



## Sangoma Vega 5000 24FXS/2FXO

## and Elastix Server

5000

Setup Guide



## 1.0 Setup Diagram

Figure 1-1 is a setup diagram for a single Vega 5000 analog gateway configuration. We're going to configure a SIP Trunk for communication between the IP Phone and PSTN.



Figure 1-1. Setup Diagram

## 2.0 Host PC Environment

Table 2-1.	<b>Host Server</b>	Environment	Details
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	Description
Hardware Type	Elastix Appliance ELX-Series
Hardware Version	ELX-3000
Software Type	Elastix
Software Version	2.3

## 3.0 Test Setup Equipment

Table 3-1. Test Setup Equipmen
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Equipment	Model	Version
IP (SIP) Phone	N/A	N/A
Sangoma	Vega 5000	FW: R088S020
Switch	N/A	N/A

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## 4.0 Setup Procedure

#### To set up the Elastix Server for the Vega 5000

- 1. Go to the web address of the Elastix Server Login page. The web address is determined by the customer, for this guide we have used the IP address 192.168.1.75
- 2. On the Login page, type the username and password for an administrative user into the Username and Password fields, see Figure 4-1. The username and password are determined by the customer.



#### Figure 4-1. Login

- 3. Press Enter or click on the Submit button to go to Elastix's Dashboard
- 4. Once inside, click on the PBX tab on the menu at the top of the screen

#### Figure 4-2. Dashboard

Əelastix 🛛	System Agenda	Email Fax	PBX	IM	Reports V	61911	14
Dashboard Network	Users S	hutdown Hardware D	etector Updates	Backu	p/Restore Preferences		
Dashboard	🙆 Dashboard					ि भा 🖌	r ?
Dashboard Applet Admin	System Resources		5	Proces	sses Status		5
History					Telephony Service	RUNNING	V
Jashboard					Instant Messaging Service	NOT RUNNING	T
Language	1.4%	13.1%	0.0%		Fax Service	RUNNING	-
Themes	CPU	RAM	SWAP		Email Service	RUNNING	-
BX Configuration					Database Service	RUNNING	-
lardware Detector	CPU Info: Uptime:	Intel(R) Atom(TM) 18 days, 19 hour(s	CPU D525 @ 1.80GHz i), 23 minute(s)		Web Server	RUNNING	-
	CPU Speed: Memory usage:	1,800.21 MHz RAM: 2,009.84 Mb	SWAP: 4,094.68 Mb	2	Elastix Call Center Service	NOT INSTALLED	-
	Hard Drives		S	Perfor	mance Graphic		S
		🛑 0% Used 🔳 100% .	Available		Simultaneous calls, memor	ry and CPU	
		Hard Disk Capacity: 4 Mount Point: / Manufacturer: ST9500	47.11GB 0423AS	3.	.0 .0 .0	1.010 1.010 1.005 1.000 200	
	Logs: 93M	Local Back	ups: 102M	0		0.990 J 150	
	Emails: 8.0K	Configural	tion: 76M		091 1710 1710 1710 1710 1710 1710 2011 2011	11:0	
(47) 35 3 6 (- 1 1 2	Volcemails: 104K	Recording	s: 8.0K		Sim. calls — CPU usage (%)		



5. We need to create a SIP Trunk to register the Vega with Elastix. Go to "PBX Configuration => Trunks => Add SIP Trunk", see Figure 4-3. This will take you to configure a SIP Trunk.

elastix*	System Agenda Email Fax PBX IM Reports V	69141
PBX Configuration Operator Pan	el Voicemail Monitoring Endpoint Configurator Conference Batch of Extensions Tools	Flash Operator Panel 🛛 VoIf 🕢 📐
PBX Configuration		☑ ⊩ ★ ?
Basic	Add a Trupk	Add Trunk
Extensions	Add a Hullk	Chappel (0) (zap)
General Settings	Add SIP Trunk	
Outbound Routes		
Trunks		
Inbound Call Control	Add Zap Trunk (DAHDI compatibility mode)	
Inbound Routes		
Zap Channel DIDs	Add IAX2 Trunk	
Announcements	Add ENUM Trunk	
Blacklist		
CallerID Lookup Sources	S Add DUNDi Trunk	
Day/Night Control	Add Custom Trunk	
Follow Me	Add cdstorn mark	

Figure 4-3. Add a SIP Trunk

6. On the "Add SIP Trunk" page (Figure 4-4), fill in the following information:

General Settings

• Trunk Name: (VegaTrunk in this example)

**Outgoing Settings** 

- Trunk Name: (Vega5000 in this example)
- Peer Details:

host=dynamic username=(Vega5000 in this example) secret=(jx8FkOU13sv6 in this example) qualify=yes type=peer insecure=very



Add SIP Trun	k	
General Settings		
Trunk Name:	VegaTrunk	



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- 7. Click on the 'Submit' button at the end of the page. The SIP Trunk will be created and you will see the page on Figure 4-5 displaying the "Apply Configuration Changes Here" pink ribbon on top of the screen.
- 8. Click on the "Apply Configuration Changes Here" link

#### Figure 4-5. Apply Configuration Changes Here

Gelastix							69142
PREEDOM TO COMUNICATE	System Ager	nda Email	Fax	PBX	IM Reports		
PBX Configuration Operator Pan	el Voicemail	Monitoring	Endpoint Configurator	Conference	Batch of Extensions	Tools	Flash Operator Panel 🛛 VoIP Pro🕢 🕨
PBX Configuration							☑ ⊩ ★ ?
			Apply Configurat	ion Changes He	ere		
Basic							
Extensions	Add a Trunk						Add Trunk
Feature Codes	@						VegaTrunk (sip)
General Settings	Add SIP Trunk						
Outbound Routes	O Add DAHDI Tru	unk					
Trunks							

9. With this you have finished creating a SIP Trunk that will be used by the Vega 5000 to register with the Elastix Server. Now, go to "PBX => PBX Configuration => Outbound Routes" to configure the outbound route to the Vega 50 Gateway. Fill in the following information: (Figure 4-6)

Route Settings

• **Route Name:** ("8\_Vega" in this example)

Dial patterns

• **Prefix:** ("8" in this example) | **Match pattern:** ("." in this example) *Trunk Sequence for Matched Routes* 

• **0:** ("VegaTrunk" in this example)

Figure 4-6. Add Route				
Add Route				
Route Settings				
Route Name:	8_Vega			

(prepend	) + 8			/ CallerId	) 🗃
+ Add Mor	e Dial Patter	n Fields			
Dial patter	ns wizards	s: (pick one)	-		
runk Sequ	ence for M	atched Routes			
Frunk Sequ	ence for M	atched Routes			
VegaTru	unk	atched Routes			,
Trunk Sequ	unk	atched Routes			)

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10. Click on "Submit" at the end of the page and Apply changes. Now, we'll create an extension for an IP Phone. Go to "PBX => PBX Configuration => Extensions" and click on "Submit" having the "Generic SIP Device" option selected. (Figure 4-7)

Figure 4-7. Add SIP Extension

<b>O</b> elastix	System Agenda Email Fax PBX IM Reports V	6 Q i # 2
PBX Configuration Operator Pan	el Voicemail Monitoring Endpoint Configurator Conference Batch of Extensions Tools	Flash Operator Panel 🛛 VoIf 🕢 🕨
PBX Configuration		☞ 빠 ★ ?
Basic		
Extensions	Add an Extension	Add Extension
Feature Codes		
General Settings	Please select your Device below then click Submit	
Outbound Routes		
Trunks	Device	
Inbound Call Control		
Inbound Routes	Device Generic SIP Device	
Zap Channel DIDs		
Announcements	Submit	
Blacklist		

11. Fill in the following information on the Add SIP Extension page (Figure 4-8):

- User Extension (302 in this example)
- **Display Name** ('IPPhone' in this example)
- **Secret** ('h7Dka3Rf9si0t' in this example)

Add SIP Exte	ension	
Add Extension		
User Extension	302	
Display Name	IPPhone	
Device Options This device uses sip secret dtmfmode	technology. h7Dka3Rf9si0t rfc2833	

- 12. Click on "Submit" at the end of the page and Apply changes. Now for incoming calls we will create an IVR. To do this, go to "PBX => PBX Configuration => IVR". Click on "Add IVR" link (Figure 4.9). Set the following:
  - **Name:** Name of IVR (WelcomeIVR in this example)
  - Announcement: Voice prompt which will be played for incoming calls.
  - Options:
    - \* Phone book.
    - o 0 302 Extension
    - $\circ~t~$  Repeat the options of IVR (Add this option by modifying the IVR after creation)

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#### Figure 4-9. IVR

igital Recep	otionist	
lit Menu Welco	melVR	
ave Delete Digit sed as Destination	al Receptionist Welcome on by 2 Objects:	EIVR
nange Name	WelcomeIVR	
nouncement	welcome 📩	
ook Directory 🖠	Phonebook Directory	Return to IVR 🗆 🔀
ons 🔳	<302> IPPhone 📩	Return to IVR 🗆 🛢
	WelcomeIVR 🚽	Return to IVR 🔲 🖀
0	ns 1 Save Dec	ns (302> IPPhone ) WelcomeIVR ) Save Decrease Options

- 13. Click on "Save" and Apply changes by clicking on the pink ribbon that appears at the top of the page. Now go to "PBX => PBX Configuration => Inbound Routes". Click on "Add Incoming Route" link (Figure 4.10). Set the following:
  - **Description:** Name of inbound route ("Incoming\_Calls" in this example)
  - Set destination: Where the call will be routed. ("WelcomeIVR" IVR in this example)

#### **Figure 4-10. Incoming Route**

d Incoming Route	
escription:	Incoming_Calls
Set Destination	

- 14. Click on "Submit" and apply changes. Now when we receive calls the "WelcomeIVR" IVR will play to the caller giving him choices to interact with Elastix Server.
- 15. To configure the gateway, you will need to enter the information from the trunk created on the Elastix Server into the Vega 5000 and set other parameters by logging into the WebUI.

For the initial configuration, refer to the Vega 5000 Admin Guide found at: <u>http://wiki.sangoma.com/Vega-5000-Technical-Documentation</u>

Note: Reset the Vega 5000 to factory default settings before continue.

# LAN1 IP AddressDHCPLAN1 IP Address (If DHCP no available)169.254.xxx.yyyWeb Access Administrator UseradminWeb Access Administrator Passwordadmin

#### **Factory default settings**



#### Use Vega default IP address If the Vega is powered up and no DHCP server is available, then the Vega will set its IP address to 169.254.xxx.yyy where xxx and yyy are defined by the MAC address of the Vega LAN interface. xxx and yyy are both one to three digit decimal values. The MAC address of the Vega LAN interface will be 00:50:58:WW:XX:YY (found on the rear of the Vega – for details see the 'Use IP ping and Arp cache' section above) where WW, XX and YY are each 2 hexadecimal digits. • The xxx value in the IP address is the decimal value of the XX hex value from the MAC address, and • The yyy value in the IP address is the decimal value of the YY hex value from the MAC address.

Go to the Vega 5000's WebUI by pointing your browser to the Vega's IP address (Figure 4-11).

Figure 4-11. Vega 5000's WebUI

Vega Config	5000 GW guration	
Login		
Enter Use	rname and Password	
Username	admin	
Password		
Login		

16. When the WebUI is loaded, go to "Quick Configuration" located on the left side of the page and click on "Continue" button (Figure 4-12).

Figure 4-12. Quick Config

	Vega 5000 GW Configuration	
Status Quick Config Expert Config		Warning
Warnings(6) Log off <u>Reboot System</u>		If you enter Cluck Setup and select Submit some of your current settings previously configured in the Expert setup pages or by a configuration upload will be changed. Please see your Administrator to decide how you should proceed. Exit Continue Show Details

17. Once there, go to "Basic Config" tab and set the following (Figure 4-13):

#### General

• Country: US

#### Figure 4-13. Quick Config – Basic Config

	Vega 5000 GW Configuration	1				
Status Quick Config	Basic Config	⑦ VolP	FXS FXO	Submit	New Install? Exit	0
Expert Config	General					
Warnings(6)	Country US 🛔 Tin	nezone Offset (HHM	M) 0000	Emergency Numbers	999,112,911,000	
Log off	LAN 1 Configuration					

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18. Go to "VoIP" tab and set the following information with the same data as step 6 (Figure 4-14):

VoIP Device Configuration

- Registration Mode: Gateway
- SIP Server IP/Name: Elastix Server's IP Address (192.168.1.75)
- **Registration and Authentication ID:** (Vega5000 in this example)
- **Authentication Password:** (jx8FkOU13sv6 in this example)

#### Figure 4-14. Quick Config – VOIP

Basic Config		FXS F	хо	Submit	New Install? Exi
VoIP Device Configuration	1				
Registration Mode	Gat	teway 🌲			
Outbound Proxy Used?		No 🛓			
SIP Domain	default-reg-dom	nain.com			
SIP Server IP/Name	192.168.1.75				
Outbound proxy IP/Name	0.0.0				
Registration and Authentication	Vega5000				
Authentication Password	INPEKOLI136V6				

19. Go to "FXS" tab and check the "Enable Caller ID Generation" option. (Figure 4-15).

Telepho	ne Conr	nections					
Interface	Enabled	Enable Caller-Id Generation	Numeric Caller ID copy to number list	Textual Caller ID	Telephone number(s) to route to the FXS interface	Registration and Authentication ID Numeric_ID	Authentication Password
0101			0101	FXS1	0101	default	****
0102			0102	FXS2	0102	default	****
0103			0103	FXS3	0103	default	****
0104			0104	FXS4	0104	default	****
0105			0105	FXS5	0105	default	****

Figure 4-15. Quick Config – FXS

20. Go to "FXO" tab and check the "Tone Based Disconnect" and "Enable Caller ID Detection" option. (Figure 4-16).

Figure 4-16. Quick Config - FXO

		9				
Tone B	ased Disc	onnect 1				
Enable	~					
chable			,			
Teleph	one Conne	ctions				
Teleph Interface	one Conne Handle Emergency Calls?	Enable Caller ID Detection	Numeric Caller ID	Textual Caller ID	Telephone number(s) to route to the FXO interface	DID to Forward to SI
Teleph Interface 0201	one Conne Handle Emergency Calls?	Enable Caller ID Detection	Numeric Caller ID	Textual Caller ID	Telephone number(s) to route to the FXO Interface	DID to Forward to SI

21. To apply the changes click on "Submit" button in red next to the tabs menu. (Figure 4-17):

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Figure 4-17. Applying changes



22. Click on "Accept". Now, go to "Status" located on the left side of the page to check whether the registration was successful (Figure 4-18).

#### Figure 4-18. Status

_		
SIP	REG	
SIP	Profile 1	- registration expiry = 600s
SIP	REG USER 1	
	address	- Vega5000@default-reg-domain.com
	auth user	- Vega5000
	contact	- <sip:vega5000@192.168.5.247></sip:vega5000@192.168.5.247>
	state	- registered (user 1)
	TTL	- 556 seconds
SIP	REG USER 2	
	address	- 02@default-reg-domain.com

23. If the gateway is not registered check you have entered the correct information. Now go to "Expert Config" menu on the left side, and click on "SIP" option. (Figure 4-19).

Figure 4-19. Expert Config - SIP

atus	SIP Configu	uratio	n		
uick Config	General				
cpert Connig	Local SIP Port	5	060		
ogging	Accept Non-Pro	oxy			
AN/WAN OTS	Submit				
ial Plan	SIP Profiles				
edia ones	SIP Profile	Nam	e Interface ID	Other SIP Profile Parameters	Chg?
P	1	profile	9901	===>	Modify
S Statistics	Add Delete				

24. In the "SIP Profiles" section click on "Modify", and set the *From header user info* parameter to *Calling party*. This setting will allow us to see the caller ID in a call. (Figure 4-20).



Status Quick Config	<u>51P</u> > 51P	Profile 1
Expert Config	SIP Profile	1 Configuration 1
System	Name	profile1
Lan/wan	Interface ID	9901
POTS	Local Domain	default-reg-domain.com
Dial Plan	Alternative Local Domain	alt-reg-domain.com
Tones	From Header 'userinfo'	Calling Party
SIP	From Header 'host'	Local Domain 🛔
Supp.Services	To Header 'host'	Local Domain 🛔
Advanced	Redirection	Local Domain *

25. Click on "Submit" button. To apply and save all the changes we have made, click on the red buttons located on the left. They are "Apply Changes" and "Save", in that order (Figure 4-21).

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#### Figure 4-21. SIP Profile

Status Quick Config	<u>511</u> × 511 1	Tome I
Expert Config	SIP Profile	1 Configuration 1
System	Name	profile1
Logging LAN/WAN	Interface ID	9901
POTS	Local Domain	default-reg-domain.com
Dial Plan	Alternative	alt-reg-domain.com
Media Tones	From Header	Calling Party
SIP	From Header	Local Domain 🛔
QoS Statistics Supp.Services	To Header	Local Domain 🗍
Advanced	Redirection	Local Domain 🛔
Apply Changes	Transport	udp 🛊
Save	Capability Set	2 - voice+t38Udp *
Warnings(7)	Reliable Provisional	

26. Once you have saved the changes, go to "Dial Plan" located on the "Expert Config" section (Figure 4.22)

Figure 4-22. Dial Plan

	Ve Co	ga nfi	5000 GW guration			
Status Quick Config	Dial F	<b>Plan</b>	ner			
xpert Config	Del?	ID	Name	E	nabled	
System		1	new_profile			Modify
ogging AN/WAN		20	To_SIP		~	Modify
DTS		23	To_FXO		~	Modify
al Plan		24	To FXS		~	Modify
edia	Add		Delete Submit			

27. Here we're going to create a dial plan to route the calls. Delete all the entries and create 3 profiles that will be used for incoming calls to FXS, FXO and SIP interface. Below is the configuration of each profile (Figure 4-23).

Figure 4-23. Dial Plan

tatus	Dial F	lan	ner		
uick Config	Profi	les			
xpert Config	Del?	ID	Name	Enabled	
System		1	Incoming_FXO		Modify
.ogging AN/WAN		2	Incoming_FXS		Modify
отз		3	Incoming SIP		Modify

For a better understanding: 9901 : SIP Interface ID 0101 : FXS 1 port ID (0102 FXS 2, 0103 FXS 3, etc) 0201 : FXO 1 port ID (0202 FXO 2)

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#### Incoming\_FXO

ID	Name	Source	Destination
1	To_SIP	IF:0201,TEL:<.*>,TELC:<.*>	IF:9901,TEL:<1>,TELC:(<2>)
2	To_FXS1	IF:0201,TEL:(0101)	IF:0101,TEL:0101

The first rule will route the calls to the SIP Trunk. The dialed number can be any except 0101.

The second rule will route the call to the FXS1 port. The call takes this route provided the dialed number is 0101 (This is the extension number of the analog phone connected in the first FXS).

#### Figure 4-24. Incoming\_FXO profile

Plans In	This Profile				
Del? Plan	Name	Source	Destination	Cost	Group
1	To_SIP	IF:0201,TEL:<.*>,TELC:<.*>	IF:9901,TEL:<1>,TELC:(<2>)	0 🛔	0 - None
□ 2	To FXS1	IF:0201,TEL:(0101)	IF:0101,TEL:0101	0 1	0 - None

#### Incoming\_FXS

ID	Name	Source	Destination
1	To_SIP	IF:0101,TEL:<.*>	IF:9901,TEL:<1>
2	To_FXO1	IF:0101,TEL:(9<.*>)	IF:0201,TEL:<1>

The first rule will route the calls to the SIP Trunk. The dialed number can be any but it has not to begin with 9.

The second rule will route the call to the FXO1 port. The call takes this route provided the dialed number begins with 9.

#### Figure 4-25. Incoming\_FXS

Plans In This Profile					
Del? Plan ID	Name	Source	Destination	Cost	Group
1	To_SIP	IF:0101,TEL:<.*>	IF:9901,TEL:<1>	0 🛓	0 - None
2	To FXO1	IF:0101,TEL:(9<.*>)	IF:0201,TEL:<1>		0 - None

#### Incoming\_SIP

.

ID	Name	Source	Destination
1	To_FXO1	IF:9901,TEL:<.*>	IF:0201,TEL:<1>
2	To_FXS1	IF:9901,TEL:(0101)	IF:0101,TEL:0101

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The first rule will route the calls to the FXO1 port. The dialed number can be any except 0101.

The second rule will route the calls to FXS1 port provided the dialed number is 0101 (This is the extension number of the analog phone connected in the first FXS).

Figure 4-26. Incoming\_SIP profile

Plar	is In Th	is Profile				
Del?	Plan ID	Name	Source	Destination	Cost	Group
	1	To_FXO1	IF:9901,TEL:<.*>	IF:0201,TEL:<1>	0 🛔	0 - None
	2	To FXS1	IF:9901,TEL:(0101)	IF:0101,TEL:0101	0 1	0 - None

- 28. For more help using expressions you can take a look on the **Regular Expression** Help and Token Help section located below of the plans of each dial plan profile. Don't forget to Apply and Save changes clicking on the red buttons located on the left.
- 29. Now that you have configured the dial plan, you are able to make the following calls:

From PSTN to Elastix server, From PSTN to Analog phones connected to Vega From Analog phones to Elastix Server From Analog phones to PSTN From Elastix Server to Analog phones connected to Vega From Elastix Server to PSTN