# 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

| On-Premise | See Additional info on following page |
|---|---|
| 3COM VCX | |
| 3CX | |
| Allwork | ✓ |
| Aastra | |
| Asterisk | |
| Atcom | |
| Avaya Aura Platform V6.2.9 or earlier | |
| Avaya IP Office Platform | |
| BlueBox | |
| Brekeke | |
| Cisco Unified Communications Manager (CUCM) | |
| Cisco Unified Communications Manager Express (CUCME) | |
| Elastix | |
| epygi QX200 | ✓ |
| Freeswitch | |
| Grandstream | ✓ |
| Interactive Intelligence | |
| iPECS (Ericsson-LG) | ✓ |
| Iwatsu ECS | ✓ |
| Kamailio | |
| Mitel 3000 | |
| NEC | |
| OfficeSIP | |
| OpenSIPS | |
| Panasonic* (with SIP Extension Card) | |
| PolyCom (SIP paging only) | |
| 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 | |

| Cloud Based Service Provider | See Additional info on following page |
|---|---|
| Callcentric | ✓ |
| iptel.org | |
| MetaSwitch | |
| Ring Central | |
| sip.antisip.com | |
| Switchvox | |
| unify | |
| Vertical Wave | ✓ |
| Voice Carrier | |
| VoIP.MS | ✓ |

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

## Allworx:

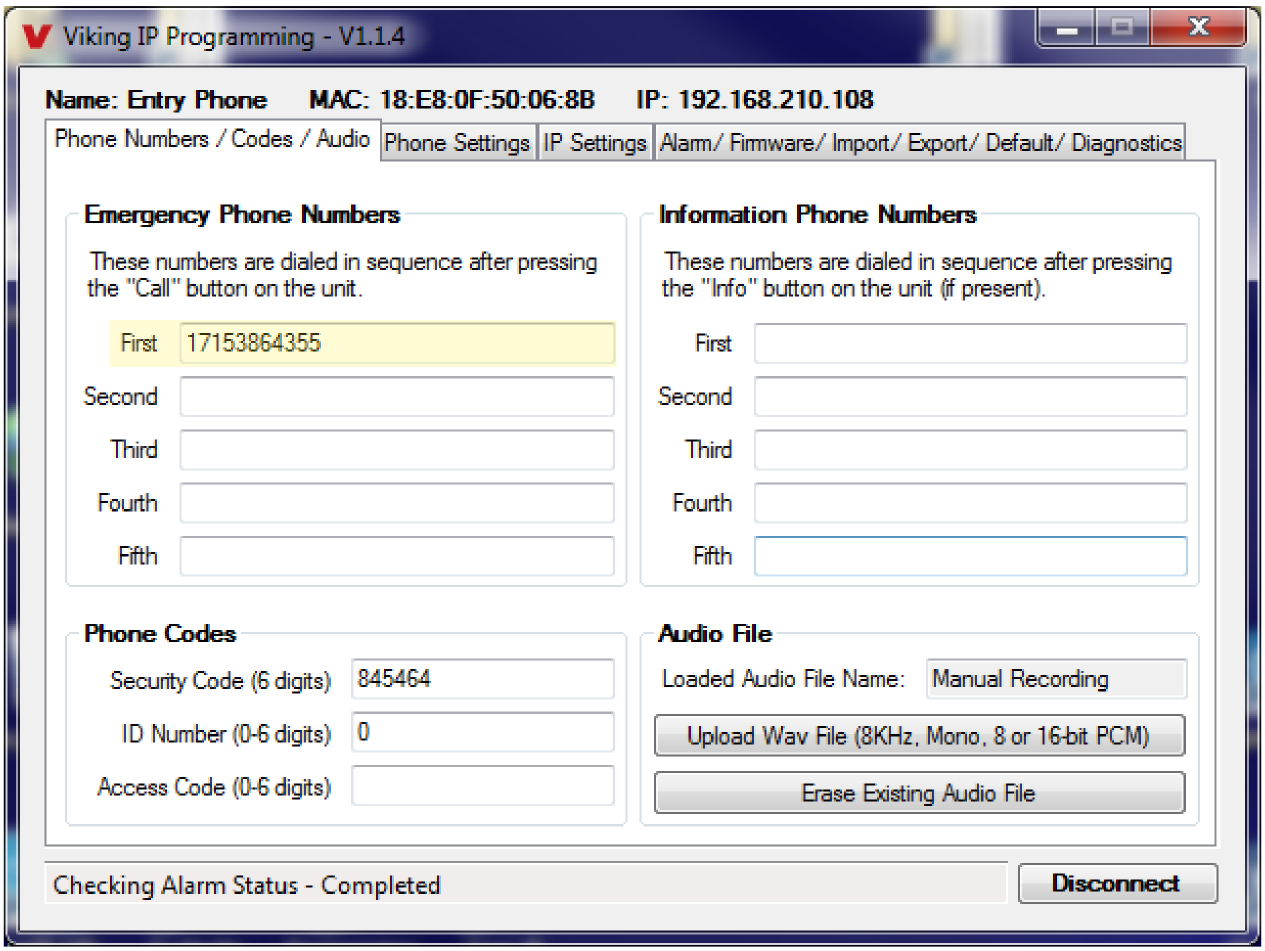
Make “user ID” and “Login ID” the same in the user account.

## Callcentric as the SIP provider for Viking IP Products:

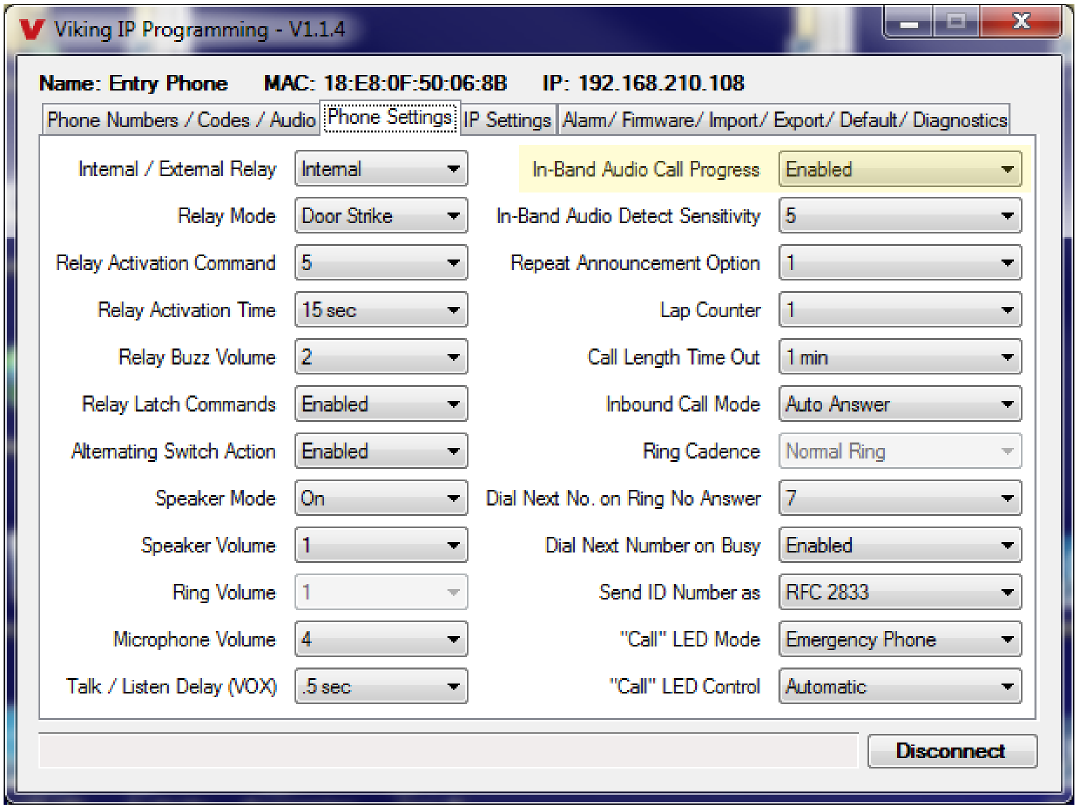
An account must be set up with Callcentric. Go to www.callcentric.com to set up an account and upon completion, a SIP username and password will be assigned. With Callcentric service, the SIP server will always be “callcentric.com”.

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

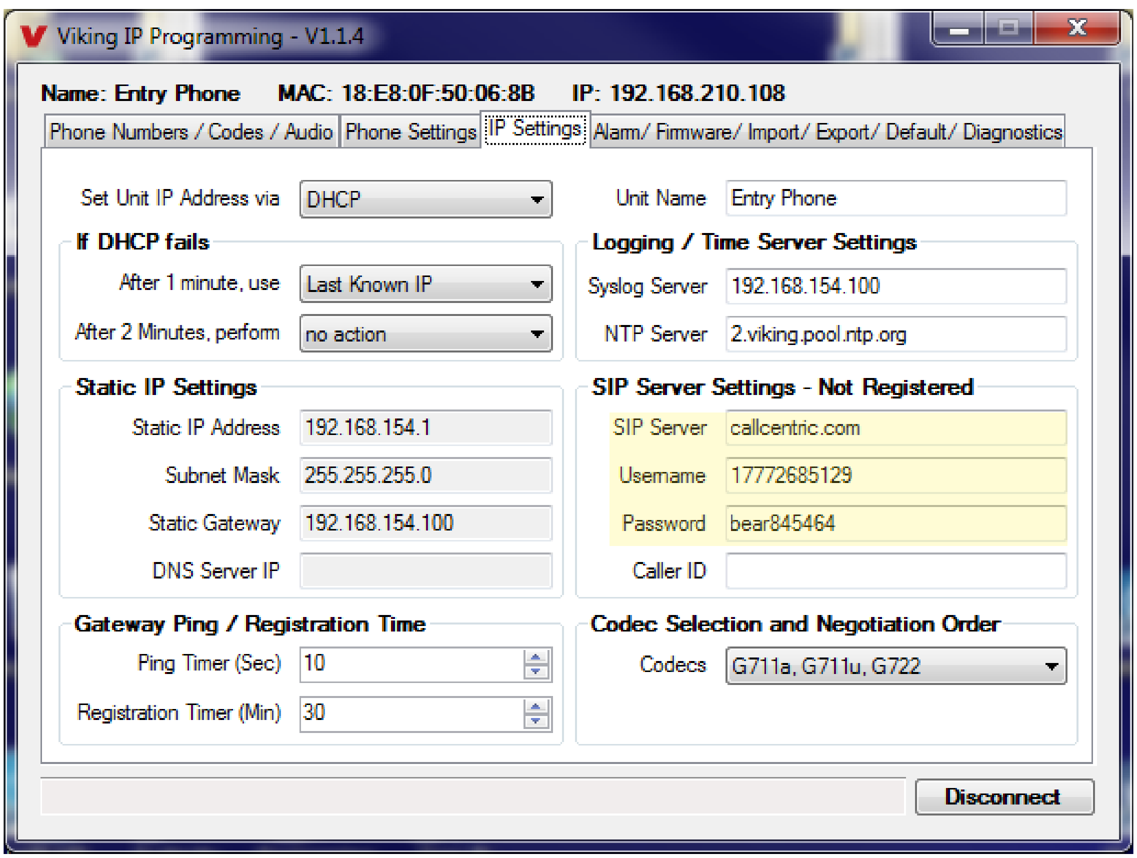
When the Viking IP has placed a call and the remote party hangs up, it takes approximately 22 seconds for the SIP server to pass the disconnect signal to the Viking IP phone.




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:<br>• MAC address<br>• Physical location of the phone<br>• Cisco Unified Communications Manager user to associate with the phone<br>• Partition, calling search space, and location information, if used<br>• 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.<br>Username should match the user that was created in step 3.<br>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:<br>• Enable the IP Phone option.<br>• Select Other from the Phone Model drop down list.<br>• Specify the Username and Password fields (Figure 1).<br>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

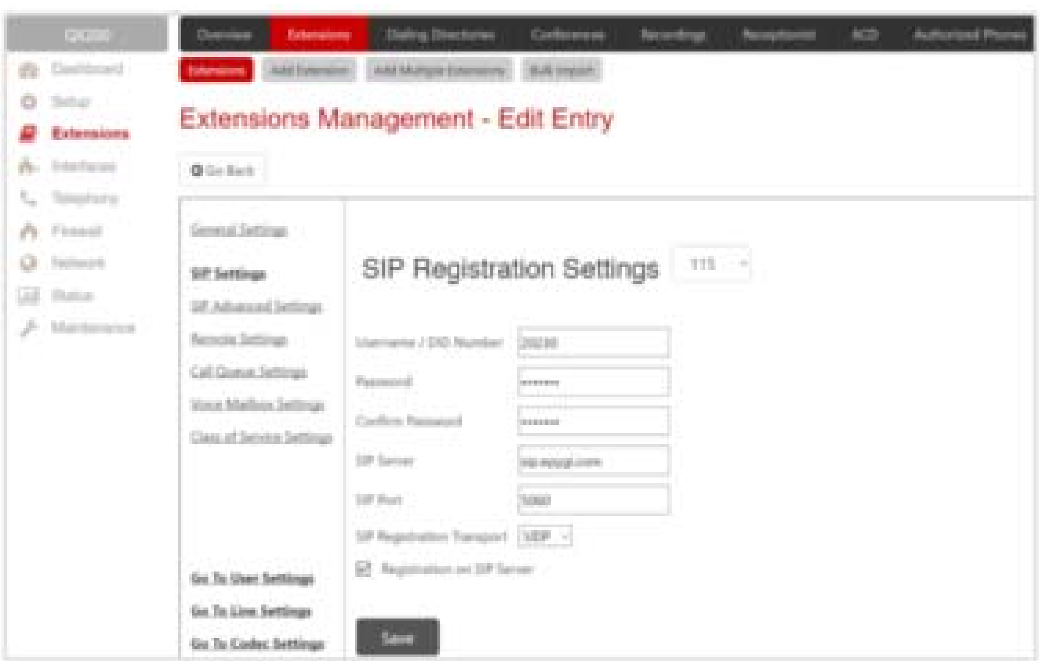



Figure 2: SIP Settings section

B. Configuring the Viking VoIP Phone

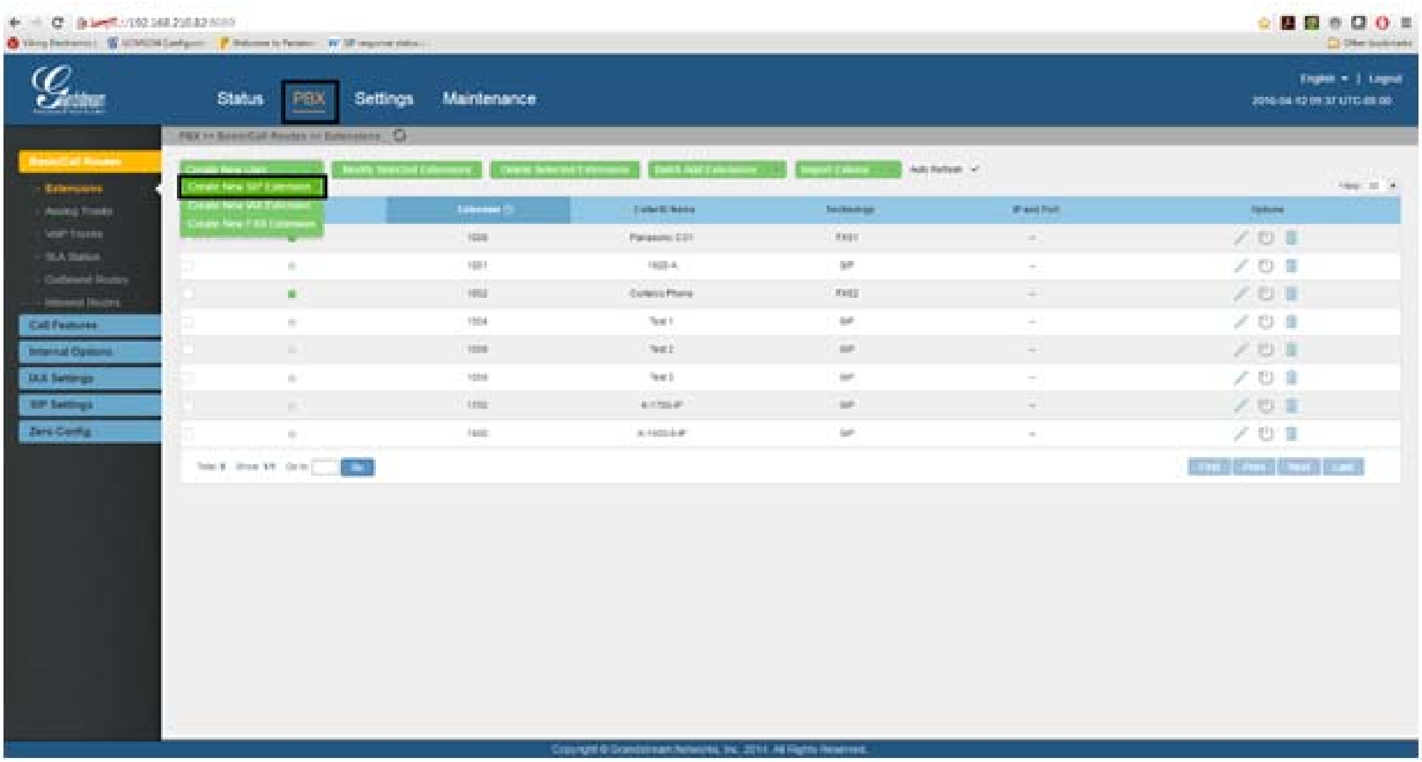
Power on the unit and connect it to the LAN interface for QX. The settings of the unit will be configured through Viking IP programming SW application installed on PC with MS windows. The following configuration steps for Viking should be done:

| | |
|---|---|
| **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:<br>•SIP Server: 172.30.0.1 (the default LAN IP address of QX)<br>•Username: locext115 (the same as configured on QX IP line settings)<br>•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.). |

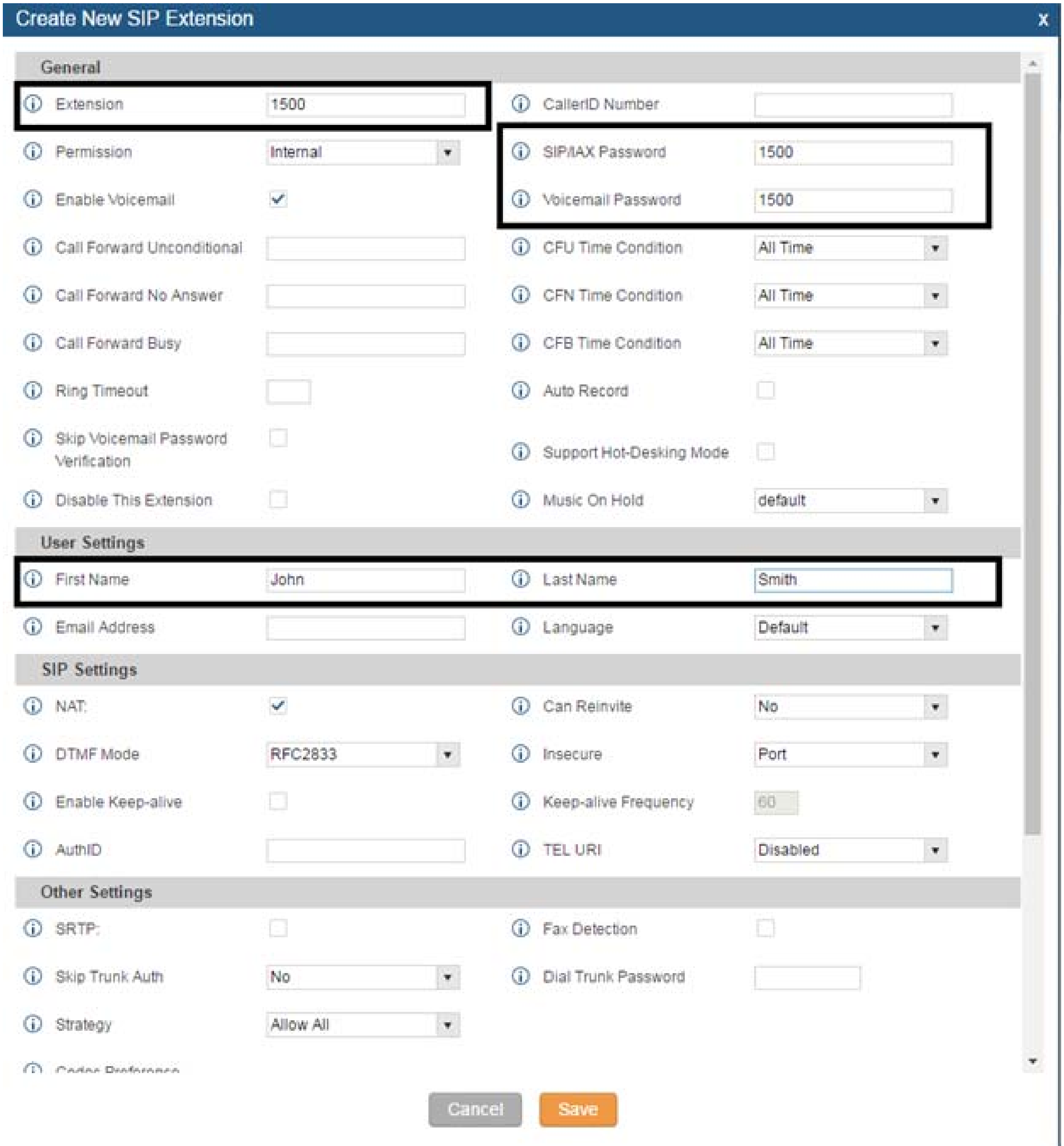
## Configuring Viking IP phones with Grandstream PBX devices:

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

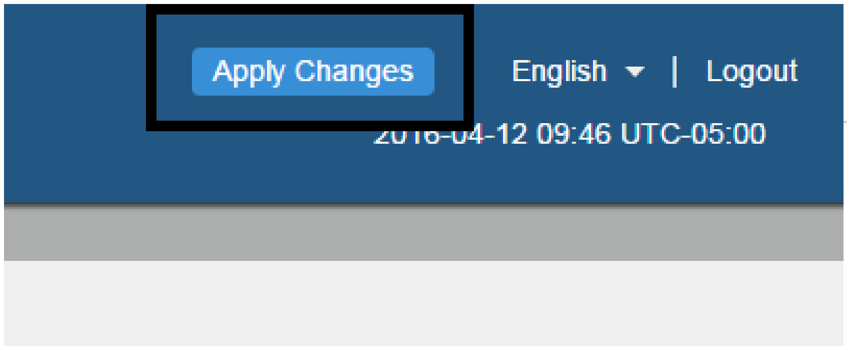

2. Click on “PBX” at the top, and then “Create New User” and “New SIP Extension”



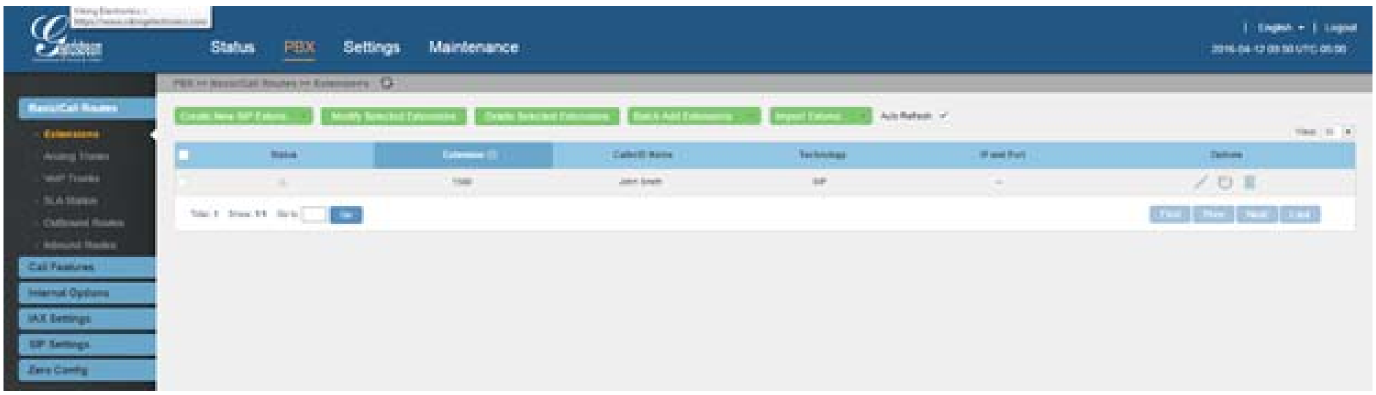
3. Put in your preference for “Extension” (which will be your username), “SIP/IAX Password” and “Voicemail Password”, along with “First Name” and “Last Name”. Click on “Save”.



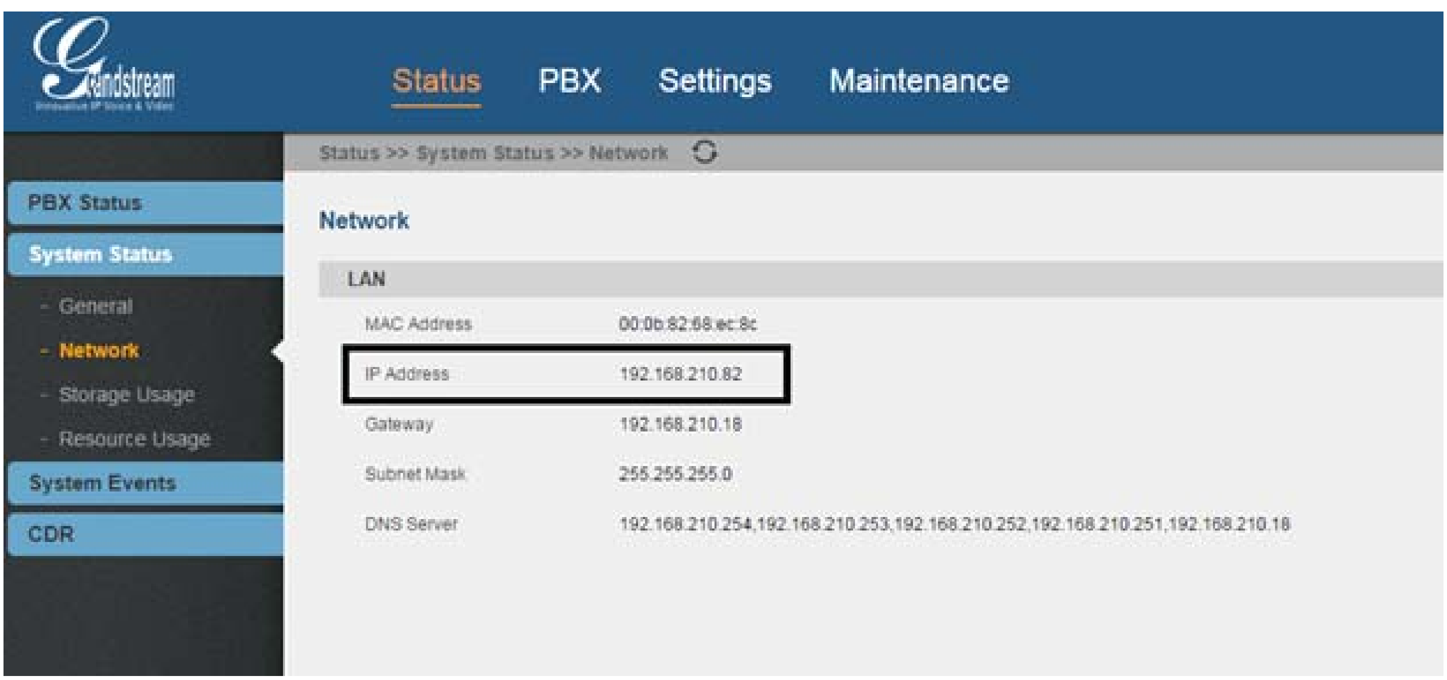
4. After saving, click on the “Apply Changes” button in the top right corner.



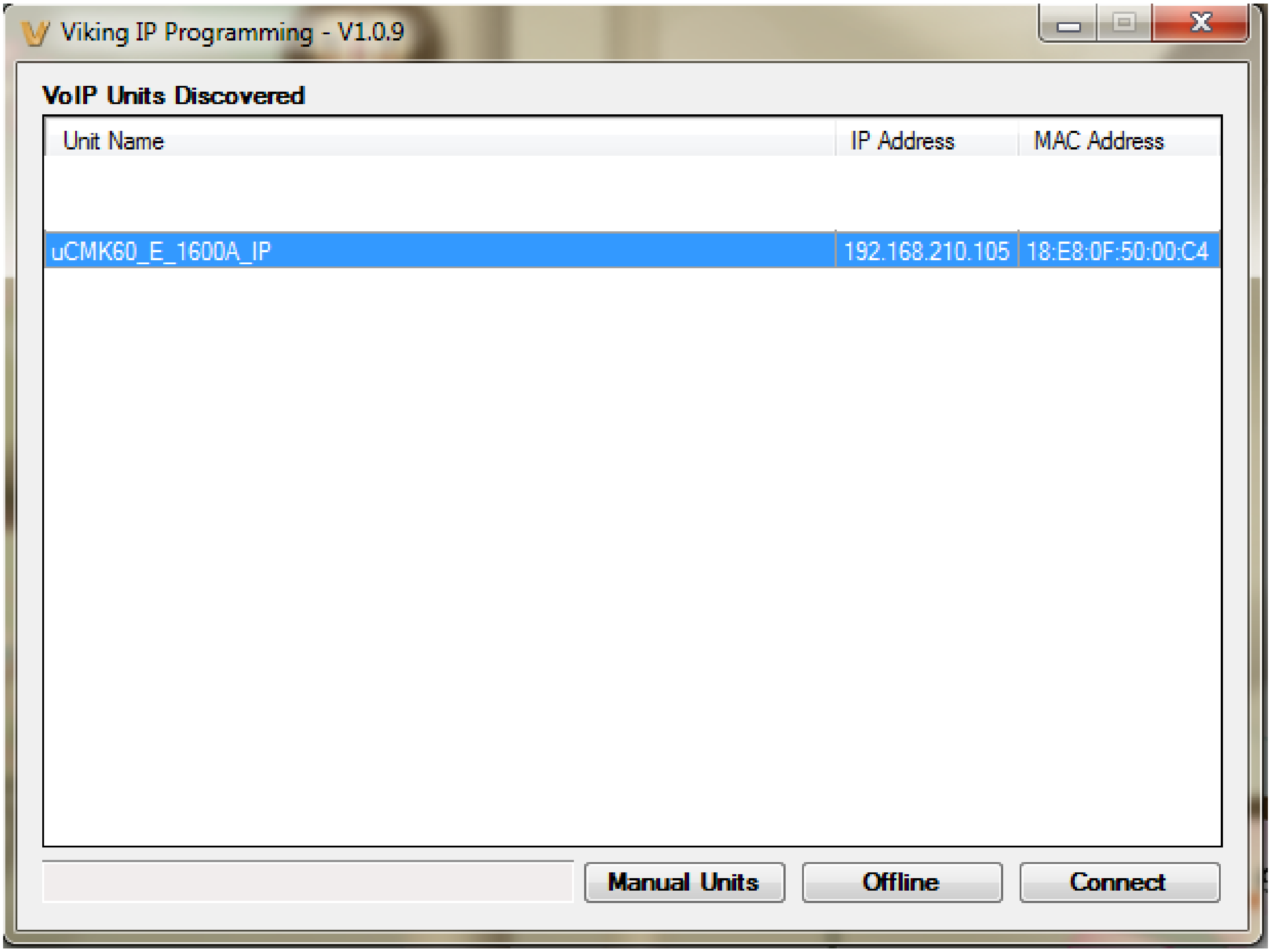

5. You will then see the extension you set up under the list of extensions.




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”.




### 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:<br>• MAC address<br>• Physical location of the Viking Voip Phone<br>• CCSU or LAN2 IP Address of the Iwatsu ECS System<br>• Extension Number for the phone or hunt group that will be called when the button is pressed<br>• 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.<br>Username should match the user that was created in step 3.<br>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. |

## 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:

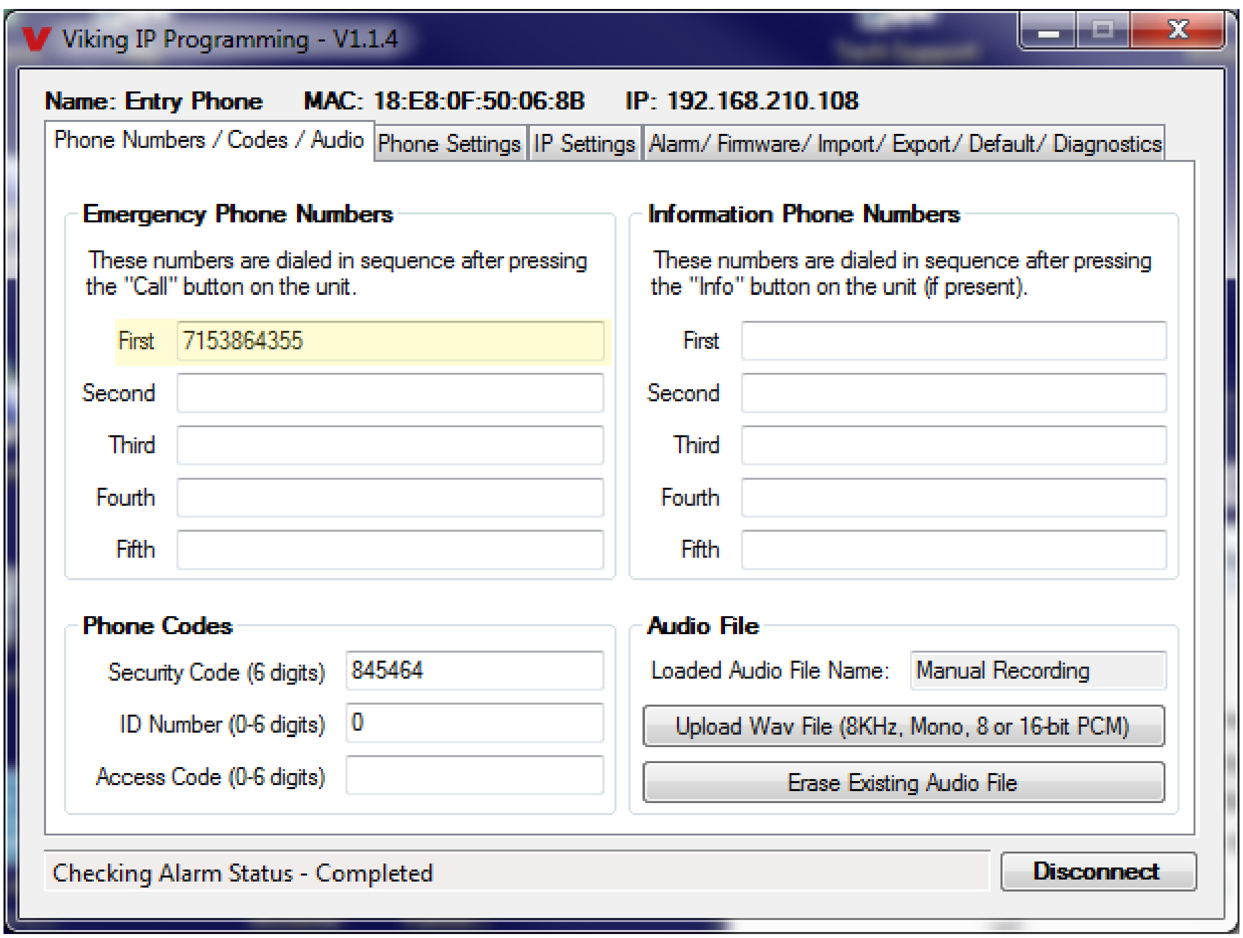
Go to “IP Telephony” and under “SIP”, “Advanced Authorization” and then “Global Adv. Parameters” uncheck “Authenticate Register”.
Our username = extension number of the wave phone.
Our password = SIP authentication password.

## VOIP.MS as the SIP provider for Viking IP Products

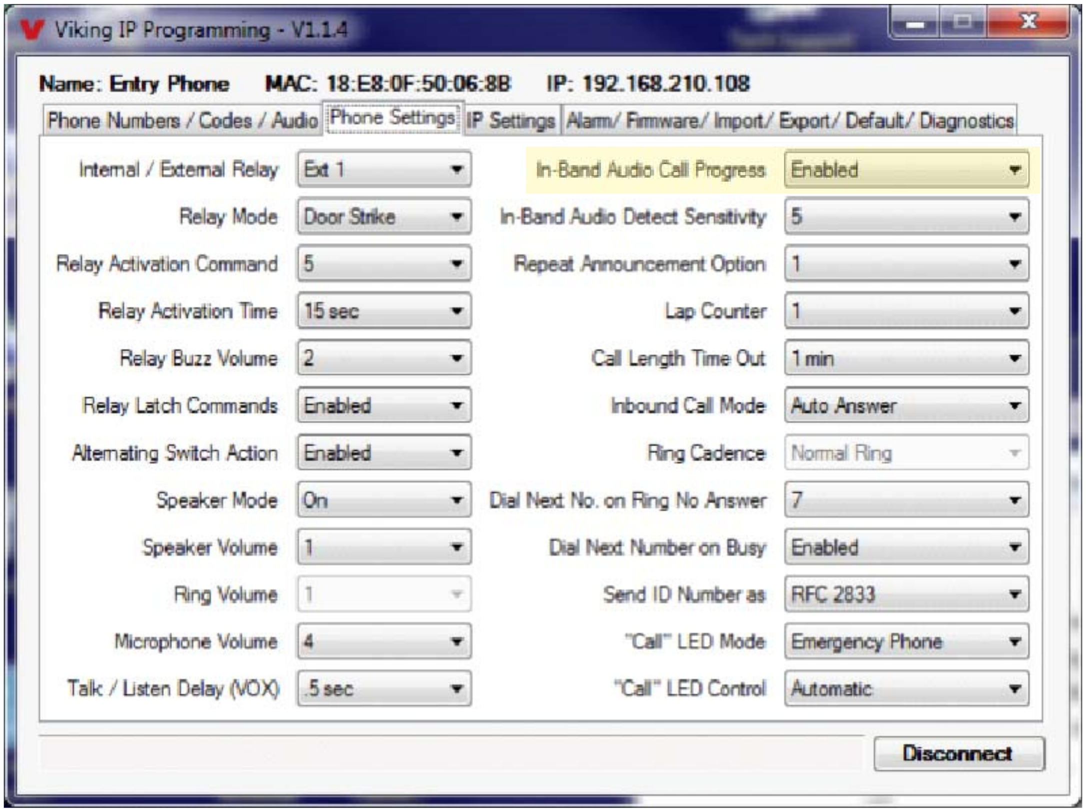
An account must be set up with VOIP.MS. Go to www.voip.ms to set up an account and upon completion, a SIP username and password will be assigned. Their account information page shows a long list of available VOIP servers. You can pick any of these servers to be used as the SIP server for the Viking IP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

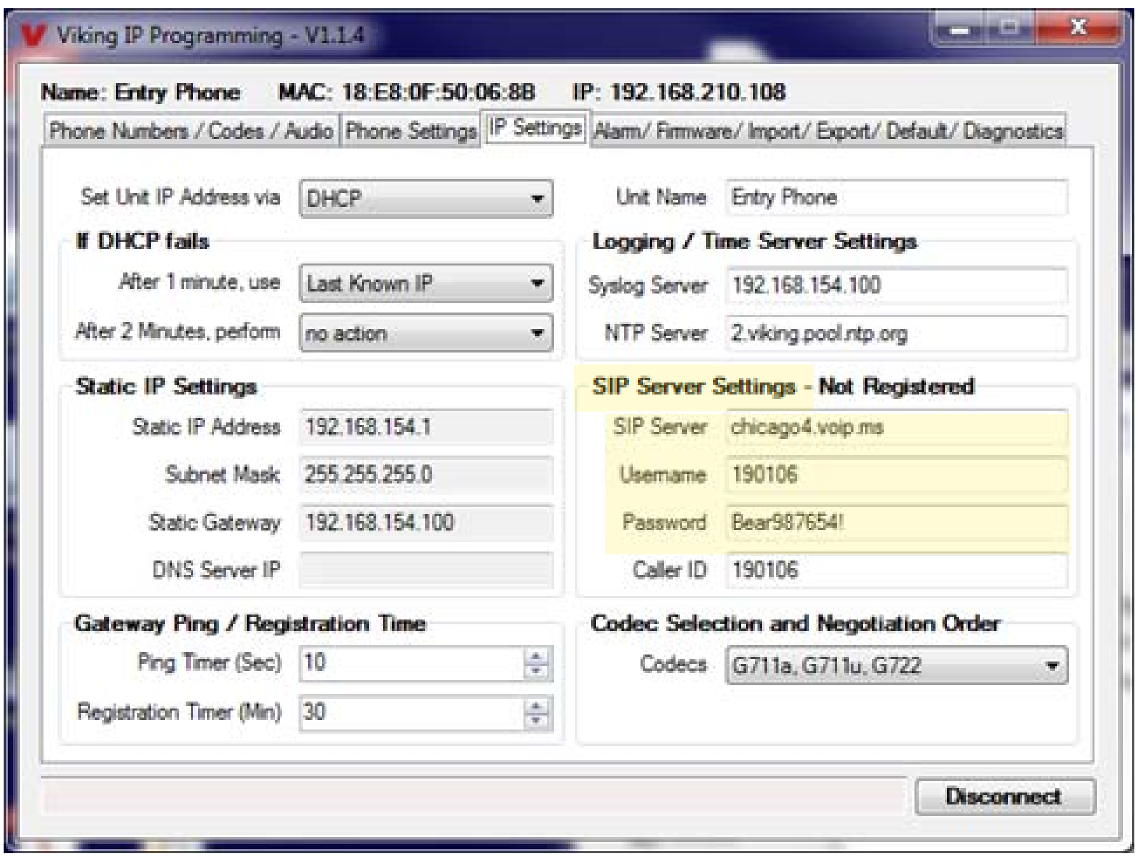
When the Viking IP has placed a call and the remote party hangs up, it takes approximately 22 seconds for the SIP server to pass the disconnect signal to the Viking IP phone.




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!”.