

Configuring Viking VoIP Phones and SIP Servers

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.

Configuration Steps	
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. Note: Viking VoIP Phones use only the SIP Profile Information section of the SIP Profile 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.

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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:

Configuration Steps	
Step 1	Gather the following information about the Viking 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.

Viking VoIP Phone Programming - v1.1.1

Phone Name	IP Address	MAC Address
Call_Box	192.168.1.128	18:E8:0F:50:00:AE

Disconnect **Refresh** **Security**

Phone Name
 Call_Box
 Read Name Set Name

Phone Codes
 Security Code 845464
 ID Number 987654
 Access Code 123456
 Read Codes Write Codes

Audio
 Upload Read Erase Debug
 -No Audio File-

Phone Firmware
 Update Read V2.6 ☒ Verify

IP Firmware
 Update R3.28.1448 Port 0

Phone Numbers
Speed Dial Numbers
 First 368
 Second
 Third
 Fourth
 Fifth

Information Phone Numbers
 First
 Second
 Third
 Fourth
 Fifth
 Read Numbers Write Numbers

Import / Export
 Import Data Export Data

Diagnostics
 Mic / Speaker Activate Relay

Phone Settings
 Talk/Listen Delay .5 sec
 Call Length Timeout 3 min
 Msg. 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
 "Call" LED Mode Emergency Phc
 Read Settings Write Settings

IP Settings
 IP Address via DHCP
If DHCP fails
 At 1 Min, use Static IP Address
 after 2 Mins, reboot
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
Logging/Time Server Settings
 Syslog Server 192.168.154.100
 NTP Server 10.2.2.10
SIP Server Settings
 IP Address 10.2.2.20
 Username 150
 Password 1234
 Caller ID 150
 Read IP Settings Write IP Settings

SIP / Network Failure Alarm
 Alarm Status: **Activated** ☒ Muted
 Enable Alarm Beeps
 Return to Factory Default Settings

Figure 1 - Iwatsu ECS Sample Configuration