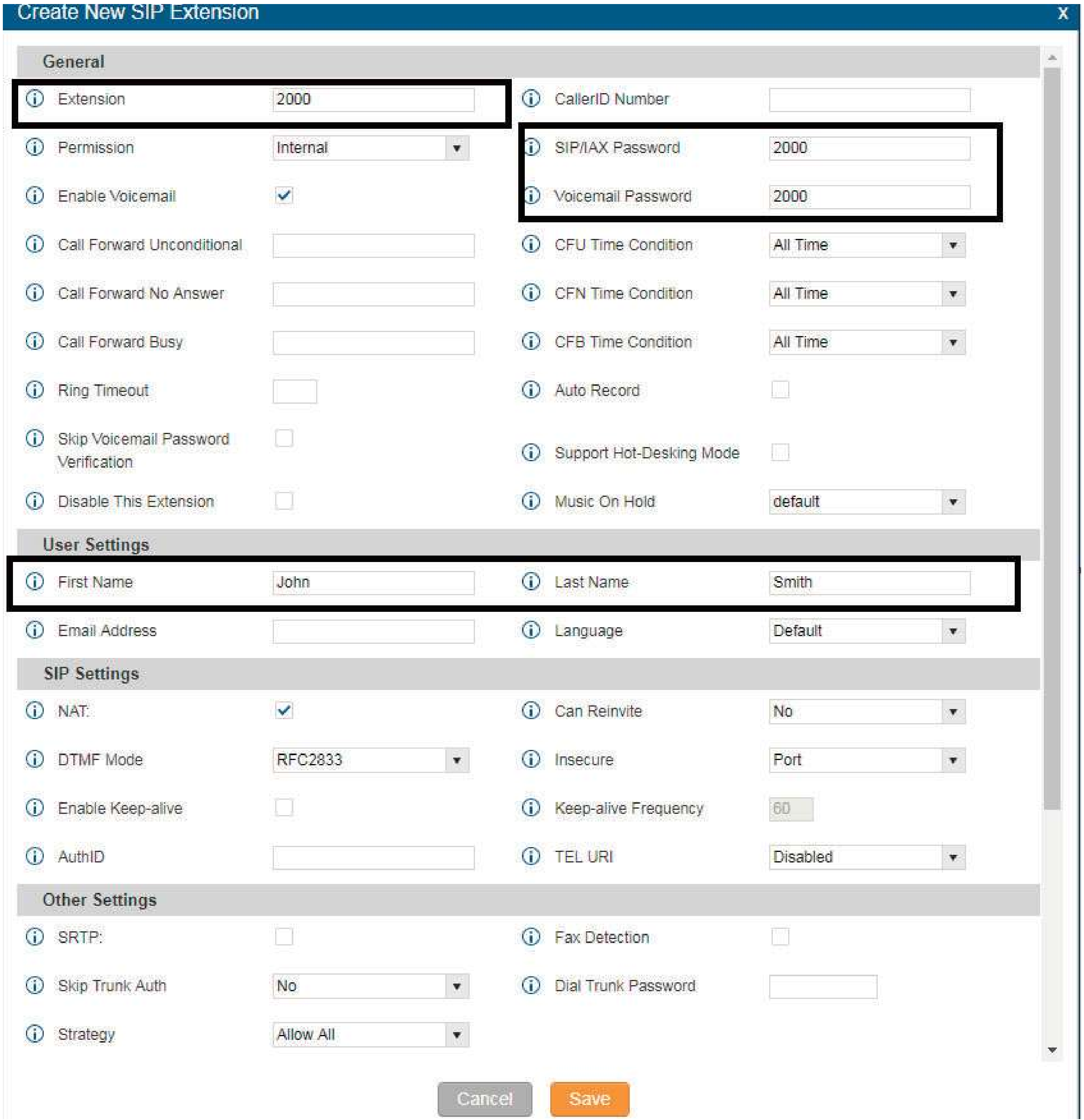
1. Log into the Grandstream configuration tool in your browser:



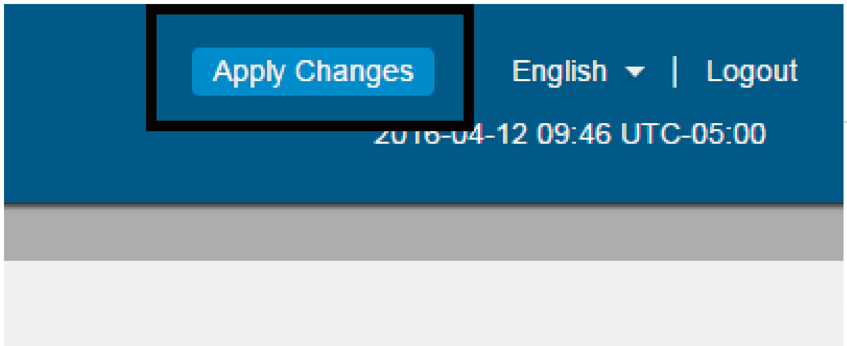
2. Click on “PBX” at the top, and then “Create New User” and “New SIP Extension”



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”.

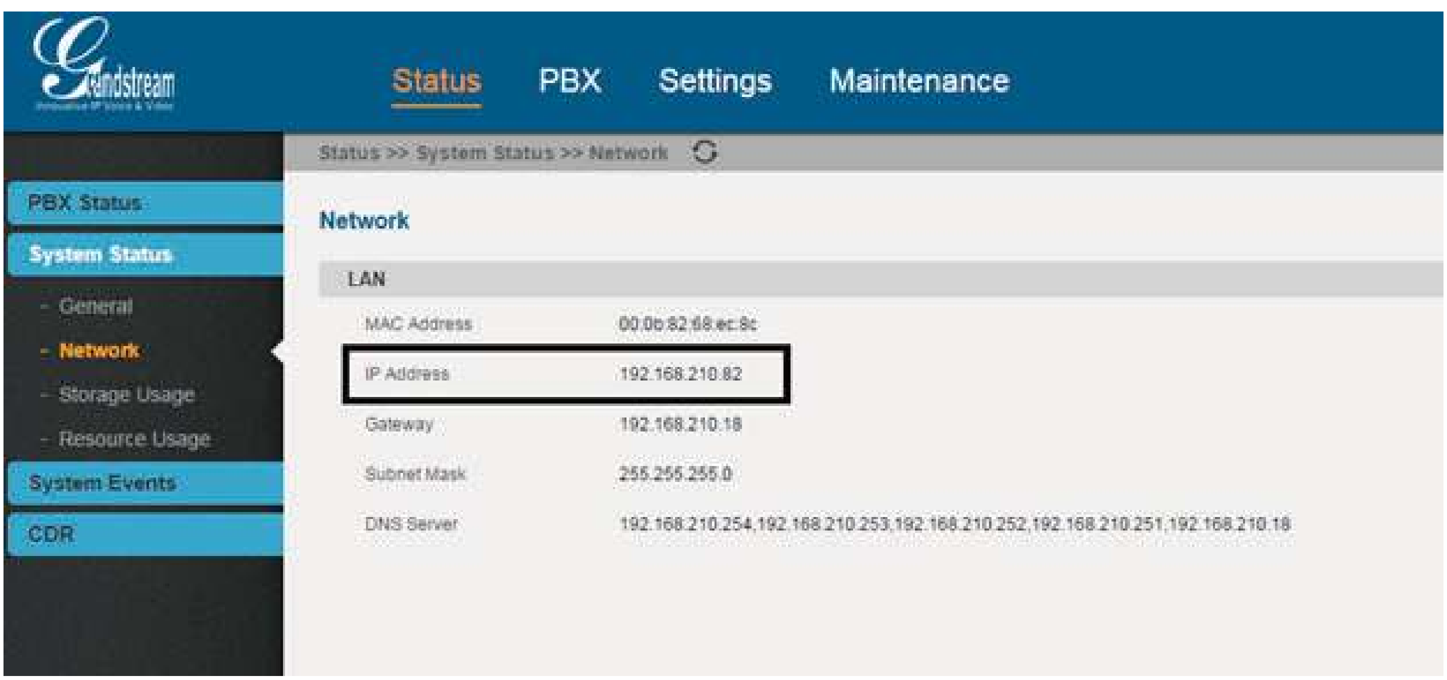


4. After saving, click on the “Apply Changes” button in the top right corner.

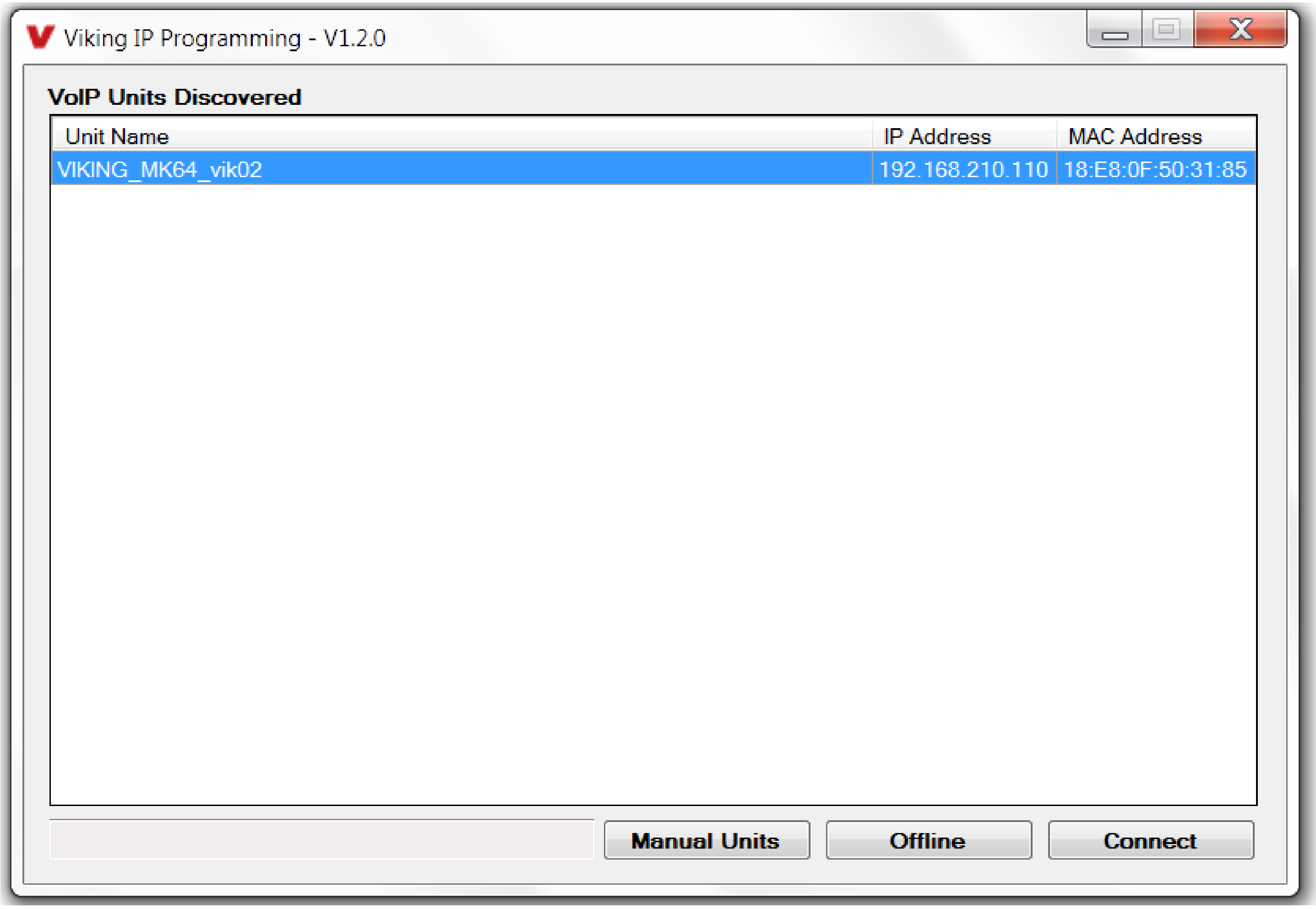


5. You will then see the extension you set up under the list of extensions.

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”.



Enter the Grandstreams IP as the domain. Enter valid extension and Password.

iPECS (Ericsson-LG):

Make sure 407 Register is “OFF’ otherwise it wants a proxy.

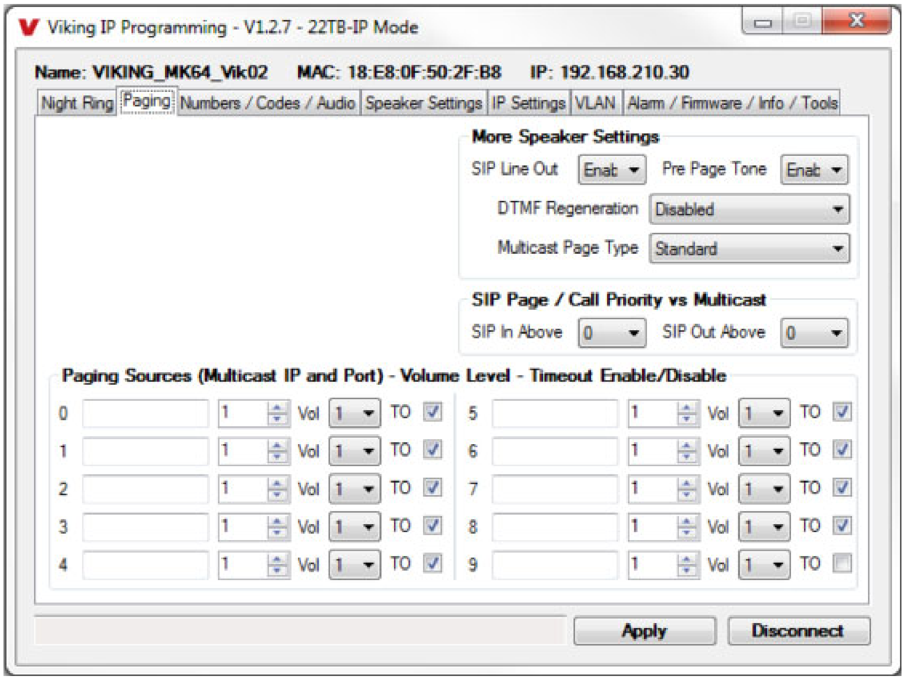
Iwatsu ECS:

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:

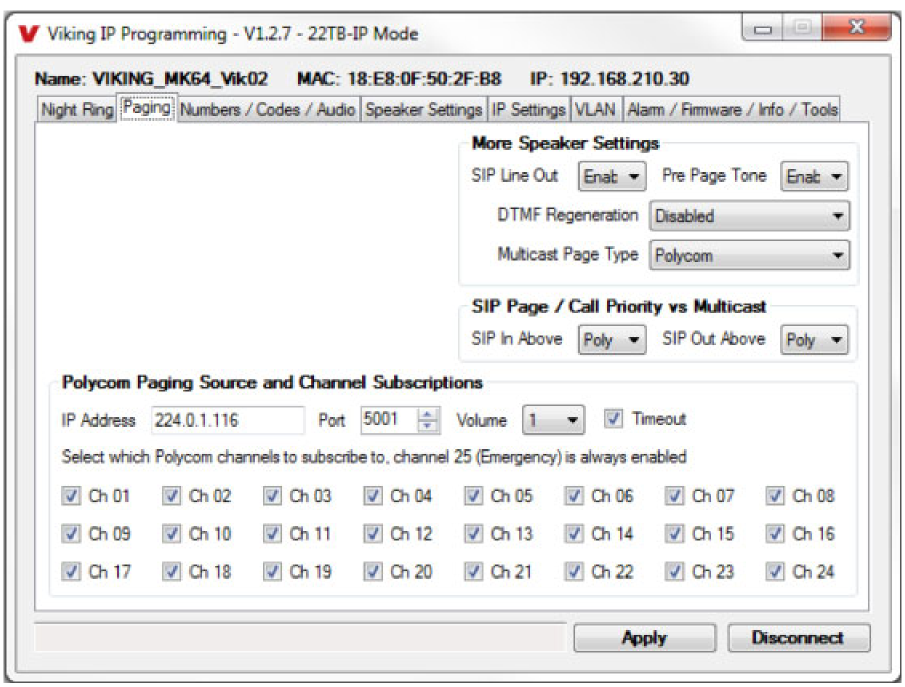
Step 1.	Gather the following information about the phone: • MAC address • Physical location of the Viking Voip Phone • CCSU or LAN2 IP Address of the Iwatsu ECS System • Extension Number for the phone or hunt group that will be called when the button is pressed • Extension Number to assign to the Viking 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.
Step 5.	Audio Call Progress must be set to ‘Enabled.’
Step 6.	Make calls with the Viking VoIP Phone.

Multicast Paging:

Standard: A Viking IP speaker can listen to up to 10 "Standard" multicasts, and play them based on their priority level (0 is the highest priority and 9 is the lowest). Enter the Multicast IP address and Port into the Paging sources fields. Any multicast at this address/port will be heard from the Viking Speaker/Paging Adaptor.



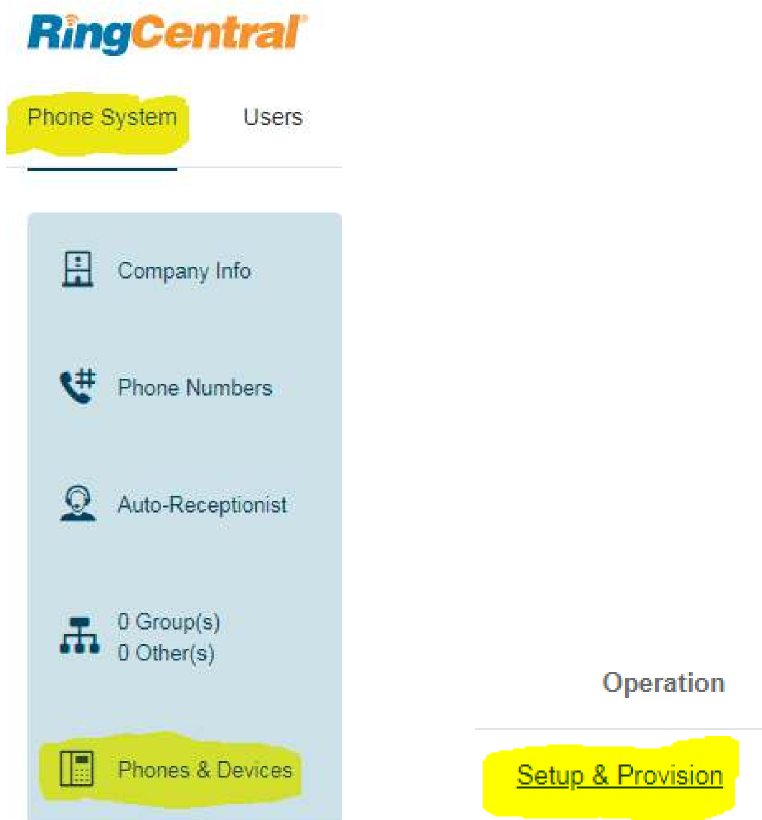
Polycom: Under the "Paging" Tab, "More Speaker Settings" select Polycom as the "Multicast Paging Type". Polycom phones will use 224.0.1.116 as the default IP Address and port 5001. Select the channels you want to enable, channel 25 cannot be disabled as it is generally used for emergency broadcasts. Any multicast page made from a Polycom phone (on the same LAN) to the desired IP address and port will be played from the Viking Speaker/Paging Adaptor. The priority of Polycom sources ranges from 1 (lowest) to 25 (highest). SIP paging and outbound SIP calls can have priority over Polycom paging (factory setting). If the SIP priority is set to "None" than Sip paging/outbound calls are not allowed during a Polycom multicast.



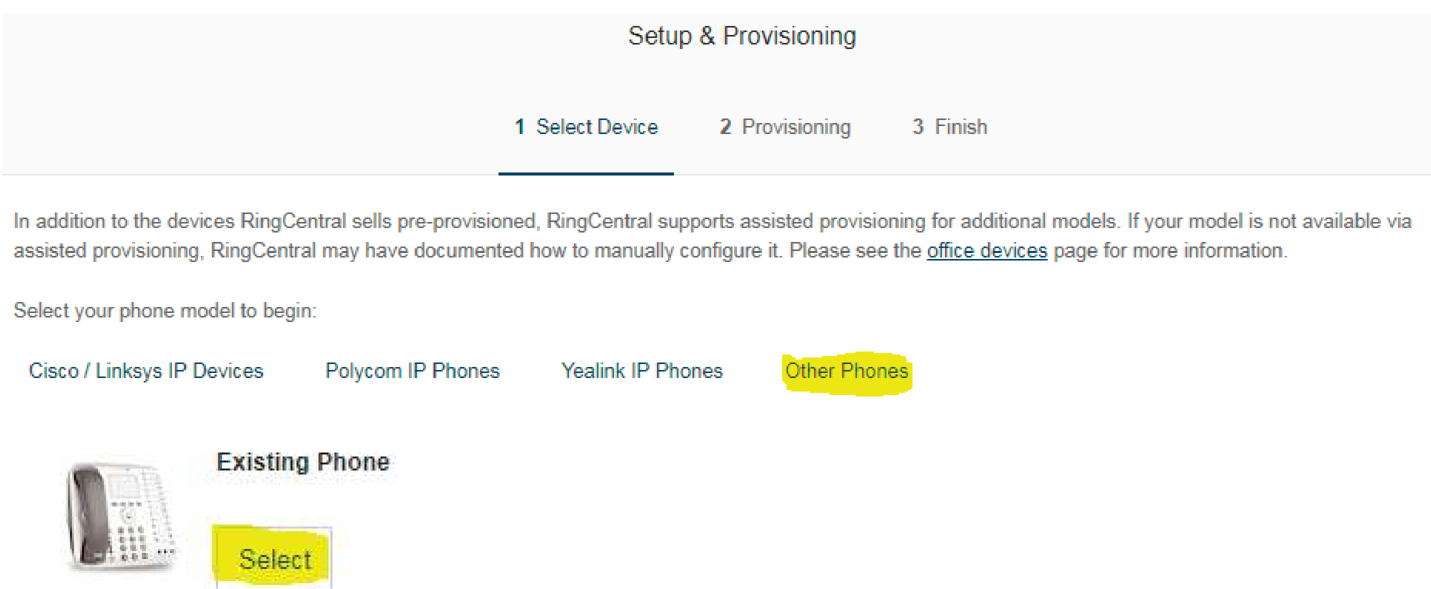
RingCentral as the provider for Viking IP Products:

Note – RingCentral requires the “outbound proxy” and “authentication ID” fields when a SIP device is configured for their service. Only MK64 version Viking devices have the capability to support these fields. If you are not sure if the Viking device you have is compatible with RingCentral, contact Viking Technical Support.

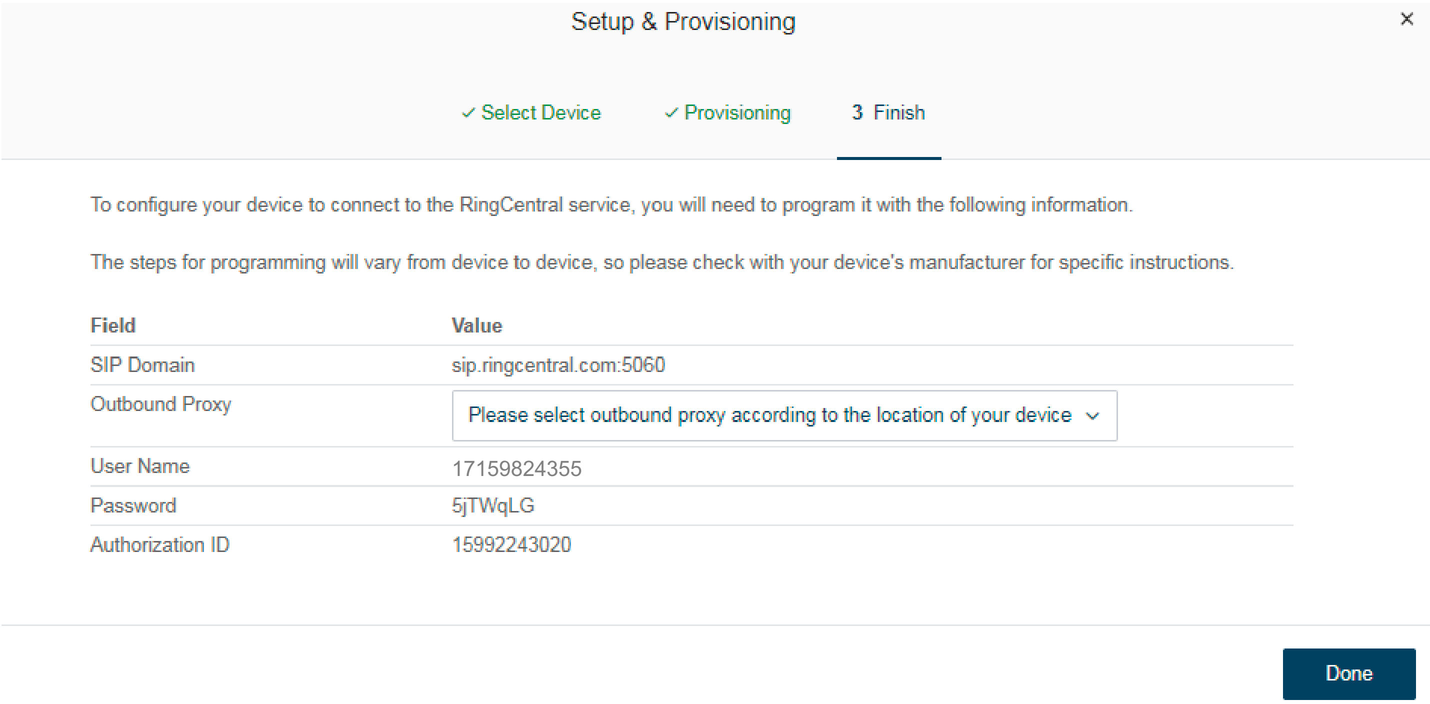
Go to ringcentral.com to set up an account and select a service plan. Once you have created your RingCentral account and are logged in to the account, go to “Phone System”, “Phones & Devices” and “Setup & Provision”.



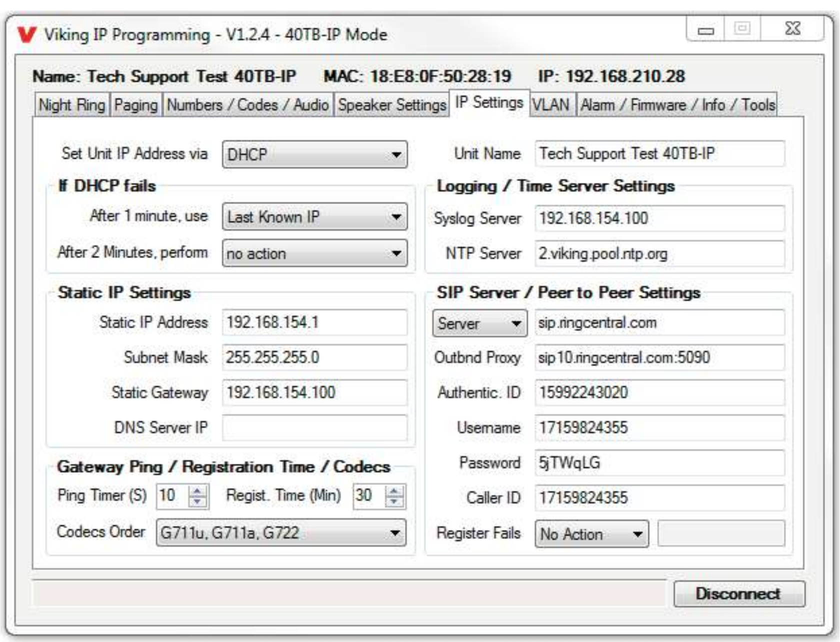
On the Setup & Provisioning page, click on “Other Phones” and the “select” by “Existing Phone”.



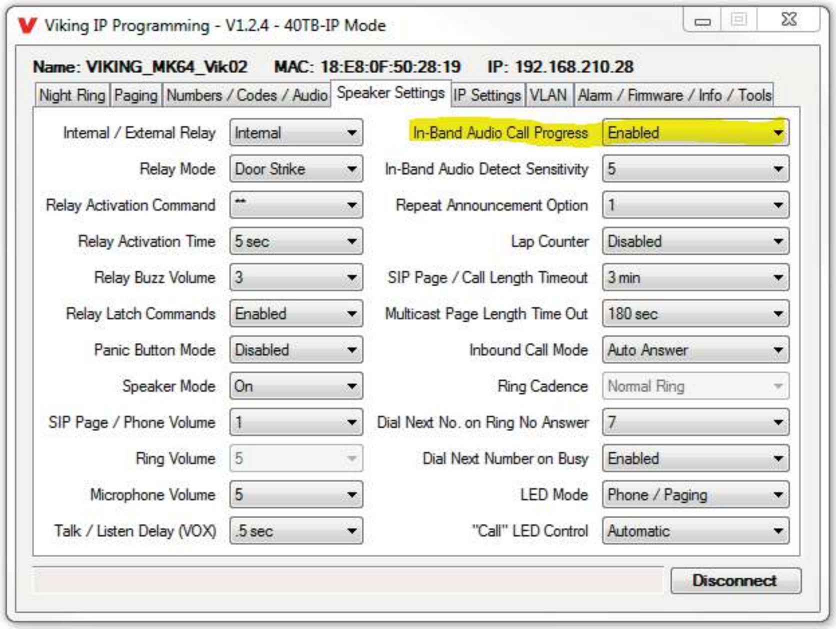
They will then provide all of the values used to configure the Viking device. Click on the down arrow by “outbound proxy” and select from one of the choices based on your physical location. North America is either “sip10.ringcentral.com:5090” or “sip20.ringcentral.com:5090”. Print this RingCentral page or note all of the values so you can use them when programming the Viking device.



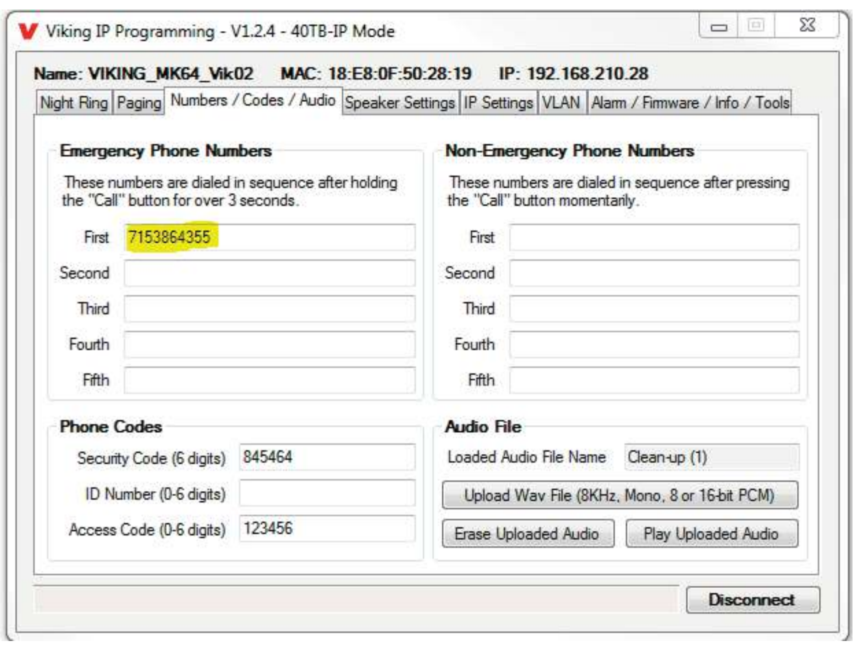
Open Viking IP Programming software, connect to the device you wish to configure with RingCentral and go to the “IP Settings” tab. The “SIP Domain” provided by RingCentral goes in the SIP Server field in IP Programming (the “:5060” port selection does not need to be entered as the Viking devices default to port 5060). The “Outbound Proxy” provided by RingCentral goes in the Outbnd Proxy field in IP Programming (the “:5090” port selection must be entered). The “User Name” provided by RingCentral goes in the Username and Caller ID fields in IP Programming. The “Password” provided by RingCentral goes in the Password field in IP Programming. The “Authorization ID” provided by RingCentral goes in the Authentic. ID field in IP Programming.



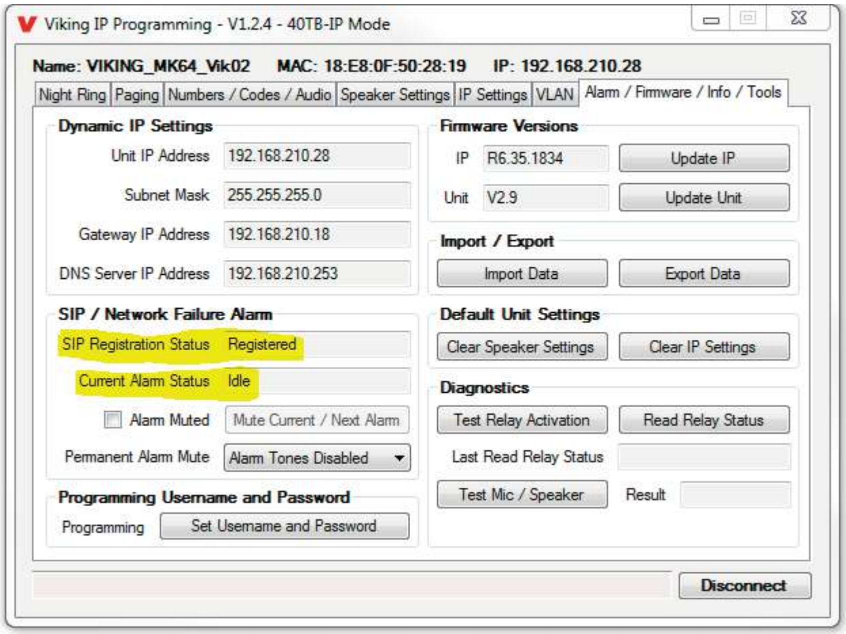
On the “Phone Settings” or “Speaker Settings” tab of the IP Programming software, make sure the “In-Band Audio Call Progress” feature is enabled.



On the “Phone numbers” or “Numbers” tab of the IP Programming software, enter the phone numbers the Viking device should call, if it will be used for outgoing calls. A “1” is not needed before the telephone numbers.



Apply your changes and you will be disconnected from the device. Reconnect to the device in the IP Programming software and go to the “Alarm/Firmware” tab of the IP Programming software to see if the Viking device is now registered with the RingCentral service. If it is registered, the “SIP Registration Status” will be Registered and “Current Alarm Status” will be Idle, like this:



If it is not registered, go back to the "IP Settings" tab in IP Programming software. Make sure you entered all of the SIP information correctly (without any typos) and included the ":5090" port selection at the end of the outbound proxy field.

ShoreTel Ring Group Limitation:

Viking IP products are not capable of dialing the access code for a ShoreTel ring group but the ShoreTel system can be programmed so that multiple phones ring at the same time when the IP product calls. This is how it is accomplished:

Step 1.	Create a "virtual" extension in the Shoretel system for our IP phone to call. The Viking IP product should be programmed to call the virtual extension.
Step 2.	Program all phones that need to ring when the IP product calls to "monitor" the virtual extension, so their phone rings any time the virtual extension rings.

Viking IP products can then ring a number of phones at the same time. The call can be answered by any phone and the relay command can be dialed to release a door/gate. The extension of the Viking IP product can be called if a user wants to control the door/gate without receiving a call from the Viking IP product.

Vertical Wave:

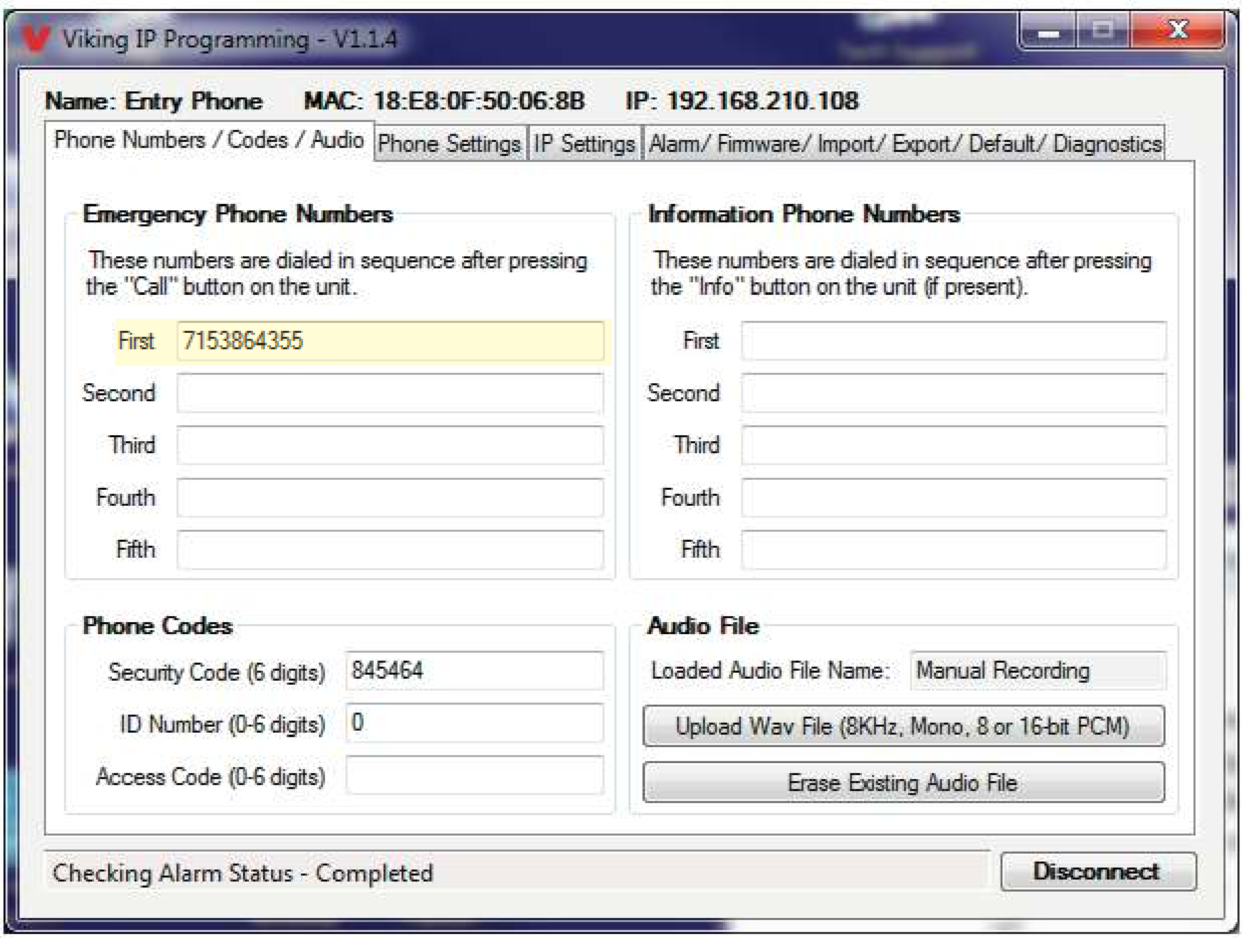
Go to “IP Telephony” and under “SIP”, “Advanced Authorization” and then “Global Adv. Parameters” uncheck “Authenticate Register”.
Our username = extension number of the wave phone.
Our password = SIP authentication password.

VOIP.MS as the SIP provider for Viking IP Products

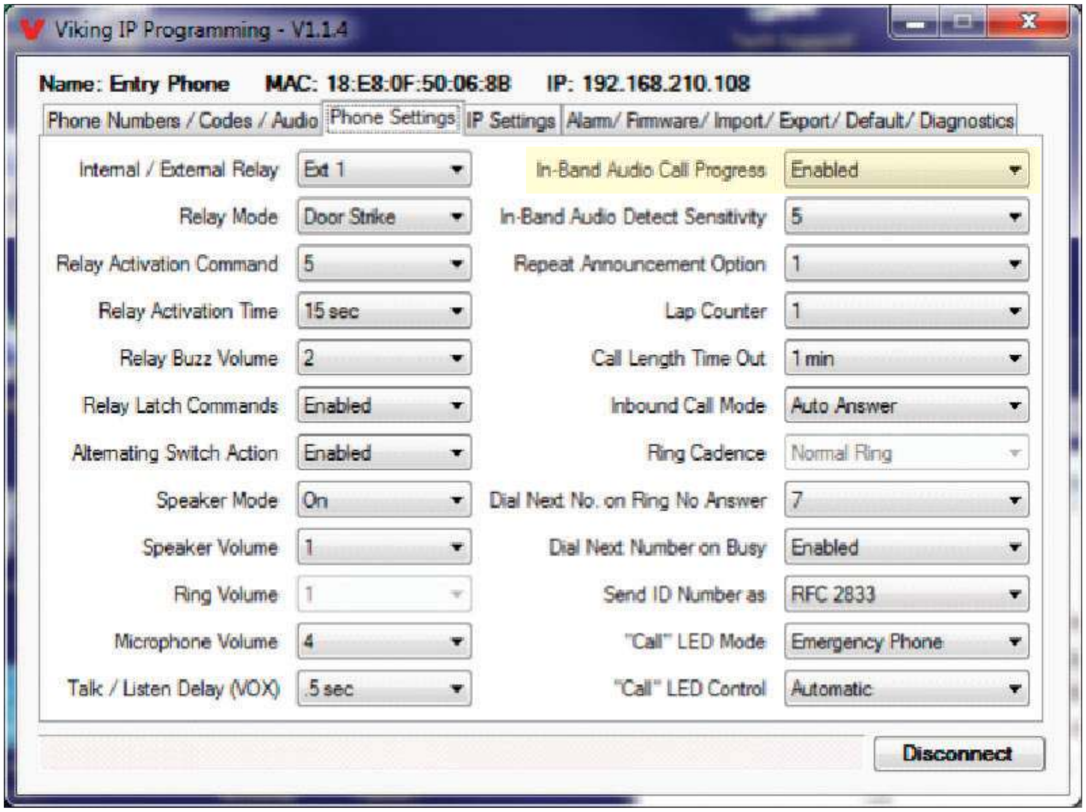
An account must be set up with VOIP.MS. Go to www.voip.ms to set up an account and upon completion, a SIP username and password will be assigned. Their account information page shows a long list of available VOIP servers. You can pick any of these servers to be used as the SIP server for the Viking IP 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).

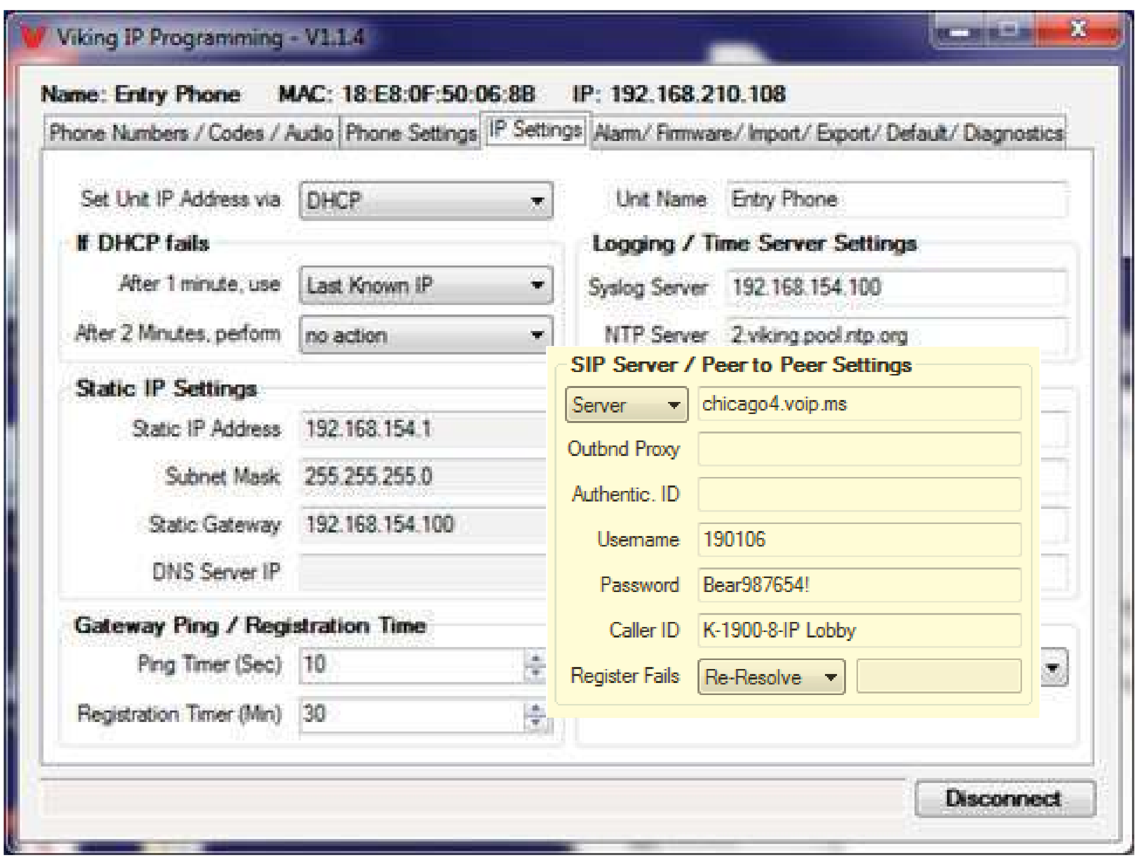
When the Viking IP has placed a call and the remote party hangs up, it takes approximately 22 seconds for the SIP server to pass the disconnect signal to the Viking IP phone.



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



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



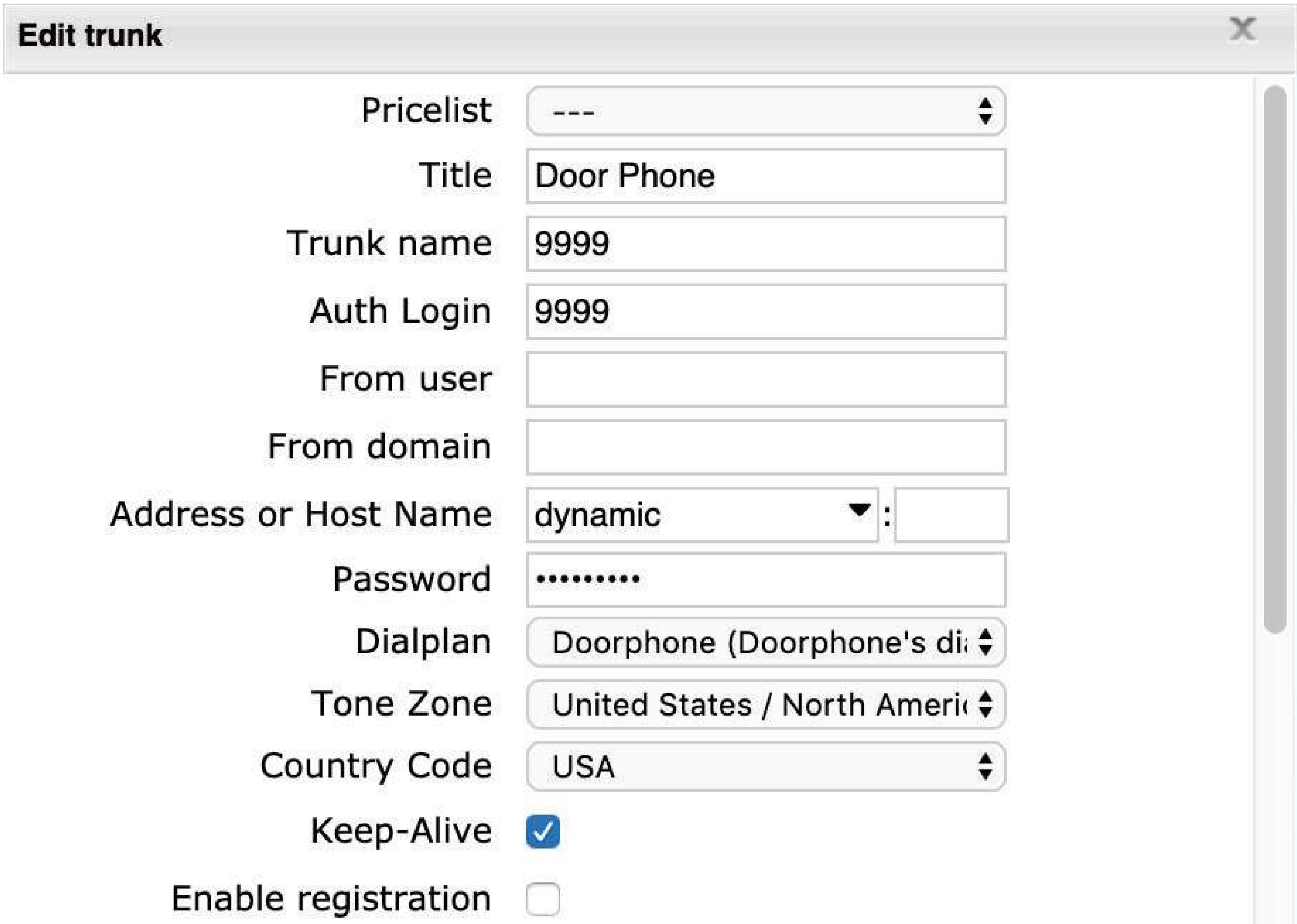
Note the SIP server settings. When we set up our account with VOIP.MS, our SIP username was "190106" and our SIP password was "Bear987654!".

Wildix:

A Wildix Configeration:

The Viking device must register with Wildix as a SIP trunk to make sure touch tones are provided to the Viking IP product for relay control or other functions.

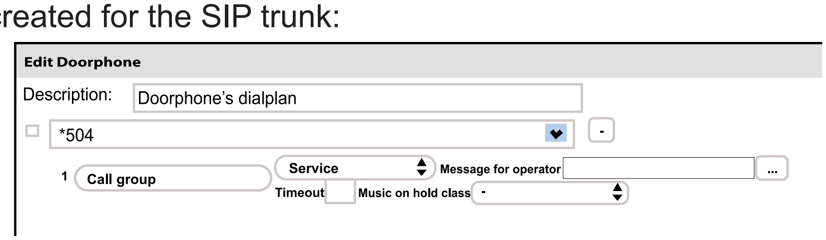
SIP Trunk Configuration:



Enter the same value in both the “Trunk name” and “Auth Login” fields, like the “9999” shown above. Note that the “Trunk name”, “Auth Login” and “Password” fields assigned on the “Edit Trunk” page will be used in the “username”, “authentication ID” and “password” fields when configuring the Viking IP product.

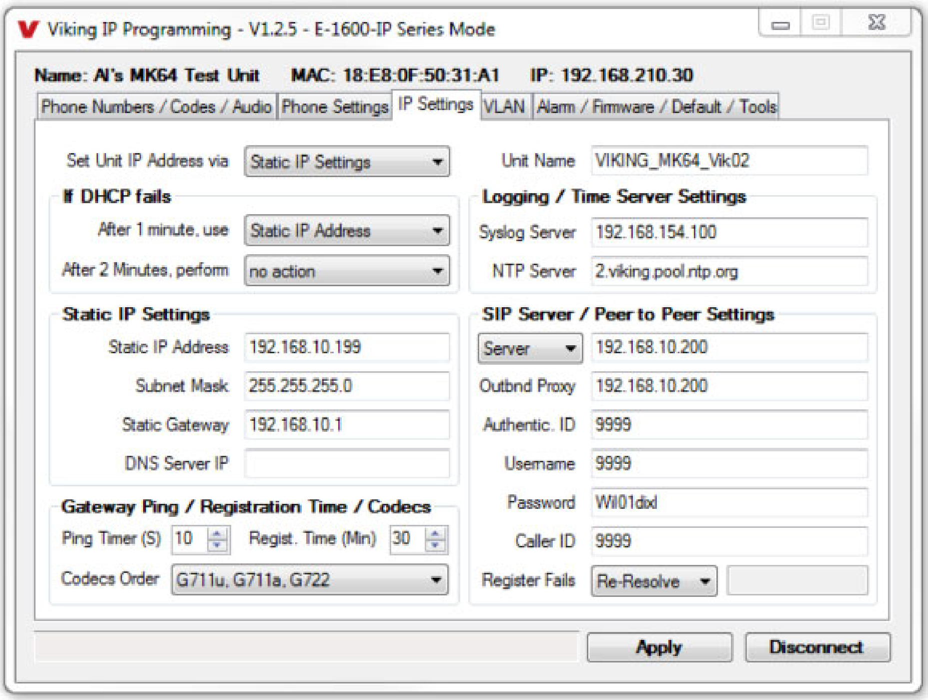
Make sure the trunk is configured to provide touch tones to the IP product:

A dial plan must be created for the SIP trunk:

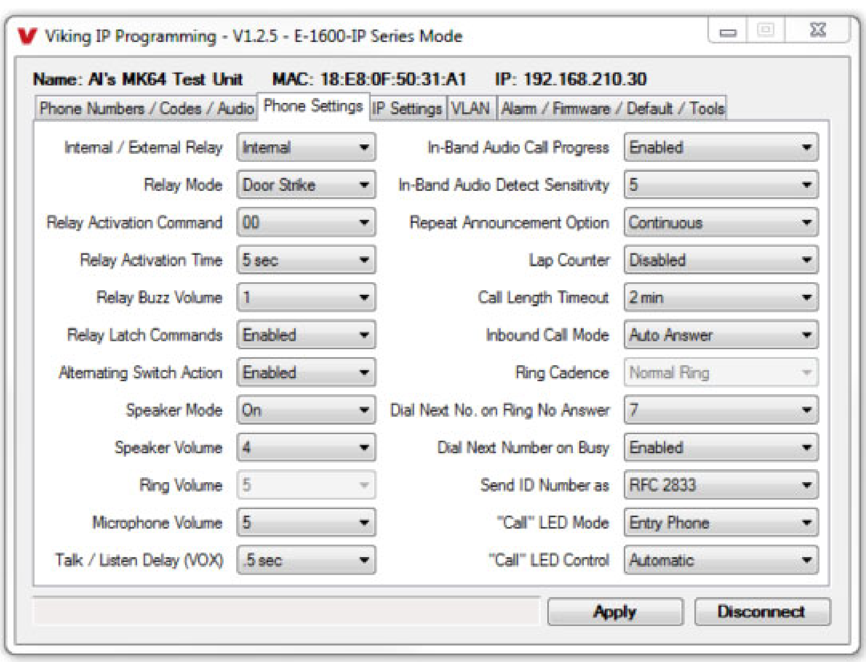


B. Viking IP product configuration:

Open Viking IP Programming software, connect to the device you wish to configure with Wildix and go to the “IP Settings” tab. Use the URL or IP address of the Wildix SIP server in the SIP Server field in IP Programming and leave the “Outbound Proxy” field blank. The “Trunk name” from the Wildix trunk configuration goes in both the Username and Caller ID fields in IP Programming. The “Password” from the Wildix trunk configuration goes in the Password field in IP Programming. The “Auth Login” from the Wildix trunk configuration goes in the Authentic. ID field in IP Programming.



On the “Phone Settings” or “Speaker Settings” tab of the IP Programming software, make sure the “In-Band Audio Call Progress” feature is enabled.



On the “Phone Numbers” or “Numbers” tab of the IP Programming software, enter the extension numbers or phone numbers the Viking device should call, if it will be used for outgoing calls.

Apply your changes and you will be disconnected from the device. Reconnect to the device in the IP Programming software and go to the “Alarm/Firmware” tab of the IP Programming software to see if the Viking device is now registered with the Wildix service. If it is registered, the “SIP Registration Status” will be Registered and “Current Alarm Status” will be Idle, like this: