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.

none Num	here / Codee / Au	dio Dharas Caniana UD Cania	N (E	
IONE NUM	Dels / Codes / Au	Phone Settings IP Setting	gs Alam/ Firmw	are/ import/ Export/ Derauit/ Diagnostics
Emerge	ncy Phone Nur	nbers	Information	Phone Numbers
These n the "Call	umbers are dialed "button on the ur	in sequence after pressing it.	These numb the "Info" bi	ers are dialed in sequence after pressing utton on the unit (if present).
First	17153864355		First	
Second			Second	
Third			Third	
Fourth			Fourth	
Fifth			Fifth	
Phone (Codes		Audio File	
Securi	ty Code (6 digits)	845464	Loaded Aud	io File Name: Manual Recording
ID No	umber (0-6 digits)	0	Upload W	av File (8KHz, Mono, 8 or 16-bit PCM)
Access	Code (0-6 digits)			Erase Existing Audio File

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.

Phone Numbers / Codes / Au	Idio Phone Settin	gs	P Settings Alarm/ Firmware/ Import/	Export/Default/Diagno	stics
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	Ŧ
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	

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

ne Numbers / Codes / A	Audio Phone Settings IP Set	tings VLAN Alarm	/ Firmware / Default / Tools
Set Unit IP Address via	DHCP	Unit Name	VIKING_MK64_Vik02
f DHCP fails		Logging / Ti	ime Server Settings
After 1 minute, use	Last Known IP	Syslog Server	192.168.154.100
After 2 Minutes, perform	no action	NTP Server	2.viking.pool.ntp.org
Static IP Settings		SIP Server	Peer to Peer Settings
Static IP Address	192.168.154.1	Server 👻	callcentric.com
Subnet Mask	255.255.255.0	Outbnd Proxy	callcentric.com
Static Gateway	192.168.154.100	Authentic. ID	
DNS Server IP		Usemame	17772685129
Gateway Ping / Regi	stration Time / Codecs	Password	Bear845464
Ping Timer (S) 10 🚔	Regist. Time (Min) 30 🛔	Caller ID	17772685129
Codecs Order G711u, G	G711a, G722 🔹	Register Fails	Re-Resolve 🔻

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).

	QX200	Overview	IP Lines	FXS	FXO	E1/T1 Trunk	ISDN Trunk	PSTN Gateways
	Dashboard	IP Lines IP	Line Settings	IP Phone	Templates	IP Phones Logo	FXS Gateways	
\$	Setup	ID Line	Sotting	e IE	Lino	12		
	Extensions	IF LINE	Setting	5 - IF	LINC	15		
h -	Interfaces	O Go Back						
C	Telephony	(
~	Firewall	IP Line 13	Ň					
0	Network	O Inactive						
dil	Status	0.12.01						
10	Maintenance	IP Phone	Phone Model	: Oth	ner		~	
			MAC Address	6	-			
			Line Appeara	nce: 2				
			Username:	loce	xt115			
			Password:	••••		Ger	nerate Password	I.
			Transport:	UD	Р ~			
			Use Template	: <	use defau	lt> 👻		
			Use Sessi	ion Timer				
			Symmetr	ic RTP				
		Save						



	QX200	Overview Extension	s Dialing Directories	Conferences	Recordings	Receptionist	ACD	Authorized Phones
23	Dashboard	Extensions Add Extensio	n Add Multiple Extensions	Bulk Import				
Ф	Setup	Extensions M	anagement F	dit Entry				
	Extensions	Extensions wi	anagement - L					
d-	Interfaces	G Go Back						
0	Telephony							
0	Firewall	General Settings						
0	Network	SIP Settings	SIP Registra	ation Setting	S 115	~		
[d]	Status	SIP Advanced Settings						
a.C	Maintenance	Remote Settings	Username / DID Number	20230				
		Call Queue Settings	Password					
		Voice Mailbox Settings	Confirm Password					
		Class of Service Settings	SIP Server	sip.epygi.com				
			SIP Port	5060				
			SIP Registration Transport	UDP ~				
		Go To User Settings	Registration on SIP Se	erver				
		Go To Line Settings	9					
		Go To Codec Settings	Save					

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"

Grandstream	Status <u>PBX</u> Set	ttings Maintenance				English + 2019-02-13 10.40 UTC	Logot -06.00
	PBX >> Basic/Call Routes >> Extensions	· O					
Basic/Call Routes	Create New User Modify	Selected Extensions Delete Sele	cled Extensions Batch Add Extonsions	Import Extensi	tefresh 🖌		
- Extensions	Create New SIP Extension					View	30 *
	Create New IAX Extension	Extension 💮	Callerif) Name	Technology	IP and Port	Options	
		1000	Panasonic Trunk-Port	FXS1	+	/0 1	
SLA Station	(a)	1002	Nick 1666	SIP	192.168 210 98 5000	/ 也 面	
 Outbound Routes Inbound Routes 	G (#	1006	Nick Test	SIP	5	/ 🙂 🖩	
Call-Features		1190	Geat Test	SP		/ 心 面	
Internal Options	0.00	1111	QA_FUNC TEST RM_DUT-50.31.9C	SIP	14	/ U 🛍	
IAX Settings		1112	GA_FUNC TEST RM_ DUT	SP		/ U 🖬	
SIP Settings	(D) (*	1113	QA_FUNC TEST RM_	SIP	÷	10 🖬	
Zero Config	- ·	1200	Phone 2	FXS2	÷	/ 🙂 🖩	
		1234	DA_FUNCTEST RM_X-LITE	BIP	÷	100	
	0 e	1600	Nick's Loud Ringer	SIP	÷	/ 10 曲	
		1700	Nick 1700	SIP	192 168 210 27 1060	/ U 🛍	
		2001	QA_Test Station 1_ Future Use	SIP	(10	/ 🙂 🛍	
	11 I.	2002	QA, Test Station 1, Zopper	SIP		/ 🕖 🖩	
		2003	QA_Test Station 1_ DUT	SIP		100	
		2004	QA_Test Station 1_ Future Use	SIP	<i>2</i>	10 1	
		2005	QA_Test Station 1_ Future Use	SIP	(B)	/ U 🗊	
						2 als -	

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".

(seneral						
D	Extension	2000		1	CallerID Number		
D	Permission	Internal		Ì	SIP/IAX Password	2000	
D	Enable Voicemail	~		Ð	Voicemail Password	2000	
D	Call Forward Unconditional			1	CFU Time Condition	All Time	¥
D	Call Forward No Answer			1	CFN Time Condition	All Time	•
D	Call Forward Busy			(i)	CFB Time Condition	All Time	•
Ð	Ring Timeout			1	Auto Record		
D	Skip Voicemail Password Verification			()	Support Hot-Desking Mode		
Ð	Disable This Extension			1	Music On Hold	default	٠
l	Jser Settings						
D	First Name	John		0	Last Name	Smith	
D	Email Address			1	Language	Default	×
	SIP Settings						
Ì	NAT:	~		(j)	Can Reinvite	No	•
D	DTMF Mode	RFC2833	•	(i)	Insecure	Port	•
D	Enable Keep-alive			(i)	Keep-alive Frequency	60	
D	AuthID			(j)	TEL URI	Disabled	٠
(Other Settings						
D	SRTP:			0	Fax Detection		
1	Skip Trunk Auth	No	•	1	Dial Trunk Password		
D	Strategy	Allow All					

9

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.

Getter	Status <u>PIIX</u> Setting	gs Mainlenance /				1. Strates - 1. Strates - Strates of the set of the set
	- Peters Associat Routes in Concession C					
Restalline	Costs New Soft News, 177 Marry Street	Attended Seaters	Income Real-Add Common	Transference Aske	which.	
· Externation					anore	. The C
many trent	and the second	100000	Cabell Store	Territoria	and that	
- weet franks	101 101		Jane lower.	14	8	101
- SLA Manage	fant berit ten					State Street States
Advantal Property						
Call Facilities 1						
Internal Options						
Waterings						
19 Lange						
for Conta						

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.

Gendstream	Status PBX Settings Maintenance	
1.103 17.100	Status >> System Status >> Network O	
PBX Status	Network	
System Status		
- General - Network	MAC Address 00.0b.82.68.ec.8c IP Address 192.168.210.82	
 Storage Usage Resource Usage 	Gateway 192.168.210.18	
System Events	Subnet Mask 255 255 255 0	
CDR	DNS Server 192.168.210.254,192.168.210.253,192.168.210.252	,192 168 210 251,192 168 210 18

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

Viking IP Programming - V1.2.0			
VoIP Units Discovered			
Unit Name		IP Address	MAC Address
VIKING_MK64_vik02		192.168.210.110	18:E8:0F:50:31:85
	Na	000	0
	Manual Units	Offline	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.

Night Ring Pag	ing Numbers /	Codes / A	udio Sp	eaker Se	ttings IP Settin	gs VLAN	Alarm / Firm	ware / In	nfo / Too	Is
					More Spe SIP Line O DTMF	eaker Setti ut Enab Regeneratio	rngs → Pre Pa n Disable	age Tone d	Enab	•
					C10 0		-			
Paging So	urces (Multic	ast IP ar	d Port)	- Volun	SIP Page SIP In Abo	ve 0	siP O	lulticast ut Above ble	0	•
Paging So	urces (Multic	vol	I T	- Volun	SIP Page SIP In Abo ne Level - Ti	ve 0	siP Or SIP Or ble/Disal	ulticast ut Above ble Vol 1	• TO	•
Paging So 0 1	urces (Multic	Vol	1 • 1 •	- Volun TO 🗹 TO 🔽 TO 🔽	SIP Page SIP In Abo ne Level - Ti 5 6 7	ve 0 meout Ena	sip ority vs M SIP O ble/Disal	Vol 1 Vol 1 Vol 1	0 • T0 • T0 • T0	• •
Paging So 0 1 2 3	urces (Multic	Vol	1 • 1 • 1 • 1 •	- Volun TO V TO V TO V TO V	SIP Page SIP In Abo the Level - Ti 5 6 7 8	e / Call Pri ve 0 meout Ena	siperity vs M siperity vs M ble/Disal 1 1 1 1 1 1 1 1 1 1 1 1 1	Uticast ut Above Vol 1 Vol 1 Vol 1 Vol 1	0 01 • 01 • 01 •	• •

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.

Night Ring Paging Numbers / Codes / Audio Speaker Settings IP Settings VLAN Alarm / Firmware / Info / Tools More Speaker Settings SIP Line Out Enat • Pre Page Tone Enat • DTMF Regeneration Disabled • Multicast Page Type Polycom • SIP Page / Call Priority vs Multicast SIP Out Above Poly • Polycom Poly SIP Page / Call Priority vs Multicast SIP In Above Poly SIP Out Above Poly Polycom Paging Source and Channel Subscriptions Image: Stead of the stable • Volume Image: Stead of the stable IP Address 224.0.1.116 Port 5001 Volume Image: Stead of the stable IP Address 224.0.1.116 Port 5001 Volume Image: Stead of the stable IP Address 224.0.1.116 Port 5001 Volume Image: Stead of the stable IP Address 224.0.1.116 Port 5001 Volume Image: Stead of the stable Volume Image: Stead of the stable IP Address 224.0.1.116 Port 5001	C4411	G_MK64_VIK	UZ MAC:	18:E8:0F:50:2	F:88 IP:	192.168.4	210.30	
More Speaker Settings SIP Line Out Enalt • Pre Page Tone Enalt • DTMF Regeneration Disabled • Multicast Page Type Polycom • SIP Page / Call Priority vs Multicast SIP In Above Poly • Polycom Page / Call Priority vs Multicast SIP In Above Poly • Polycom Polycom IP Address 224.0.1.116 Port 5001 Volume 1 • I'' Timeout Select which Polycom channels to subscribe to, channel 25 (Emergency) is always enabled V Ch 01 V Ch 03 V Ch 05 V Ch 07 V Ch 08 V Ch 09 V Ch 10 V Ch 12 V<	Night Ring Pa	ging Numbers	/ Codes / Audio	Speaker Settin	ngs IP Setting	S VLAN A	larm / Firmware	/ Info / Tools
SIP Line Out Enab ▼ Pre Page Tone Enab ▼ DTMF Regeneration Disabled ▼ Multicast Page Type Polycom ▼ SIP Page / Call Priority vs Multicast SIP In Above Poly ▼ SIP Out Above Poly ▼ Polycom Paging Source and Channel Subscriptions SIP In Above Poly ▼ SIP Out Above Poly ▼ IP Address 224.0.1.116 Port 5001 ↓ Volume 1 ♥ Timeout Select which Polycom channels to subscribe to, channel 25 (Emergency) is always enabled ♥ Ch 01 ♥ Ch 03 ♥ Ch 05 ♥ Ch 07 ♥ Ch 08 ♥ Ch 09 ♥ Ch 10 ♥ Ch 12 ♥ Ch 14 ♥ Ch 15 ♥ Ch 16 ♥ Ch 17 ♥ Ch 18 ♥ Ch 20 ♥ Ch 21 ♥ Ch 23 ♥ Ch 24					More Spea	ker Settin	gs	
DTMF Regeneration Disabled Multicast Page Type Polycom SIP Page / Call Priority vs Multicast SIP In Above Poly Polycom Paging Source and Channel Subscriptions IP Address 224.0.1.116 Pot 5001 Volume ■ Volume ■ Volume ■ V Ch 01 V Ch 02 V Ch 03 V Ch 04 V Ch 05 V Ch 06 V Ch 09 V Ch 10 V Ch 11 V Ch 12 V Ch 13 V Ch 14 V Ch 17 V Ch 18					SIP Line Out	Enab •	Pre Page To	ne Enab 🔻
Multicast Page Type Polycom SIP Page / Call Priority vs Multicast SIP Page / Call Priority vs Multicast SIP In Above Poly v SIP Out Above Poly v Polycom Paging Source and Channel Subscriptions IP Address 224.0.1.116 Port 5001 ÷ Volume 1 v Vimeout Select which Polycom channels to subscribe to, channel 25 (Emergency) is always enabled Volume 1 v Volume 1 v Volume 1 v V Ch 01 V Ch 02 V Ch 03 V Ch 04 V Ch 05 V Ch 06 V Ch 07 V Ch 08 V Ch 09 V Ch 10 V Ch 11 V Ch 12 V Ch 13 V Ch 14 V Ch 15 V Ch 16 V Ch 17 V Ch 18 V Ch 19 V Ch 20 V Ch 21 V Ch 22 V Ch 23 V Ch 24					DTMF R	generation	Disabled	•
SIP Page / Call Priority vs Multicast SIP In Above Poly SIP Out Above Poly Polycom Paging Source and Channel Subscriptons IP Address 224.0.1.116 Port 5001 Volume 1 Imeout Select which Polycom channels to subscribe to, channel 25 (Emergency) is always enabled IV Ch 01 IV Ch 02 IV Ch 03 IV Ch 05 IV Ch 06 IV Ch 07 IV Ch 08 IV Ch 09 IV Ch 10 IV Ch 11 IV Ch 12 IV Ch 13 IV Ch 14 IV Ch 15 IV Ch 16 IV Ch 17 IV Ch 18 IV Ch 19 IV Ch 20 IV Ch 21 IV Ch 22 IV Ch 23 IV Ch 24					Multicast	Page Type	Polycom	•
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Polycom Paging Source and Channel Subscriptions IP Address 224.0.1.116 Port 5001 ÷ Volume 1 • ✓ Timeout Select which Polycom channels to subscribe to, channel 25 (Emergency) is always enabled Image: Ch 01 Image: Ch 02 Image: Ch 03 Image: Ch 04 Image: Ch 05 Image: Ch 06 Image: Ch 07 Image: Ch 08 Image: Ch 09 Image: Ch 10 Image: Ch 11 Image: Ch 12 Image: Ch 13 Image: Ch 14 Image: Ch 15 Image: Ch 16 Image: Ch 17 Image: Ch 18 Image: Ch 19 Image: Ch 20 Image: Ch 21 Image: Ch 23 Image: Ch 24					CID In Aller			
IP Address 224.0.1.116 Port 5001 ⇒ Volume 1 ✓ Timeout Select which Polycom channels to subscribe to, channel 25 (Emergency) is always enabled ✓ Ch 01 ✓ Ch 02 ✓ Ch 03 ✓ Ch 04 ✓ Ch 05 ✓ Ch 06 ✓ Ch 07 ✓ Ch 08 ✓ Ch 09 ✓ Ch 10 ✓ Ch 11 ✓ Ch 12 ✓ Ch 13 ✓ Ch 14 ✓ Ch 15 ✓ Ch 16 ✓ Ch 17 ✓ Ch 18 ✓ Ch 19 ✓ Ch 20 ✓ Ch 21 ✓ Ch 22 ✓ Ch 23 ✓ Ch 24					SIP IN ADOVE	Poly -	SIP OUT ADO	ve Poly -
Select which Polycom channels to subscribe to, channel 25 (Emergency) is always enabled Image: Ch 01 Image: Ch 02 Image: Ch 03 Image: Ch 04 Image: Ch 05 Image: Ch 06 Image: Ch 07 Image: Ch 08 Image: Ch 09 Image: Ch 10 Image: Ch 11 Image: Ch 12 Image: Ch 13 Image: Ch 14 Image: Ch 15 Image: Ch 16 Image: Ch 17 Image: Ch 18 Image: Ch 19 Image: Ch 20 Image: Ch 21 Image: Ch 23 Image: Ch 24	Polycom I	Daging Source	e and Chann	el Subscriptio	ns	Poly •	SIP OUT ADO	ve Poly 🔻
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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".



In addition to the devices RingCentral sells pre-provisioned, RingCentral supports assisted provisioning for additional models. If your model is not available via assisted provisioning, RingCentral may have documented how to manually configure it. Please see the <u>office devices</u> page for more information.



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.

	Setup & Provisioning
	✓ Select Device ✓ Provisioning 3 Finish
To configure your device to co	onnect to the RingCentral service, you will need to program it with the following information.
The steps for programming w	ill vary from device to device, so please check with your device's manufacturer for specific instructions.
The steps for programming w	ill vary from device to device, so please check with your device's manufacturer for specific instructions.
The steps for programming w Field SIP Domain	ill vary from device to device, so please check with your device's manufacturer for specific instructions. Value sip.ringcentral.com:5060
The steps for programming w Field SIP Domain Outbound Proxy	Value sip.ringcentral.com:5060 Please select outbound proxy according to the location of your device v
The steps for programming w Field SIP Domain Outbound Proxy User Name	Value sip.ringcentral.com:5060 Please select outbound proxy according to the location of your device ~ 17159824355
The steps for programming w Field SIP Domain Outbound Proxy User Name Password	Value sip.ringcentral.com:5060 Please select outbound proxy according to the location of your device v 17159824355 5jTWqLG

Done

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.

ala Dina Dinatan Musikas	- (C-d (A-d- C) C	Hinne IP Settinge	WAN Alexe / Deverse / Infa / Task
	s / Codes / Audio Speaker S	ettings in Settings	VLAN Alarm / Firmware / Into / Tools
Set Unit IP Address via	DHCP -	Unit Name	Tech Support Test 40TB-IP
If DHCP fails		Logging / Ti	me Server Settings
After 1 minute, use	Last Known IP 👻	Syslog Server	192.168.154.100
After 2 Minutes, perform	no action 💌	NTP Server	2.viking.pool.ntp.org
Static IP Settings		SIP Server	Peer to Peer Settings
Static IP Address	192.168.154.1	Server 💌	sip.ringcentral.com
Subnet Mask	255.255.255.0	Outbnd Proxy	sip10.ringcentral.com:5090
Static Gateway	192.168.154.100	Authentic. ID	15992243020
DNS Server IP		Usemame	17159824355
Gateway Ping / Regi	stration Time / Codecs	Password	5jTWqLG
Ping Timer (S) 10 🚔	Regist. Time (Min) 30 🚔	Caller ID	17159824355
Codecs Order G711u.	G711a, G722 🔹 🔻	Register Fails	No Action 👻

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

light Ring Paging Numbers	/ Codes / Audio Sp	eaker Settings IP Settings VLAN Ala	m / Fimware / Info / Tools
Internal / External Relay	Internal	In-Band Audio Call Progress	Enabled
Relay Mode	Door Strike	In-Band Audio Detect Sensitivity	5
Relay Activation Command	** ,	Repeat Announcement Option	[1
Relay Activation Time	5 sec	Lap Counter	Disabled
Relay Buzz Volume	3.	SIP Page / Call Length Timeout	3 min
Relay Latch Commands	Enabled	Multicast Page Length Time Out	180 sec •
Panic Button Mode	Disabled	Inbound Call Mode	Auto Answer
Speaker Mode	On	Ring Cadence	Normal Ring
SIP Page / Phone Volume	1	Dial Next No. on Ring No Answer	7
Ring Volume	5	Dial Next Number on Busy	Enabled
Microphone Volume	5	LED Mode	Phone / Paging
Talk / Listen Delay (VOX)	.5 sec	Call" LED Control	Automatic

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.

ight Ring	Paging Numbers	/ Codes / Audio Speaker Se	ttings IP Settings \	/LAN Alam	m / Firmware / Info / Too
Emerge	ncy Phone Nur	nbers	Non-Emergen	cy Phone	Numbers
These n the "Call	umbers are dialed "button for over 3	in sequence after holding seconds.	These numbers the "Call" butto	are dialed n momentar	in sequence after pressin rily.
First	7153864355		First		
Second			Second		
Third			Third		
Fourth			Fourth		
Fifth			Fifth		
Phone (Codes		Audio File		
Securi	ty Code (6 digits)	845464	Loaded Audio F	ile Name	Clean-up (1)
ID N	umber (0-6 digits)		Upload Wav	File (8KHz,	Mono, 8 or 16-bit PCM)
Access	Code (0-6 digits)	123456	Erase Uploade	d Audio	Play Uploaded Audio

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:

ght Ring Paging Number	s / Codes / Audio Speaker Set	tings IP Settings VLAN Ala	arm / Firmware / Info / Tools
Dynamic IP Settings		Firmware Versions	
Unit IP Address	192.168.210.28	IP R6.35.1834	Update IP
Subnet Mask	255.255.255.0	Unit V2.9	Update Unit
Gateway IP Address	192.168.210.18	Import / Export	
DNS Server IP Address	192.168.210.253	Import Data	Export Data
SIP / Network Failure	e Alarm	Default Unit Settings	
SIP Registration Status	Registered	Clear Speaker Settings	Clear IP Settings
Current Alarm Status	Idle	Diagnostics	
Alarm Muted	Mute Current / Next Alarm	Test Relay Activation	Read Relay Status
Permanent Alarm Mute	Alarm Tones Disabled 🔹	Last Read Relay Statu:	5
Programming Userna	me and Password	Test Mic / Speaker	Result
Programming Set	Usemame and Password		

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.

Phone Num	hers / Codes / Au	dio Dhone Cattings ID Cattin		muuro / Impart / Empat / Default / Disconsting
THORE Hum		rnone settings ir settin	gs (Avami/ Fin	mware/ import/ Export/ Default/ Diagnostics
Emerge	ncy Phone Num	bers	Informati	ion Phone Numbers
These not the "Call	umbers are dialed " button on the un	in sequence after pressing it.	These nu the "Info"	mbers are dialed in sequence after pressing 'button on the unit (if present).
First	7153864355		First	
Second			Second	
Third	Č.		Third	
Fourth			Fourth	
Fifth]	Fifth	
Phone (Codes		Audio Fil	le
Security Code (6 digits) 845464 ID Number (0-6 digits) 0		Loaded A	udio File Name: Manual Recording	
		Upload Wav File (8KHz, Mono, 8 or 16-bit PCM)		
Access Code (0-6 digits)		Erase Existing Audio File		

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

Phone Numbers / Codes / Au	idio Phone Setting	gs 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	Ŧ	Send ID Number as	RFC 2833 🔹
Microphone Volume	4	•	"Call" LED Mode	Emergency Phone •
Talk / Listen Delay (VOX)	.5 sec	•	"Call" LED Control	Automatic 🔹

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

Name: Entry Phone M	AC: 18:E8:0F:50:06:8	B IP: 192.1	58.2	10.108	
Phone Numbers / Codes / /	Audio Phone Settings IP S	ettings Alarm/ Fi	mwa	re/Import/Export/Default/D	agnostics
Set Unit IP Address via	DHCP	• Unit N	ame	Entry Phone	
If DHCP fails		Logging	/ 1	ime Server Settings	
After 1 minute, use	Last Known IP	Syslog Se	rver	192.168.154.100	
After 2 Minutes, perform	no action	NTP Se	rver	2 viking pool ntp.org	1
Static IP Settings		SIP Server	r / Po	eer to Peer Settings	
Static IP Address	192.168.154.1	Server		nicago4.voip.ms	
Subnet Mask	255.255.255.0	Authentic II	y >		
Static Gateway	192.168.154.100	Usemame	e 19	90106	
DNS Server IP		Passwor	d Be	ear987654!	
Gateway Ping / Regi	stration Time	Caller I	K-	-1900-8-IP Lobby	55
Ping Timer (Sec)	10	Register Fail	s R	le-Resolve 🔻	
Registration Timer (Min)	30	B			
				[

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:

Edit trunk		X
Pricelist	\$	
Title	Door Phone	
Trunk name	9999	
Auth Login	9999	
From user		
From domain		
Address or Host Name	dynamic 🔻 :	
Password	•••••	
Dialplan	Doorphone (Doorphone's dia 🗘	
Tone Zone	United States / North Americ 🗘	
Country Code	USA 🗘	
Keep-Alive		
Enable registration		

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:

Transport	UDP			\$
DTMF mode	rfc2833 \$	Payload	101	٦

A dial plan must be created for the SIP trunk:

Edi	t Doorpho	ne
Des	scription:	Doorphone's dialplan
	*504	
	1 Call g	roup Service

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.

hone Numbers / Codes / A	Audio Phone Settings IP Settin	gs VLAN Alarm /	Firmware / Default / Tools
Set Unit IP Address via	Static IP Settings 🔹	Unit Name	VIKING_MK64_Vik02
If DHCP fails		Logging / Ti	me Server Settings
After 1 minute, use	Static IP Address	Syslog Server	192.168.154.100
After 2 Minutes, perform	no action 🔹	NTP Server	2.viking.pool.ntp.org
Static IP Settings		SIP Server /	Peer to Peer Settings
Static IP Address	192.168.10.199	Server •	192.168.10.200
Subnet Mask	255.255.255.0	Outbnd Proxy	192.168.10.200
Static Gateway	192.168.10.1	Authentic. ID	9999
DNS Server IP		Usemame	9999
Gateway Ping / Regi	stration Time / Codecs	Password	Wil01did
Ping Timer (S) 10 🚔	Regist. Time (Min) 30 🚔	Caller ID	9999
Codecs Order G711u, G	G711a, G722 🔹	Register Fails	Re-Resolve

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

Phone Numbers / Codes / Au	idio Phone Settings	IP Settings VLAN Alarm / Firmware	/ Default / Tools
Internal / External Relay	Internal	In-Band Audio Call Progress	Enabled -
Relay Mode	Door Strike	In-Band Audio Detect Sensitivity	5 🗸
Relay Activation Command	00	Repeat Announcement Option	Continuous 👻
Relay Activation Time	5 sec	Lap Counter	Disabled -
Relay Buzz Volume	1	Call Length Timeout	2 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	4	Dial Next Number on Busy	Enabled -
Ring Volume	5	Send ID Number as	RFC 2833 -
Microphone Volume	5	Call" LED Mode	Entry Phone
Talk / Listen Delay (VOX)	.5 sec	"Call" LED Control	Automatic 🔹

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:

Phone Numbers / Codes / A	Audio Phone Settings IP Setting	s VLAN Alarm / Firmware	e / Default / Tools
Dynamic IP Settings		Firmware Versions	
Unit IP Address	192.168.210.30	IP R6.33.1826	Update IP
Subnet Mask	255.255.255.0	Unit V4.0	Update Unit
Gateway IP Address	192.168.210.18	Import / Export	
These are the current / la	ast known good IP addresses	Import Data	Export Data
SIP / Network Failur	e Alarm	Default Unit Setting	S
SIP Registration Status	Registered	Clear Phone Settings	Clear IP Settings
Current Alarm Status	Idle	Diagnostics	
Aarm Muted	Mute Current / Next Alarm	Test Relay Activation	Read Relay Status
Permanent Alarm Mute	Alarm Tones Enabled 👻	Last Read Relay Statu	JS
Alarm can be triggered I SIP server, Gateway ping (if unit forced to	by unit failing to register with failure, or IP address conflict use static settings).	Test Mic / Speaker	Result