

How to configure Synway Gateway for a connection with Elastix System

Technical Department

2015-10

Catalog

1. Abstract.....	3
2. Test Environment.....	3
3. Installing Elastix.....	3
4. Configuring Synway Digital Gateway for a Connection with Elastix.....	4
5. Configuring Synway Analog Gateway for a Connection with Elastix.....	15

1. Abstract

Elastix is an Open Source Software to establish Unified Communications. About this concept, Elastix goal is to incorporate all the communication alternatives, available at an enterprise level, into a unique solution.

Synway SMG gateway family helps customers access to IP networks from legacy telephony applications more reliably and efficiently. Synway Gateway got certified and is now part of the Hardware that has been successfully tested to be interoperable with Elastix.

This document will help you to configure Synway Digital or Analog Gateway for a connection with Elastix System.

2. Test Environment

Elastix 2.4.0 (32 bit)

Synway Digital Gateway: SMG2120, 1.6.1_2015062617

Synway Analog Gateway:SMG1032, 1.5.2_Release+2015052812

3. Installing Elastix

Please refer to <http://blogs.elastix.org/en/manuals/#toggle-id-1>, you can find [Elastix 2 Installation Guide manual](#).

4. Configuring Synway Digital Gateway for a Connection with Elastix

Elastix IP Address: **192.168.10.163**

Synway Digital Gateway IP Address: **192.168.10.248**

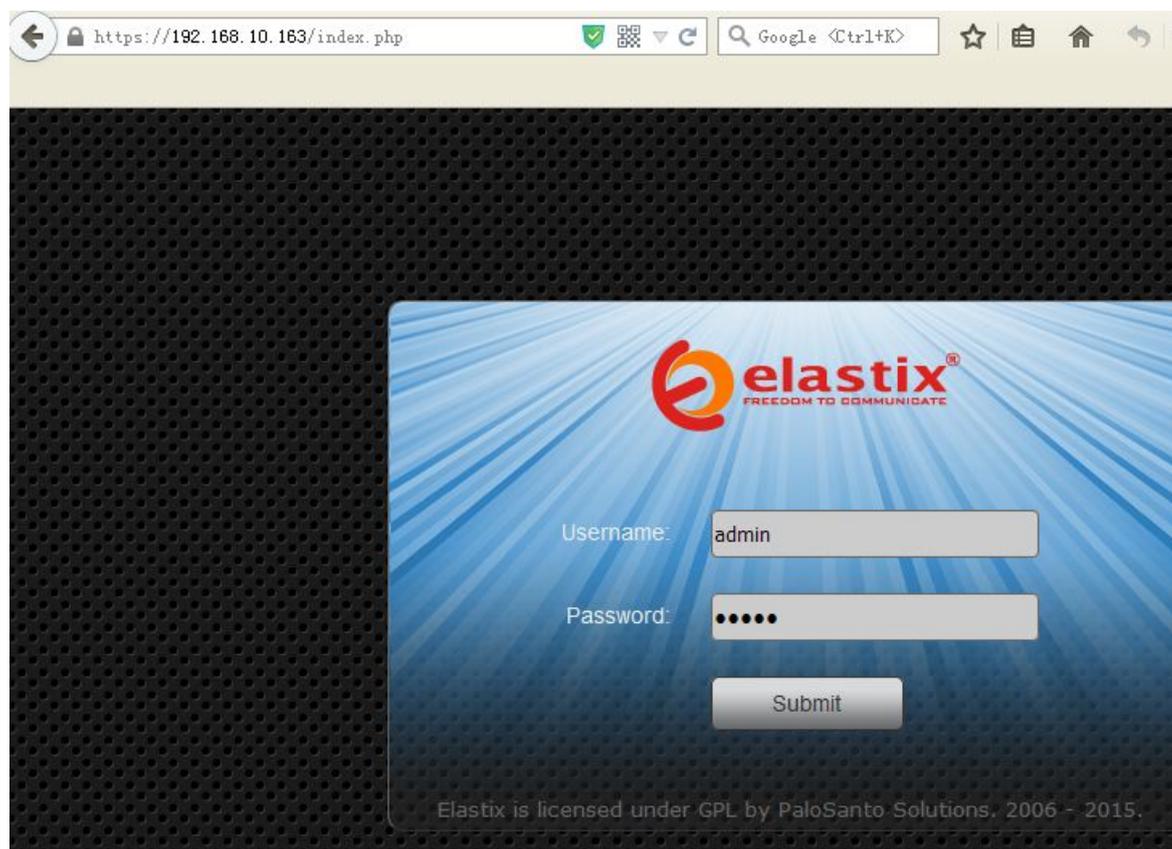
Synway Digital Gateway E1 port PCM0 connects to E1 endpoint , here we use Synway SHD digital trunk board as the E1 endpoint, both are configured in PRI protocol

Below is the configuration among Elastix System, Eyebeam, and Synway Digital Gateway, try to make calls from these scenarios:

a) Eyebeam-->Elastix-->Digital Gateway-->E1 Endpoint

b) E1 Endpoint-->Digital Gateway-->Elastix-->Eyebeam

1) To configure the Elastix system, start a web browser and enter the IP address of the Elastix System. Here is an example, <http://192.168.10.163>.



- 2) To add an extension, go to the 'PBX' menu, which by default goes to the 'PBX Configuration' section, choose the option 'Extensions' on the left panel, select device type as 'Generic SIP Device', click 'Submit', then specify a user extension, display name and password for this extension, do not forget to click 'Apply Config' after any change at last.

The screenshot displays the Elastix web interface for PBX configuration. The browser address bar shows the URL `https://192.168.10.163/config.php?type=setup&display=ext`. The interface features a top navigation bar with tabs for System, Agenda, Email, Fax, and PBX. Below this is a secondary navigation bar with options: PBX Configuration, Operator Panel, Voicemails, Calls Recordings, Batch Configurations, and Conference. The left sidebar lists various configuration categories: Basic (with sub-items: Extensions, Feature Codes, Outbound Routes, Trunks), Inbound Call Control (with sub-items: Inbound Routes, DAHDI Channel DIDs, Announcements, Blacklist, CallerID Lookup Sources, Call Flow Control, Follow Me), IVR, Queue Priorities, Queues, Ring Groups, and Time Conditions. The main content area is titled 'Add an Extension' and contains the instruction: 'Please select your Device below then click Submit'. Below this instruction, there is a label '- Device' followed by a horizontal line. A 'Device' dropdown menu is set to 'Generic SIP Device'. A 'Submit' button is located below the dropdown.

elastix
FREEDOM TO COMMUNICATE

System | Agenda | Email | Fax | **PBX** | II

PBX Configuration | Operator Panel | Voicemails | Calls Recordings | Batch Configurations | Conference

PBX Configuration

- Basic
 - Extensions
 - Feature Codes
 - Outbound Routes
 - Trunks
- Inbound Call Control
 - Inbound Routes
 - DAHDI Channel DIDs
 - Announcements
 - Blacklist
 - CallerID Lookup Sources
 - Call Flow Control
 - Follow Me
 - IVR
 - Queue Priorities
 - Queues
 - Ring Groups

Add SIP Extension

- Add Extension

User Extension [?]

Display Name [?]

CID Num Alias [?]

SIP Alias [?]

- Device Options

This device uses sip technology. extension password

secret [?]

dtmfmode [?]

nat [?]

elastix
FREEDOM TO COMMUNICATE

System | Agenda | Email | Fax | **PBX** | IM | Reports | v

PBX Configuration | Operator Panel | Voicemails | Calls Recordings | Batch Configurations | Conference | Tools | Flash C

PBX Configuration

- Basic
 - Extensions
 - Feature Codes
 - Outbound Routes

3) To set an Eyebeam registering to the Elastix System, the User name and Password fields here must match the extension in the Elastix System.



Settings ✕

Choose Setting Category

- [-] SIP Accounts
 - [+] **192.168.10.163**
 - [+] Add a New SIP Account
- [+] Media
- [+] System
- [+] User Interface
- [+] Diagnostics
- [+] License Key

Enable this SIP account

User Details

Display Name	1001
User name	1001
Password	*****
Authorization user name	
Domain	192.168.10.163

Domain Proxy

Register with domain

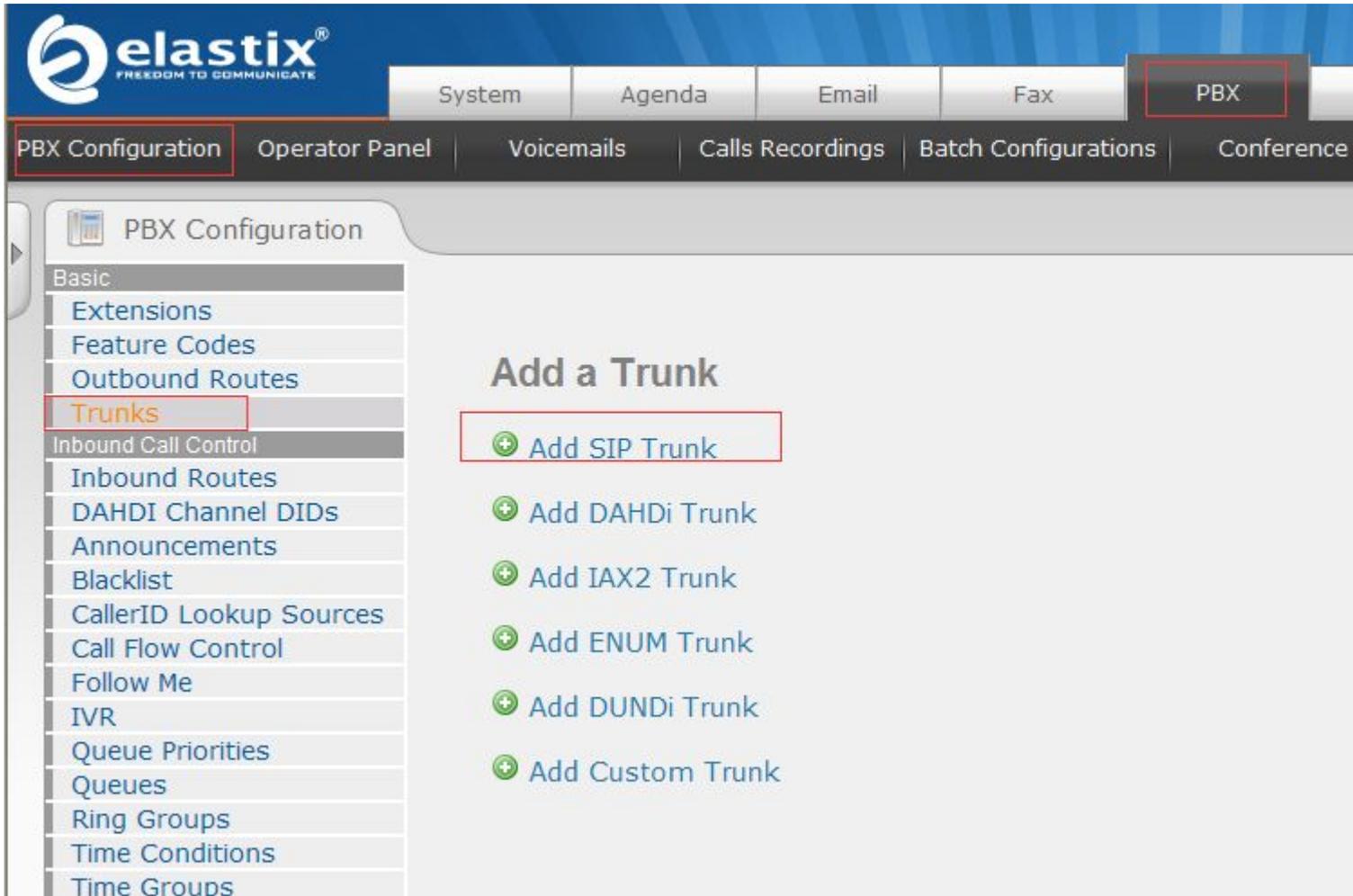
Use as Outbound Proxy

Manual Override Host

SIP Listen Port

Manual override

- 4) To add a Synway Digital Gateway as a Sip Trunk, click on 'Trunks' from the toolbar, follow below steps to add a 'Digital_SMG' sip trunk.



The screenshot displays the Elastix PBX Configuration web interface. At the top, the Elastix logo is visible on the left, and navigation tabs for 'System', 'Agenda', 'Email', 'Fax', and 'PBX' are on the right. Below these, a secondary navigation bar includes 'PBX Configuration', 'Operator Panel', 'Voicemails', 'Calls Recordings', 'Batch Configurations', and 'Conference'. The main content area is titled 'PBX Configuration' and features a left-hand sidebar with a tree view of configuration categories. The 'Trunks' category is highlighted in orange. The main panel is titled 'Add a Trunk' and lists six options, each with a green plus icon in a circle: 'Add SIP Trunk', 'Add DAHDi Trunk', 'Add IAX2 Trunk', 'Add ENUM Trunk', 'Add DUNDi Trunk', and 'Add Custom Trunk'. The 'Add SIP Trunk' option is highlighted with a red rectangular border.

- Basic
 - Extensions
 - Feature Codes
 - Outbound Routes
 - Trunks
- Inbound Call Control
 - Inbound Routes
 - DAHDI Channel DIDs
 - Announcements
 - Blacklist
 - CallerID Lookup Sources
 - Call Flow Control
 - Follow Me
 - IVR
 - Queue Priorities
 - Queues
 - Ring Groups
 - Time Conditions
 - Time Groups
- Internal Options & Configuration
 - Conferences
 - Languages
 - Misc Applications
 - Misc Destinations
 - Music on Hold
 - PIN Sets
 - Paging and Intercom
 - Parking Lot
 - System Recordings
 - VoiceMail Blasting
- Remote Access
 - Callback
 - DISA
- Option
 - Unembedded FreePBX®

Add SIP Trunk

General Settings

Trunk Name [?]:

Outbound CallerID [?]:

CID Options [?]:

Maximum Channels [?]:

Asterisk Trunk Dial Options [?]: Override

Continue if Busy [?]: Check to always try next trunk

Disable Trunk [?]: Disable

Dialed Number Manipulation Rules [?]

(prepend) + prefix | match pattern

Dial Rules Wizards [?]:

Outbound Dial Prefix [?]:

Outgoing Settings

Trunk Name [?]:

PEER Details [?]:

→ Synway Digital Gateway IP address is 192.168.10.248

- 5) Create an outbound call rule to Synway Digital SMG. For example, when making an outbound call from the extension 1001, the other side will receive a call with caller Id 1001, here the dial pattern is '8.', which means the callee id starting with 8, can be routed to the 'Digital_SMG' SIP trunk.

- Feature Codes
- Outbound Routes
- Trunks
- Inbound Call Control
- Inbound Routes
- DAHDI Channel DIDs
- Announcements
- Blacklist
- CallerID Lookup Sources
- Call Flow Control
- Follow Me
- IVR
- Queue Priorities
- Queues
- Ring Groups
- Time Conditions
- Time Groups
- Internal Options & Configuration
- Conferences
- Languages
- Misc Applications
- Misc Destinations
- Music on Hold
- PIN Sets
- Paging and Intercom
- Parking Lot
- System Recordings
- VoiceMail Blasting
- Remote Access
- Callback
- DISA
- Option
- Unembedded FreePBX®

Add Route

Route Settings

Route Name: Elastix_to_Digital_SM

Route CID: Override Extension

Route Password:

Route Type: Emergency Intra-Company

Music On Hold?: default

Time Group: ---Permanent Route---

Route Position: Last after Elastix_to_Digital_SMG

Additional Settings

Call Recording: Allow

PIN Set: None

Dial Patterns that will use this Route

(prepend) + prefix | [8. / CallerID]

+ Add More Dial Pattern Fields

Dial patterns wizards: (pick one)

Trunk Sequence for Matched Routes

0 Digital_SMG

- 6) Create an inbound call rule with DID Number '12345', which means call from the Synway Digital Gateway Sip Trunk with callee ID 12345 will be accepted by the Elastix System, and this call will be transferred to extension '1001'.

- Extensions
- Feature Codes
- Outbound Routes
- Trunks
- Inbound Call Control
- Inbound Routes
- DAHDI Channel DIDs
- Announcements
- Blacklist
- CallerID Lookup Sources
- Call Flow Control
- Follow Me
- IVR
- Queue Priorities
- Queues
- Ring Groups
- Time Conditions
- Time Groups
- Internal Options & Configuration
- Conferences
- Languages
- Misc Applications
- Misc Destinations
- Music on Hold
- PIN Sets
- Paging and Intercom
- Parking Lot
- System Recordings
- VoiceMail Blasting
- Remote Access
- Callback
- DISA

Add Incoming Route

Add Incoming Route

Description [?]:

DID Number [?]:

CallerID Number [?]:

CID Priority Route [?]:

Options

Alert Info [?]:

CID name prefix [?]:

Music On Hold [?]: ▼

Signal RINGING [?]:

Pause Before Answer [?]:

Privacy

Privacy Manager [?]: ▼

Language

Set Destination

Extensions ▼
<1001> 1001 ▼

- 7) To configure Synway Digital Gateway connecting with Elastix System, start a web browser and enter the IP address of the Synway Digital Gateway.

The screenshot shows a web browser at the URL 192.168.10.248/en/navigation.php. The page displays the 'System Info' section of the gateway's configuration interface. The left sidebar contains various menu items like 'Operation Info', 'PSTN Status', 'Call Count', 'VoIP', 'PCM', 'ISDN', 'Fax', 'Route', 'Number Filter', 'Num Manipulate', and 'System Tools'. The main content area shows details for LAN 1 and LAN 2, including MAC addresses, IP addresses, DNS servers, packet statistics, and current speeds. It also displays runtime information, operating mode, and current version details.

LAN 1			
MAC Address	00:00:E0:10:10:9B		
IP Address	192.168.10.248	255.255.255.0	192.168.10.254
DNS Server	0.0.0.0		
Receive Packets	All:3317907	Error:0	Drop:0
Transmit Packets	All:147575	Error:0	Drop:0
Current Speed	Receive:11.7 KB/s	Transmit:10.5 KB/s	
Work Mode	100Mb/s Full Duplex		

LAN 2			
MAC Address	00:00:E0:10:10:9C		
IP Address	192.168.0.101	255.255.255.0	192.168.0.254
DNS Server	0.0.0.0		
Receive Packets	All:0	Error:0	Drop:0
Transmit Packets	All:0	Error:0	Drop:0
Current Speed	Receive:0 B/s	Transmit:0 B/s	
Work Mode	10Mb/s Half Duplex		

Runtime	13d 0h 39m 28s
Operating Mode	ISDN(user)
Current Version	
Serial Number	000000106(4)
WEB	1.6.1_2015062617
Gateway	1.6.1_2015062617
Uboot	2.0.6_201407
Kernel	#208 Thu Mar 26 15:10:01 CST 2015
Firmware	18

- 8) Click on 'SIP Trunk' from the toolbar, add the Elastix System as a sip trunk, here the Elastix IP address is 192.168.10.163, and port is 5060.

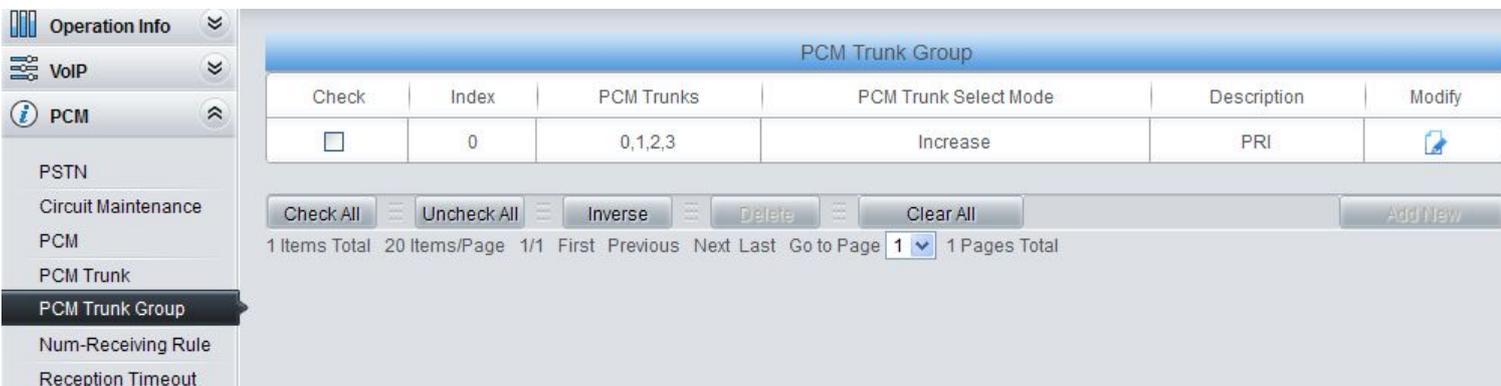
The screenshot shows the 'SIP Trunk' configuration page in the Synway Digital Gateway web interface. The left sidebar has 'SIP Trunk' selected. The main area displays a table with columns for 'Check', 'Index', 'Remote Address', 'Remote Port', 'WAN Option', and 'Outgoing Voice R'. A single entry is shown with index 0, remote address 192.168.10.163, and remote port 5060. Below the table are buttons for 'Check All', 'Uncheck All', 'Inverse', 'Delete', and 'Clear All'.

Check	Index	Remote Address	Remote Port	WAN Option	Outgoing Voice R
<input type="checkbox"/>	0	192.168.10.163	5060	NET 1	128

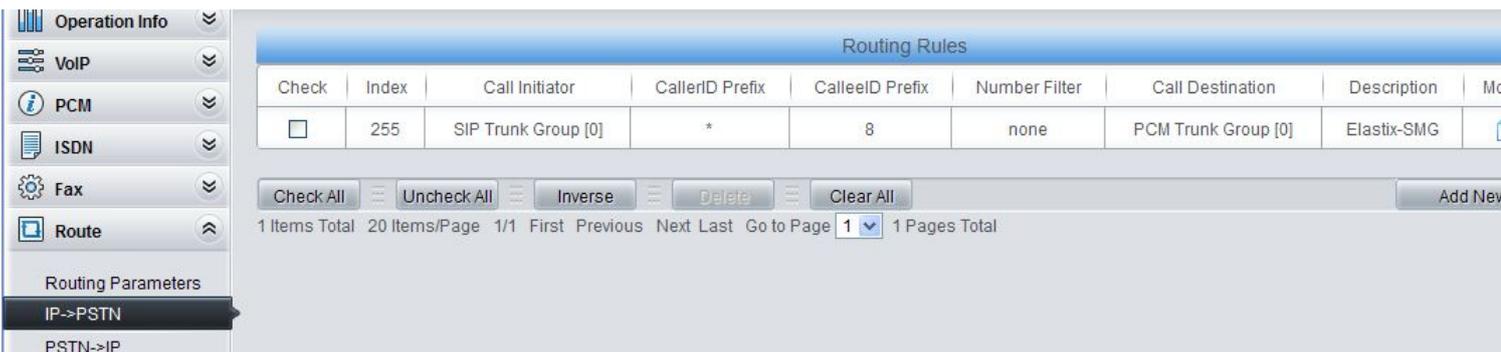
9) Click on 'SIP Trunk Group' on the toolbar, add the SIP Trunk 0 into SIP Trunk Group 0



10) Click on 'PCM Trunk Group' on the toolbar, according to the requirement, add related PCM Trunk(s) into PCM Trunk Group. Here add all the PCM Trunks 0,1,2,3 into PCM Trunk Group 0.



11) Click on 'Route\ IP->PSTN' on the toolbar, call from the Elastix System will be routed to PCM Trunk Group 0.



12) Click on 'Route\ PSTN->IP' from the toolbar, call from E1 Endpoint will be routed to SIP Trunk Group 0.

Operation Info

VoIP

PCM

ISDN

Fax

Route

Routing Parameters

IP->PSTN

PSTN->IP

Routing Rules

Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Number Filter	Call Destination	Description	Mo
<input type="checkbox"/>	255	PCM Trunk Group [0]	*	*	none	SIP Trunk Group [0]	SMG-Elastix	

1 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total

- 13) Extension 1001 in the Elastix System made a call 8001 to Synway Digital Gateway, it finally reached to E1 Endpoint with caller Id 1001, and callee Id 8001.

Test (32bit version)

System Function Test New Function Test Help

SS7 FSK BUS CONF SET PCM DIGT SPY

C...	Type	State	RCnt	TnC...	DtmfRcvBuffer	CallerId	CalleeD.	SuspendClause	V..
0	ISDNNetio	Idle							
1	ISDNNetio	Idle							
2	ISDNNetio	Idle							
3	ISDNNetio	Idle							
4	ISDNNetio	Idle							
5	ISDNNetio	Idle							
6	ISDNNetio	Idle							
7	ISDNNetio	Idle							
8	ISDNNetio	Idle							
9	ISDNNetio	Idle							
10	ISDNNetio	Idle							
11	ISDNNetio	Talking				1001	8001		
12	ISDNNetio	Idle							
13	ISDNNetio	Idle							
14	ISDNNetio	Idle							
15	ISDNNetio	Idle							
16	ISDNNetio	Idle							
17	ISDNNetio	Idle							
18	ISDNNetio	Idle							
19	ISDNNetio	Idle							
20	ISDNNetio	Idle							

BasicFun PlaybackFun RecordFun EventDriven IP

Pickup/Hanguup

SsmPickup

SsmHangup

SsmClearCallerId

SsmClearCallerIdEx

Send DTMF

DTMF: 114

SsmAutoDial

SsmTxDtmf

SsmAppendPhoNum

P... Sig... PcmSy... AutoConn CallDire... Rece.

Call established

8001@192.168.10.163

ADD TO CONTACTS

0:00:24

BAN

g711a

1 2 3 4 5 6

XFER HOLD PARK AA AC DND CONF

FLASH REC REDIAL

1 2 3

4 5 6

GHJ IKL MNO

Receive

EndCha

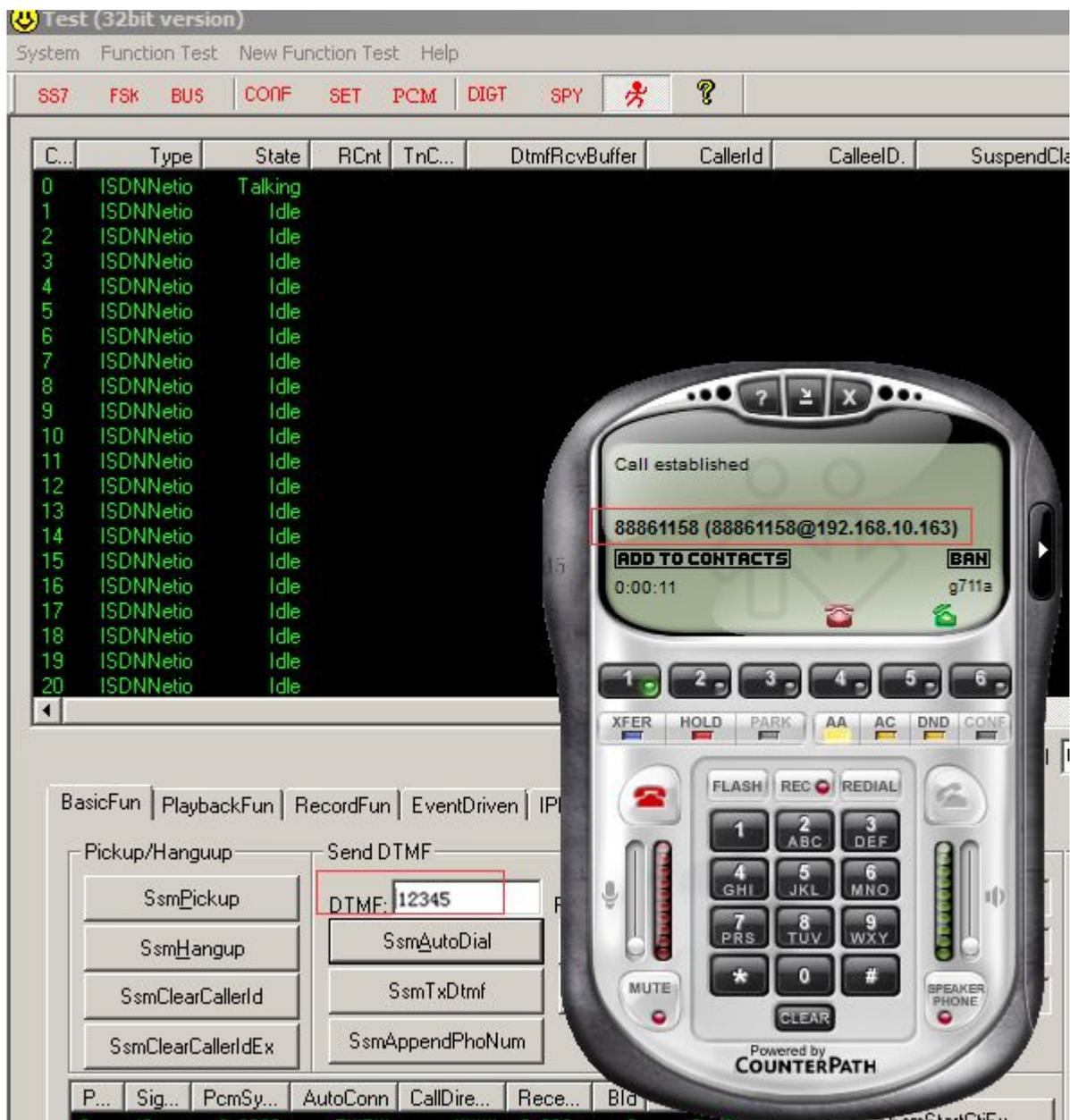
TimeOu

MaxLer

Ssm

ChkWt

14) E1 endpoint made a call '12345' to the Synway Digital Gateway, it finally reached to extension 1001 in the Elastix System.



5. Configuring Synway Analog Gateway for a Connection with Elastix

Elastix IP Address: **192.168.10.163**

Synway Analog Gateway IP Address: **192.168.10.189**

Below is the configuration among Elastix System, and Synway Analog Gateway, FXS 9 and FXS 10 in the Synway Analog Gateway registered to Elastix, try to make calls from these scenarios:

a) FXS-->Elastix-->FXS

- b) FXS-->Elastix-->Eyebeam
- c) Eybeam-->Elastix-->FXS
- d) Eybeam-->Elastix-->Analog Gateway-->PSTN
- e) PSTN-->Analog Gateway-->Elastix-->Eyebeam

1) Add two more extensions 1003 and 1004 in the Elastix System, for details how to add an extension, please refer to Chapter4, Section2 in this document.

2) Add the Synway Analog Gateway as a VoIP SIP Trunk, the Analog Gateway IP address is 192.168.10.189.

The screenshot shows the Asterisk SIP Trunk configuration page for 'Analog_SMG'. The left sidebar contains a navigation menu with categories like 'Trunks', 'Inbound Call Control', 'Internal Options & Configuration', 'Remote Access', and 'Option'. The main content area is titled 'Delete Trunk Analog_SMG' and shows the trunk is 'In use by 1 route'. It is divided into three sections: 'General Settings', 'Dialed Number Manipulation Rules', and 'Outgoing Settings'. In the 'General Settings' section, 'Trunk Name' is 'Analog_SMG', 'Outbound CallerID' is empty, 'CID Options' is 'Allow Any CID', 'Maximum Channels' is empty, and 'Asterisk Trunk Dial Options' is empty with an 'Override' checkbox. In the 'Outgoing Settings' section, 'Trunk Name' is 'Analog_SMG' and 'PEER Details' contains 'host=192.168.10.189' and 'type=peer'. The 'Dialed Number Manipulation Rules' section shows a rule with '(prepend) + prefix | match pattern' and buttons for '+ Add More Dial Pattern Fields' and 'Clear all Fields'.

- 3) Create an outbound call rule to Synway Analog SMG. For example, when making an outbound call from the extension 1001, the other side will receive a call with caller Id 1001, here the dial pattern is '0.', which means the callee id starting with 0, can be routed to the 'Analog_SMG' SIP trunk.

- Outbound Routes
- Trunks
- Inbound Call Control
- Inbound Routes
- DAHDI Channel DIDs
- Announcements
- Blacklist
- CallerID Lookup Sources
- Call Flow Control
- Follow Me
- IVR
- Queue Priorities
- Queues
- Ring Groups
- Time Conditions
- Time Groups
- Internal Options & Configuration
- Conferences
- Languages
- Misc Applications
- Misc Destinations
- Music on Hold
- PIN Sets
- Paging and Intercom
- Parking Lot
- System Recordings
- VoiceMail Blasting
- Remote Access
- Callback
- DISA
- Option
- Unembedded FreePBX®

Add Route

Route Settings

Route Name:

Route CID: Override Extension

Route Password:

Route Type: Emergency Intra-Company

Music On Hold?:

Time Group:

Route Position:

Additional Settings

Call Recording:

PIN Set:

Dial Patterns that will use this Route

(prepend) + prefix || / CallerID

[+ Add More Dial Pattern Fields](#)

Dial patterns wizards:

Trunk Sequence for Matched Routes

- 4) To configure Synway Analog Gateway connecting with Elastix System, start a web browser and enter the IP address of the Synway Analog Gateway.

The screenshot shows a web browser window with the address bar displaying `192.168.10.189/en/navigation.php`. The page content includes a navigation menu on the left and a main content area titled "System Info".

System Info

LAN 1			
MAC Address	00:00:E0:10:0E:CD		
IP Address	192.168.10.189	255.255.255.0	192.168.10.254
DNS Server	0.0.0.0		
Receive Packets	All:691	Error:0	Drop:0
Transmit Packets	All:1372	Error:0	Drop:0
Current Speed	Receive:2.2 KB/s	Transmit:866 B/s	
Work Mode	100Mb/s Full Duplex		
LAN 2			
	Disable		
Runtime	6m 9s		
Current Version			
WEB	1.5.2_Release+2015052812		
Gateway	1.5.2_Release+2015052812		
Serial Num	000000014		
Authorization Code	0x7		
U-boot	Nov 24 2014 - 09:24:52		
Kernel	#186 PREEMPT Mon Mar 2 09:06:53 CST 2015		
Firmware	104		
Device Type	1a4		

At the bottom right of the System Info panel, there is a "Refresh" button.

- 5) Click on 'VoIP\SIP' from the toolbar, as two FXS extensions in the Analog Gateway should register to Elastix system, here set the Register IP Address is 192.168.10.163.

Wed Oct 14 2015 10:30:17 GMT+0800 Language English Current t

- Operation Info
- Quick Config
- VoIP
- SIP
- Sip Compatibility
- NAT Setting
- Media
- Advanced
- Port
- Route
- Num Manipulate
- System Tools

SIP Settings

SIP Address	LAN 1: 192.168.10.189
SIP Port	5060
Register Status	Unregistered
Register Gateway	No
Registrar IP Address	192.168.10.163
Registrar Port	5060
Spare Registrar Server	<input type="checkbox"/> Enable
Registry Validity Period (s)	3600
Multi-Registrar Server Mode	<input type="checkbox"/> Enable
SIP Transport Protocol	UDP
IMS Network	<input type="checkbox"/> Enable

- 6) Port 9 and Port 10 are FXS type, set these two ports registering to Elastix with extension number 1003 and 1004.

- Operation Info
- Quick Config
- VoIP
- Advanced
- Port
- FXS
- FXO
- Port Group
- Route
- Num Manipulate
- System Tools

FXS-Modify

Port	9
Type	FXS
Register Port	Yes
SIP Account	1003
Password	••••••••
Auto Dial Number	
Wait Time before Auto Dial (s)	0
Echo Canceller	<input type="checkbox"/> Enable
Forbid Outgoing Call	<input type="checkbox"/> Enable
CID	<input checked="" type="checkbox"/> Enable
Call Waiting	<input type="checkbox"/> Enable

- Operation Info
- Quick Config
- VoIP
- Advanced
- Port
- FXS
- FXO
- Port Group
- Route
- Num Manipulate
- System Tools

FXS-Modify

Port	10
Type	FXS
Register Port	Yes
SIP Account	1004
Password	••••••••
Auto Dial Number	
Wait Time before Auto Dial (s)	0
Echo Canceller	<input checked="" type="checkbox"/> Enable
Forbid Outgoing Call	<input type="checkbox"/> Enable
CID	<input checked="" type="checkbox"/> Enable
Call Waiting	<input type="checkbox"/> Enable
DND (Do Not Disturb)	<input type="checkbox"/> Enable
Call Forward	<input type="checkbox"/> Enable
Advanced Configuration	<input type="checkbox"/> Enable

FXS Settings											
Type	SIP Account	Authentication Username	Auto Dial Num	Forbid Outgoing Call	DND	Forward	FWD Type	FWD Number	CID	Call Waiting	Reg Status
FXS	1003	---	---	Disable	Disable	Disable	---	---	Enable	Disable	Registered
FXS	1004	---	---	Disable	Disable	Disable	---	---	Enable	Disable	Registered
FXS	200	---	---	Disable	Disable	Disable	---	---	Enable	Disable	Unregistered
FXS	201	---	---	Disable	Disable	Disable	---	---	Enable	Disable	Unregistered

7) FXS 9(1003) made a call to FXS 10 (1004).

Channel State									
Channel	Type	Number	Voltage(v)	State	Direction	CallerID	CalleeID	Reg Status	
1	FXO	8001	29		--	--	--	Unregistered	
2	FXO	8002	27		--	--	--	Unregistered	
3	FXO	8003	0		--	--	--	Unregistered	
4	FXO	8004	0		--	--	--	Unregistered	
5	FXO	8005	0		--	--	--	Unregistered	
6	FXO	8006	0		--	--	--	Unregistered	
7	FXO	8007	0		--	--	--	Unregistered	
8	FXO	8008	0		--	--	--	Unregistered	
9	FXS	1003	0		TEL->IP	1003	1004	Registered	
10	FXS	1004	0		IP->TEL	1003	1004	Registered	

8) FXS 9(1003) made a call to Eyebeam(1001)

Channel State									
Channel	Type	Number	Voltage(v)	State	Direction	CallerID	CalleeID	Reg Status	
1	FXO	8001	27		--	--	--	Unregistered	
2	FXO	8002	29		--	--	--	Unregistered	
3	FXO	8003	0		--	--	--	Unregistered	
4	FXO	8004	0		--	--	--	Unregistered	
5	FXO	8005	0		--	--	--	Unregistered	
6	FXO	8006	0		--	--	--	Unregistered	
7	FXO	8007	0		--	--	--	Unregistered	
8	FXO	8008	0		--	--	--	Unregistered	
9	FXS	1003	0		TEL->IP	1003	1001	Registered	
10	FXS	1004	0		--	--	--	Unregistered	

9) Eyebeam(1001) made a call to FXS 10(1004).

Channel State									Channel	Type	N
Channel	Type	Number	Voltage(v)	State	Direction	CallerID	CalleeID	Reg Status			
1	FXO	8001	29		---	---	---	Unregistered	17	---	
2	FXO	8002	27		---	---	---	Unregistered			
3	FXO	8003	0		---	---	---				
4	FXO	8004	0		---	---	---				
5	FXO	8005	0		---	---	---				
6	FXO	8006	0		---	---	---				
7	FXO	8007	0		---	---	---				
8	FXO	8008	0		---	---	---				
9	FXS	1003	0		---	---	---				
10	FXS	1004	0		IP->TEL	1001	1004				
11	FXS	200	0		---	---	---				

10) Eyebeam(1001) made a call to PSTN 088861158.

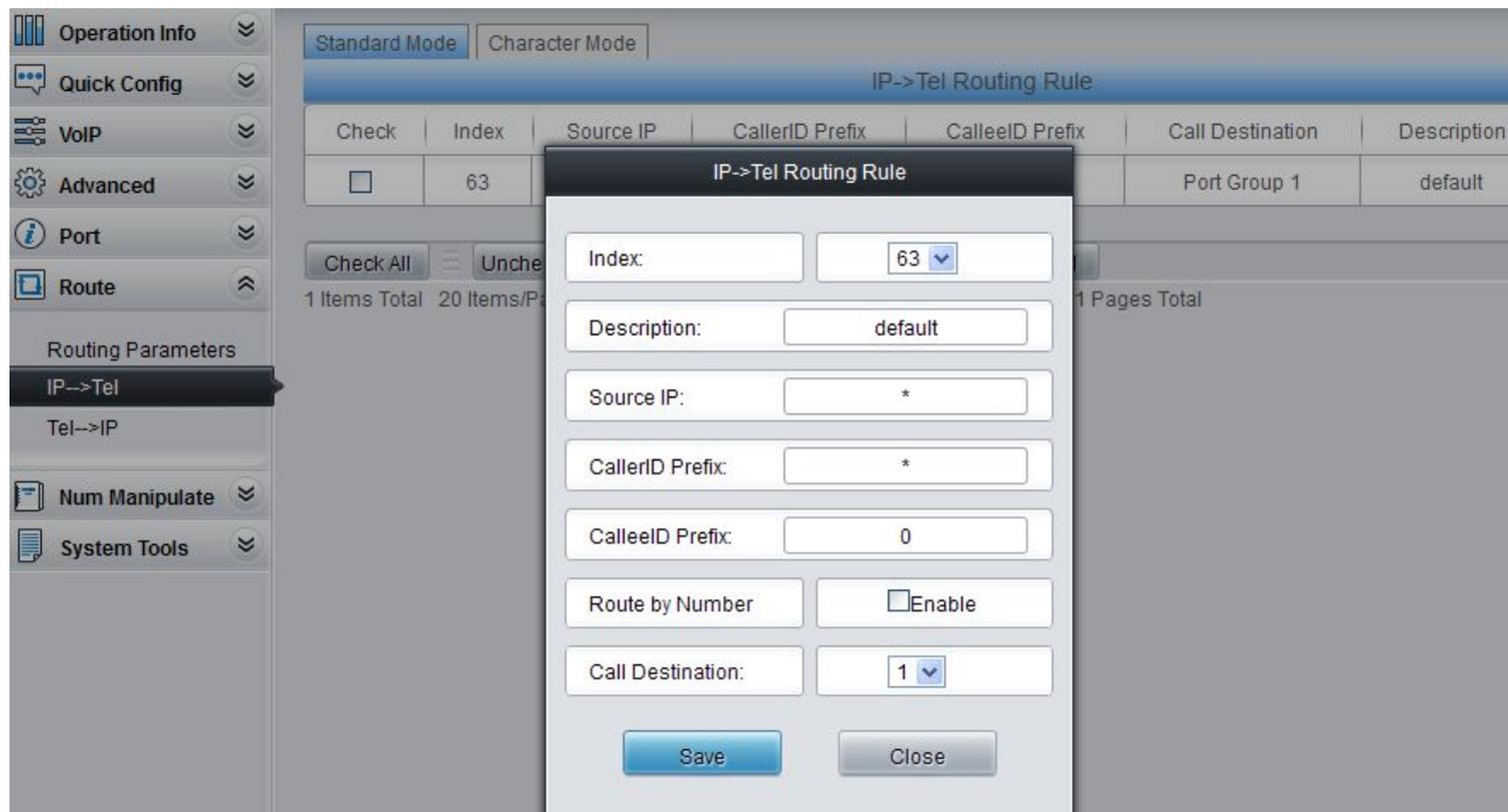
Click on Port\Port Group from the toolbar, add FXO port into Port Group 1.

- Operation Info
- Quick Config
- VoIP
- Advanced
- Port
 - FXS
 - FXO
 - Port Group

Port Group Settings									
Check	Index	Description	SIP Account	Authentication Username	Ports	Port Select Mode	Rule for Ringing by Turns	Timeout for Ringing by	
<input type="checkbox"/>	1	FXO	---	---	1	Increase	---		---

1 Items Total 20 Items/Page 1/1 First Previous Next Last Go to Page 1 1 Pages Total

Create the IP-TEL routing rule, when a call comes in to the gateway on a SIP channel, with callee ID prefix 0, this call will be routed to FXO port, then this FXO port will make an outbound call to PSTN.



Eyebeam(1001) made a call to PSTN 088861158.



11) PSTN 088861158 made a call to Eyebeam(1001).

Create the TEL-IP routing rule, when a call comes in to the gateway on a FXO port, this call will be routed to SIP channel, then the SIP channel will make an outbound call to the Elastix System.

Check	Index	Call Initiator	CallerID Prefix	CalleeID Prefix	Destination IP	Destination Port	Description
<input type="checkbox"/>	63	Port Group 1	*	*	192.168.10.163	5060	default

PSTN 8357 made a call to Eyebeam(1001).

Channel	Type	Number	Voltage(v)	State	Direction	CallerID	CalleeID	Reg Status
1	FXO	8001	7		TEL->IP	8357	1001	Unregistered
2	FXO	8002	0		---			Registered
3	FXO	8003	0					
4	FXO	8004	0					
5	FXO	8005	0					
6	FXO	8006	0					
7	FXO	8007	0					