

Porta  Switch™



PortaSwitch Handbook: SIP Services

Maintenance Release 15

Part I

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V.1.15.2

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1. Setting up Standard SIP Services

PortaBilling Web Reference Guide

Basic SIP service

Checklist

Operation	Done
General configuration	

Network configuration

Rating configuration

Routing

Account provisioning

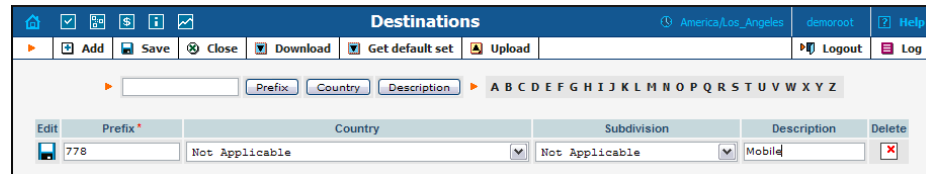
Testing


Creating destinations "one-by-one":

Destinations

 Add

Description



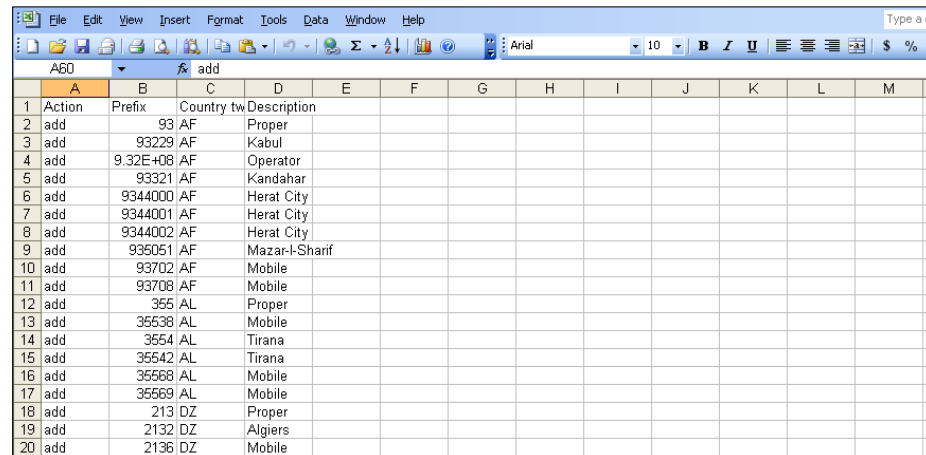
Prefix	Country	Subdivision	Description	Delete
778	Not Applicable	Not Applicable	Mobile	

Save

Uploading a set of destinations from a file:

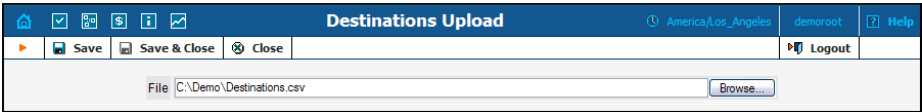
Destinations

Default set



Action	Prefix	Country	Description
add	93	AF	Proper
add	93229	AF	Kabul
add	9.32E+08	AF	Operator
add	93321	AF	Kandahar
add	9344000	AF	Herat City
add	9344001	AF	Herat City
add	9344002	AF	Herat City
add	935051	AF	Mazar-I-Sharif
add	93702	AF	Mobile
add	93708	AF	Mobile
add	355	AL	Proper
add	35538	AL	Mobile
add	3554	AL	Tirana
add	35542	AL	Tirana
add	35568	AL	Mobile
add	35569	AL	Mobile
add	213	DZ	Proper
add	2132	DZ	Algiers
add	2136	DZ	Mobile

Upload



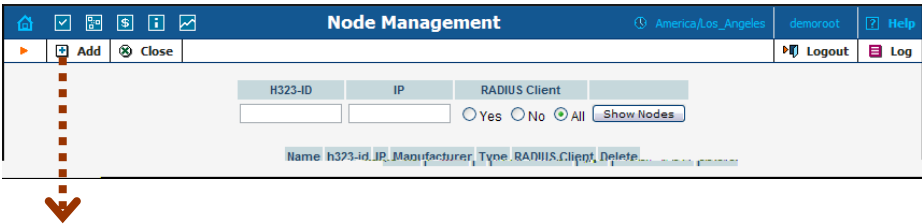
Browse...
Save&Close.

Destinations for SIP phones

1202781

N/A
777
SIP phones
77700001, 7770002, ... 7770999, ...

Create Nodes



The screenshot shows the 'Add Node' web form with the following fields and values:

- Node Name:** DemoSIP
- h323-id:** demosip.mydomain.com
- VoIP Password:** cisco
- NAS-IP-Address:** 207.52.37.45
- Manufacturer:** PortaOne
- Type:** PortaSIP
- RADIUS Client:** ☒
- RADIUS Key:** is6ainob
- RADIUS Source IP:** 207.52.37.45
- RADIUS Dictionary:** Cisco

Below the form, a warning message is displayed:

Submitted information is being cached in the billing engine and will not take effect immediately. Default caching time is 10 minutes. Please contact your system administrator for more information.

Nodes.

Add

- Node name
- H323-ID
hostname.domainname
- VoIP Password
cisco
- NAS-IP-Address
- Auth. Translation rule

- Manufacturer
PortaOne
- Type
PortaSIP
- Radius Client

- Radius Key
key
- Radius Source IP
Node ID, NAS IP address and
PortaBilling Administrator
Guide

Save&Close

NOTE: There is some propagation delay between the database and the Radius server configuration file; however, it is no more than 15 minutes.

Create Tariff

■

■

The screenshot shows the 'Tariff Management' interface. At the top, there's a 'Tariff Management' header with navigation icons and user information. Below it, there's a search bar with filters for 'Applied To', 'Service Type', and 'Managed By'. A table lists various tariffs with columns: Name, Currency, Applied To, Service Type, Managed By, Routing, Description, Rates, and Delete. An arrow points from the 'Add' button in the top left to the 'Add Tariff' form below. The 'Add Tariff' form has a 'General Info' tab. Fields include: Name (SIP Phone Subscribers), Currency (USD - US Dollar), Applied To (Customer), Service Type (Voice Calls), Managed By (Administrator only), Off-Peak Period (startstop:hr(20-5)), Off-Peak Description (PERIOD: From 20:00 until 06:00), Destination Group Set, Free Seconds (0), Post Call Surcharge (0 %), Login Fee (0), Connect Fee (0), Round Charged Amount (XXXXXX.XX000), Default Formula, Short Description (SIP Phone Subscriber's Tariff), and Description (This tariff applies to all SIP phone subscribers).

Tariffs

Add

Add Tariff

- Name
- Currency

NOTE: The currency for the tariff may be chosen only once, and cannot be changed later.

- Applied To

Customer

- | Vendor | Routing |
|-------------------------------------------------------------------------------------------------------------------|--------------------------------------------------------------------------|
| <ul style="list-style-type: none"> ○ Managed By ○ Service Type ○ Off-peak Period | <p>Administrator Only</p> <p>Applied to: Customer</p> <p>Voice Calls</p> |



Help

- Off-peak Description
- Destination Group Set
- Free Seconds
- Post-Call Surcharge
- Login Fee
- Connect Fee
- Round Charged Amount
- Default Formula
- Short Description
- Description
 - Save

Vendor Applied To
Routing

Enter Rates

Call Billing Parameters

PortaBilling Administrator Guide

Managing rates online

'SIP Phone Subscribers' tariff rates

Effective From: Now Destination: [Country] Prefix Group Country

Edit	Destination *	Country	Description	Interval, second	Price, USD / minute	Effective From				Delete
				First *	Next *	First *	Next *			
			Peak							
			Off-Peak							

'SIP Phone Subscribers' tariff rates

Effective From: Now Destination: [Country] Prefix Group Country

Edit	Destination *	Country	Description	Interval, second	Price, USD / minute	Effective From				Delete
				First *	Next *	First *	Next *			
	420		Peak	1	1	0.12	0.12	immediately		
			Off-Peak	1	1	0.11	0.11			



Rates

Rates

Edit Rates



Add

- **Destination**
420


Destination

NOTE: The phone prefix you are trying to create a rate for must already exist in Destinations.



- **Interval First**
- **Interval Next**
- **Price First**
- **Price Next**

- Off-peak Interval First
- Off-peak Interval Next
- Off-peak Price First
- Off-peak Price Next

NOTE: Off-peak fields appear only if an **off-peak period** has been defined for the tariff.

- Rate Formula Wizard 
- Effective from

NOTE: When using the calendar, you can specify that the date you are entering is in a different time zone than your present one. PortaBilling will then automatically adjust the time.

- Hidden Forbidden Discontinued
 Save 

Tariffs with routing extensions

- Route category

Default

- Preference

- Huntstop

Managing rates offline

NOTE: Templates are available in PortaBilling, a powerful tool for uploading rates from custom format data files. However, in this particular example we assume that you will enter data using the PortaBilling default format.

Tariffs

Download

Now

File download

File Edit View Insert Format Tools Data Window Help																
	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P
1	Name	Currency	Description	Short Desc	Off-Peak	Description										
2	SIP Phone	USD	This tariff is SIP phone PERIOD: From 20:00 until 06:00													
3																
4	Off-peak P	Destination	Free Seco	Post Call	Login Fee	Connect Fee										
5	startstop:hr(20-5)		0	0	0	0										
6																
7	Destination	Destination	Country	Description	First Interv	Next Interv	First Price	Next Price	Off-peak F	Off-peak N	Off-peak F	Off-peak N	Forbi	Hide	Discontin	Effective For
8	420		CZECH RE	Proper	30	6	0.15	0.15	30	6	0.1	0.1				5/11/2004 3
9	420602		CZECH RE	Mobile	30	6	0.17	0.17	30	6	0.15	0.15				5/11/2004 3
10																
11																

Country Description

Create destinations

Effective from

	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P
3																
4	Off-peak P	Destination	Free Seco	Post Call	Login Fee	Connect Fee										
5	startstop: hr(20-5)		0	0	0	0										
6																
7	Destination	Destination Country	Descriptor	First Interv	Next Interv	First Price	Next Price	Off-peak F	Off-peak N	Off-peak F	Off-peak N	Forbi	Hide	Discontin	Effective	For
8	420	CZECH RE	Proper	30	6	0.15	0.15	30	6	0.1	0.1					
9	420602	CZECH RE	Mobile	30	6	0.17	0.17	30	6	0.15	0.15					
10	420601			30	6	0.17	0.17	30	6	0.15	0.15					
11																



may contain features that are not compatible with CSV (Comma delimited)
Yes

Do you want to save the changes you made

Tariff

Upload

Browse...

Save&Close

Tariff

Edit Rates

Main menu

Home

Create All Required Tariffs



Create Tariff Enter Rates

•

•

•

Managed by NNN

NNN

Create Product

Product Management America/Los_Angeles demootool Help

Add Close

Managed By ANY Search

Search

\$?	Name	Currency	Managed By	Description	Delete
-----	------	----------	------------	-------------	--------

Add Product America/Los_Angeles demootool Help

Save Close

Product Name SIP Subscribers Currency USD - US Dollar

Managed By Administrator only

General Info Maintenance Fee

Breakage 0

Account Default ACL Account self-care

Default Discount Plan None

Info URL

Description Product for SIP users. No monthly fee.

Edit 'SIP Subscribers' Product America/Vancouver demootool Help

Add Save Save & Close Close Rate Lookup Logout Log

Product Name SIP Subscribers Currency USD

Managed By Administrator only

General Info Maintenance Fee Online Signup Accessibility Subscriptions Notepad

Service Type *	Node	Access Code	Info Digits	Tariff *	Delete
Voice Calls	DemoSIP		ANY	SIP Phone Subscrib	X

Edit 'SIP Subscribers' Product America/Vancouver demootool Help

Add Save Save & Close Close Rate Lookup Logout Log

Product Name SIP Subscribers Currency USD

Managed By Administrator only

General Info Maintenance Fee Online Signup Accessibility Subscriptions Notepad

Edit	Service Type *	Node	Access Code	Info Digits	Tariff *	Delete
	NOT SELECTED	ANY		ANY		
	Voice Calls	DemoSIP			SIP Phone Subscribers	X

Products

+ Add

- Product name

- Currency
- Managed by
- Breakage
 - None
- Account Default ACL
- Default Discount Plan
 - None
- Info URL
 - http://www.myproduct.com
- Description
 - Save Accessibility

Enter Node and Tariff into the product's accessibility list

Voice Calls	Service Type	+ Add
Access code	Info Digits	
Save		

Create Vendors

Vendors

+ Add

Add Vendor (America/Los_Angeles) demo-root Help

Save Save & Close Close Logout

Vendor Name: GlobalNet Currency: USD - US Dollar Opening Balance: 0

Address Info Additional Info User Interface

Company Name: GlobalNet Contact:
 Mr./Ms./...:
 First Name: M.I.:
 Last Name:
 Address: 123 Main Street
 Province/State: CA
 Postal Code: 54321
 City: Los Angeles
 Country/Region: USA

Phone:
 Fax:
 Alt. Phone:
 Alt. Contact:
 E-mail: info@globalnet.com
 BCC:
 Send Statistics: Full Statistics
 Description:

Edit Vendor (America/Vancouver) demo-root Help

Save Save & Close Close xDRs Connections Logout Log

Vendor Name: GlobalNet Opening Balance: 0.00000 USD
 Balance: 0.00000 USD

Address Info Maintenance Additional Info User Interface Accounts Notepad

Login: globalnet Time Zone: America/Los_Angeles
 Password: ob5risig Auto Web Interface Language: en - English
 Access Level: Vendor

Output Format

Date: YYYY-MM-DD 2003-12-31
 Time: HH24:MI:SS User Defined
 Date & Time: YYYY-MM-DD HH24:MI:SS User Defined

Input Format

Date: YYYY-MM-DD 2003-12-31
 Time: HH24:MI:SS User Defined

Add Vendor

Main form (top)

- Vendor name
- Currency
- Opening Balance

Additional info:

- Billing Period

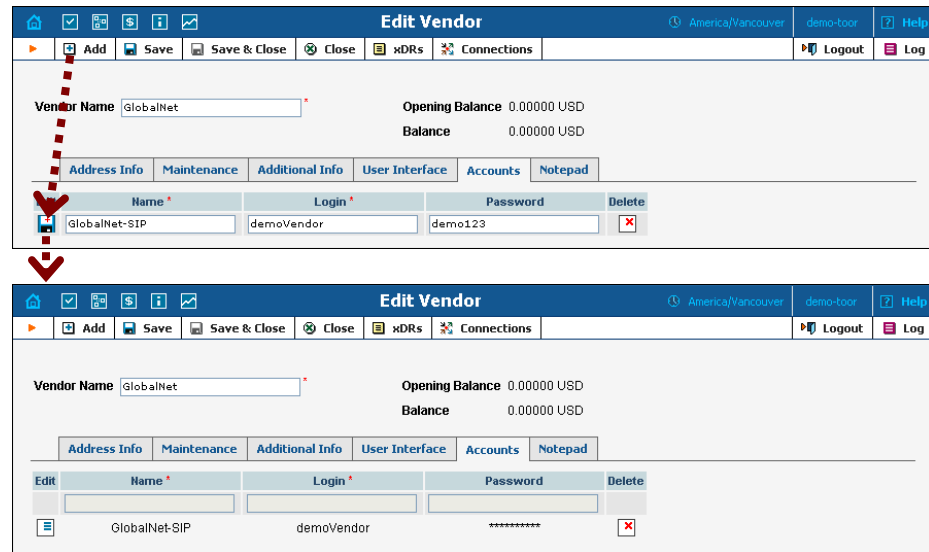
User Interface:

- o Time Zone

 Save

Vendor

account




Edit Vendor America/Vancouver demo-tool Help

Add Save Save & Close Close xDRs Connections Logout Log

Vendor Name GlobalNet Opening Balance 0.00000 USD
Balance 0.00000 USD

Address Info Maintenance Additional Info User Interface Accounts Notepad


Name *	Login *	Password	Delete
GlobalNet-SIP	demoVendor	demo123	

Edit Vendor America/Vancouver demo-tool Help

Add Save Save & Close Close xDRs Connections Logout Log

Vendor Name GlobalNet Opening Balance 0.00000 USD
Balance 0.00000 USD

Address Info Maintenance Additional Info User Interface Accounts Notepad

Name *	Login *	Password	Delete
GlobalNet-SIP	demoVendor	*****	

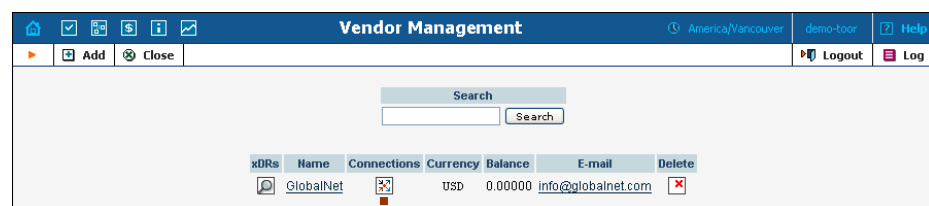
Close

Vendors

Define Connections

Vendors




Connections



Vendor Management America/Vancouver demo-tool Help

Add Close

Search

xDRs	Name	Connections	Currency	Balance	E-mail	Delete
	GlobalNet		USD	0.00000	info@globalnet.com	

Vendor 'GlobalNet' connections

America/Los_Angeles

demo-toor

Help

</

- Name
- Invoice Template

Do not create invoice

- Terms

- [PortaBilling Web Reference Guide](#)

- Description






 Save&Close

Create a Customer

Customers

 Add

New Customer

Add Customer

America/Los_Angeles
demo root
Help

Save
Save & Close
Close

Customer Name
EasyCall Ltd.
Currency
USD - US Dollar
Blocked
☐
Opening Balance
0
Type
Retail
Customer Class
Retail

Address Info
Additional Info
Payment Info
User Interface
Call Features

Company Name
Mr./Ms./...
First Name
M.I.
Last Name
Address
11 Broadway, Suite #444
Province/State
NY
Postal Code
01802
City
New York
Country/Region

Contact
Phone
Fax
Alt. Phone
Alt. Contact
E-mail
admin@easycall.com
BCC
Send Statistics
Full Statistics
Description

Main form (top)

- Name
- Currency
- Opening Balance
- Type
- Customer Class

Address info tab

- Email
- Bcc
- Send Statistics Summary only

Full statistics
Do not send

Additional info tab

- Billing period

PortaBilling Administrator Guide

Payment info tab

- Credit Limit
- Balance Warning Threshold

-

-

User Interface tab

- **Time Zone**
 - America/New_York**
 - Monthly**
- **Web Interface Language**

Call Features tab

- **Limit simultaneous calls**
- **VoiceVPN**
- **First login greeting**
- **Dialing rules**

 **Save&Close**

Create Accounts

Customers

The first screenshot shows the 'Customer Management' page. It has a top navigation bar with 'America/Vancouver', 'demo-root', and 'Help'. Below the navigation bar are 'Add' and 'Close' buttons. The main content area has a search section with 'Type' (Direct Customers), 'Representative' (ANY), and a 'Search' button. Below this is a table with columns: xDRs, Name, Accounts / Subcustomers, Currency, Type, Credit limit, Balance, E-mail, Status, and Delete. The table contains one row for 'EasyCall Ltd.' with a credit limit of 0.00000 and an email of admin@easycall.com.

The second screenshot shows the 'Accounts of Retail Customer 'EasyCall Ltd.' page. It has a top navigation bar with 'America/Los_Angeles', 'demo-root', and 'Help'. Below the navigation bar are 'Add', 'Account Generator', and 'Close' buttons. The main content area has a search section with 'Account ID', 'Batch' (ANY), 'Ctrl #', and 'SIP Status' (ANY). There is a 'Show Accounts' button and an 'Advanced search' link.

The third screenshot shows the 'Add Account for Retail Customer 'EasyCall Ltd.' page. It has a top navigation bar with 'America/Los_Angeles', 'demo-root', and 'Help'. Below the navigation bar are 'Save', 'Save & Close', and 'Close' buttons. The main content area has input fields for 'Account ID' (16041234567), 'Product' (USD - SIP Subscribers), 'Blocked' (checkbox), and 'Opening Balance' (10). There are tabs for 'Account Info', 'Subscriber', 'Additional Info', 'Life Cycle', 'User Interface', and 'Call Features'. The 'Account Info' tab is active, showing 'Type' (Debit, Credit, Voucher), 'VoIP Password' (scho7rtw), 'E-mail', and 'Batch' (easycall).

Accounts

Add

- Account ID
- Product
- Blocked
- Opening Balance

Account Info tab

- Type
- Credit limit
- VoIP Password

- **Email**

- **Batch**

Additional Info tab

- **IP Phone**
- **IP Phone Port**

Life Cycle tab


- **Activation date**
- **Expiration date**
- **Lifetime**

User Interface tab

- **Login**
- **Password**
- **Time Zone**
- **Web Interface Language**

Call Features tab

- **Preferred IVR Language**

- Associated Number
 - UM Enabled
 - Forward Mode
 -
 -
 -
 -
 -
- Forward
Follow-me
-  **Save&Close**

Set up Dialing Rules for the Customer
(optional)

Edit Customer America/Vancouver demo-tool Help

Save Save & Close Close xDRs Accounts E-Payments Log Logout Log

Customer Name EasyCall Ltd. **Opening Balance** 0.00000 USD
Blocked ☐ **Balance** 0.00000 USD
Type Retail **Customer Class** Enterprise

Address Info **Maintenance** **Additional Info** **Payment Info** **User Interface** **Periodic Payments** **Call Features** **Abbreviated Dialing** **Volume Discounts** **Subscrip**

Limit simultaneous calls: No
Set CLI to Account ID: No
[Set CLI To Centrax](#)
Hide CLI: No
Call Parking: No
Voice VPN: No
Voice VPN Distinctive Ring: No
Legal Intercept: No
First Login Greeting: No
Music on Hold: No Frills Cumbia (c) 2001 Kevin ?

Enable Dialing Rules **Dialing rules wizard**

Load Sample:
Your country code:
Your area code(s):
Always dial the area code as a part of the number:
Prefix for accessing the outside phone network:
Prefix for domestic calls, but outside of your area code (e.g. 1, 0):
International dialing prefix (e.g. 011, 00, 0011):
Emergency numbers (e.g. 911, 112):
Exceptions (e.g. *98):
Local dialing number length:
Convert ANI (CLI) for incoming calls into this dialing format:

Dialing rules wizard Help

Save Save & Close Close

Your country code: 1
Your area code(s): 604, 778
Always dial the area code as a part of the number: ☒
Prefix for accessing the outside phone network: 9
Prefix for domestic calls, but outside of your area code (e.g. 1, 0): 0
International dialing prefix (e.g. 011, 00, 0011): 00
Emergency numbers (e.g. 911, 112): 911
Exceptions (e.g. *98): 411
Local dialing number length:
Convert ANI (CLI) for incoming calls into this dialing format: ☐

Sample settings

1.164
North America, WA, 7 digit dialing
North America, BC, 10 digit dialing
Europe, Czech Rep., always dial using the areacode
Europe, Czech Rep., local and domestic dialing (obsolete)
Australia, Sydney

Clear Reset Load Sample

Check yourself

To call **604 1234567** outside of your office, but within the same area you dial: **9 604 1234567**
To call long distance **425 1234567** (within your country) you dial: **9 8 425 1234567**
To call **420 2 12345678** internationally you dial: **9 00 420 2 12345678**

Set up Abbreviated Dialing for the Customer (optional)

Abbreviated Number Length

Save
Add

To Dial

NOTE: If you enter an off-net PSTN number in **# To Dial**; it must be in the E.164 format, i.e. you cannot enter the number in the customer's dialing format.

 Save



Edit Customer America/Vancouver demo-tool Help

[Add](#) [Save](#) [Save & Close](#) [Close](#) [xDRs](#) [Accounts](#) [E-Payments Log](#) [Logout](#) [Log](#)

Customer Name: EasyCall Ltd. Opening Balance: 0.00000 USD
 Blocked: ☐ Balance: 0.00000 USD
 Type: Retail Customer Class: Enterprise

[Address Info](#) [Maintenance](#) [Additional Info](#) [Payment Info](#) [User Interface](#) [Periodic Payments](#) [Call Features](#) [Abbreviated Dialing](#) [Volume Discounts](#) [Subscrip](#)

Abbreviated Number Length: 3

Abbreviated #	# To Dial	Description	SIP	Delete
102				
123	16049994321	Joe Brown		

Select Account [Close](#)

Account ID	Batch	Ctrl #
	Unknown Batch	

[Show Accounts](#)

Account ID	Batch	Status	SIP
16041234567	easycall		

Testing the Whole System

○

Accounts of Retail Customer 'EasyCall Ltd.' America/Vancouver demo-tool Help

[Add](#) [Account Generator](#) [Close](#) [Logout](#)

Account ID	Batch	Ctrl #	SIP Status	Advanced search
	ANY		ANY	Show Accounts

xDRs	Account ID	Idle, days	Currency	Balance	Credit Limit	Type	Product	Batch	Status	SIP
	16041234567		USD	10.00000		Debit	SIP Subscribers	easycall		
	16041234568		USD	0.00000	20.00000	Credit	SIP Subscribers	easycall		
	16041234569		USD	10.00000		Debit	SIP Subscribers	easycall		
	16041234570		USD	0.00000	10.00000	Credit	SIP Subscribers	easycall		

○ User Agent Contact

Account Info / Retail Customer 'EasyCall Ltd.'		America/Vancouver	demo-tool	Help
Save	Save & Close	Close	xDRs	Rate Lookup
E-Payments Log	Logout	Log		
Account ID	16041234567	Product	USD - SIP Subscribers	
Blocked	<input type="checkbox"/>	Balance	10.00000 USD	
User Agent	Sipura-2000	Contact	sip:16041234567@192.168.0.111:5060	
Account Info	Maintenance	Subscriber	Additional Info	Life Cycle
User Interface	Call Features	Subscriptions	Notepad	
Customer	EasyCall Ltd.		Opening Balance	10.00000 USD
Type	Debit		Refunds	0 USD
VoIP Password	cheeha3t	Auto	Non call related charges	0 USD
E-mail				
Batch	easycall			
Control number	7			

Check the Log Files

```

/var/log/sipenv-
<sipserverIP>/log/sip.log
less more

```

Using the SIP log viewer



SIP Log Viewer

PortaSIP node

Generate

SIP Log Viewer America/Los_Angeles demoroot Help

Logout Close

Node: 193.28.87.106 - PortaSIP

Trace a call

Call-ID: **but not H323-Conf-ID!**

Another Call-ID: optional, e.g. for callback calls

Search by call ID: --SELECT--

☐ Include log file

☒ With extension

☐ Log with call data

For last: 5 minutes

Filter: ☒ do not show selftest calls

Generate

View log

SIP Log Viewer America/Los_Angeles demoroot Help

Close Logout

List of call attempts in current sip.log file on 193.28.87.106 PortaSIP node since 11 Jun 22:08:28 EEST (UTC +0300):

Setup Time (in SIP server TZ)	Caller's IP:UdpPort	CLI	CLD	Call-ID	Caller's User Agent
11 Jun 22:09:43	216.231.44.168:9062	16041234568	6831234	98dda488-69c74dd3@192.168.0.250	Sipura/SPA2000-3.1.5

siplogview version: 1.29, experimental mode.

PortaSIP node: 193.28.87.106

Call-ID: 98dda488-69c74dd3@192.168.0.250

H323-Conf-ID: [A9B6252E](#) [3863D707](#) [17A91CCD](#) [744348CB](#)

PortaSIP	UA	ser	b2bua	AAA	UA
server	216.231.44.168	193.28.87.106	193.28.87.106	PortaBilling	193.28.187.3
time	Sipura/SPA2000-3.1.5	PortaSIP	PortaSIP		
11 Jun					
22:09:43	0-> (A? 101/I) * INVITE ----->				
22:09:43	<- (A? 101/I) 100 trying - -->				
22:09:43		0-> (A? 101/I) * INVITE ----->			
22:09:43		<- (A? 101/I) 401 Unauthorized -->			
22:09:43	<- (A? 101/I) 401 Unauthorized -->				
22:09:43		0-> (A? 101/A) ACK ----->			
22:09:43		<- (A? 101/A) ACK ----->			
22:09:43	0-> (A? 102/I) * INVITE ----->				
22:09:43	<- (A? 102/I) 100 trying - -->				
22:09:43		0-> (A? 102/I) * INVITE ----->			
22:09:43		<- (A? 102/I) 100 Trying ----->			
22:09:43			0-> Authorization request ---->		
22:09:43		<- (A? 102/I) 100 Trying ----->			
22:09:44			<- Auth request accepted ----->		

11 Jun 22:09:43/GLOBAL/ser[98210]: RECEIVED message from 216.231.44.168:9062:

```

INVITE sip:6831234@193.28.87.106 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-1035d24f
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 INVITE
Max-Forwards: 70
Contact: John Doe <sip:16041234568@192.168.0.250:9062>
Expires: 240
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 428
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
  
```

SIP user agent attempts to make a call via the SIP server.

```
11 Jun 22:09:43/GLOBAL/ser[98218]: RECEIVED message from 216.231.44.168:9062:
INVITE sip:6831234@193.28.87.106 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-1035d24f
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 INVITE
Max-Forwards: 70
Contact: John Doe <sip:16041234568@192.168.0.250:9062>
Expires: 240
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 428
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura
Content-Type: application/sdp

v=0
o=- 16430652 16430652 IN IP4 192.168.0.250
s=-
c=IN IP4 192.168.0.250
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv

10 Jun 01:04:59/9154867a-2393e376@192.168.0.250/ser[53367]: processing INVITE
received from 216.231.44.168
```

SIP user agent is informed that his request is being processed.

```
11 Jun 22:09:43/GLOBAL/ser[98218]: SENDING message to 216.231.44.168:9062:
SIP/2.0 100 trying -- your call is important to us
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-
1035d24f;rport=9062;received=216.231.44.168
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 INVITE
Server: Sip EXpress router (0.9.4 (i386/freebsd))
Content-Length: 0
```

Request is sent to B2BUA.

```
11 Jun 22:09:43/GLOBAL/ser[98218]: SENDING message to 193.28.87.106:5061:
INVITE sip:6831234@193.28.87.106:5061 SIP/2.0
Record-Route: <sip:193.28.87.106;ftag=873d0427882f87o0;lr>
Via: SIP/2.0/UDP
193.28.87.106;branch=z9hG4bK52d.65c85fbb7b48dc83837bf35f80ab19f5.0
Via: SIP/2.0/UDP
192.168.0.250:9062;rport=9062;received=216.231.44.168;branch=z9hG4bK-1035d24f
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 INVITE
```



```
Max-Forwards: 16
Contact: John Doe <sip:16041234568@216.231.44.168:9062>
Expires: 240
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 477
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura
Content-Type: application/sdp
PortaBilling-Notify: NAT
```

```
v=0
o=- 16430652 16430652 IN IP4 192.168.0.250
s=-
c=IN IP4 216.231.44.168
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250
```

B2BUA receives this INVITE request.

```
11 Jun 22:09:43/GLOBAL/b2bua: RECEIVED message from 193.28.87.106:5060:
INVITE sip:6831234@193.28.87.106:5061 SIP/2.0
Record-Route: <sip:193.28.87.106;ftag=873d0427882f87o0;lr>
Via: SIP/2.0/UDP
193.28.87.106;branch=z9hG4bK52d.65c85fbb7b48dc83837bf35f80ab19f5.0
Via: SIP/2.0/UDP
192.168.0.250:9062;rport=9062;received=216.231.44.168;branch=z9hG4bK-1035d24f
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 INVITE
Max-Forwards: 16
Contact: John Doe <sip:16041234568@216.231.44.168:9062>
Expires: 240
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 477
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura
Content-Type: application/sdp
PortaBilling-Notify: NAT
```

```
v=0
o=- 16430652 16430652 IN IP4 192.168.0.250
s=-
c=IN IP4 216.231.44.168
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
```

```
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250
```

In the rest of the log example, we will skip the request transmission between SER and B2BUA (request received by SER, request sent to B2BUA, request received by B2BUA) since this would only duplicate the same information.

```
11 Jun 22:09:43/GLOBAL/ser[]: SENDING message to 216.231.44.168:9062:
```

PortaSIP requests digest authentication from the SIP UA, providing a challenge.

```
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP
192.168.0.250:9062;received=216.231.44.168;rport=9062;branch=z9hG4bK-1035d24f
Record-Route: <sip:193.28.87.106;ftag=873d0427882f87o0;lr>
From: "John Doe" <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 INVITE
Server: Sippy
WWW-Authenticate: Digest
realm="193.28.87.106",nonce="3051864d7d0c6578460cc4b0e28ad43b448c6a77"
```

```
11 Jun 22:09:43/GLOBAL/ser[98219]: RECEIVED message from 216.231.44.168:9062:
```

SIP UA acknowledges that it has received an authorization request (ACKs will be skipped in the rest of the document).

```
ACK sip:6831234@193.28.87.106 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-1035d24f
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 101 ACK
Max-Forwards: 70
Contact: John Doe <sip:16041234568@192.168.0.250:9062>
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 0
```

SER receives a reply to the authorization request, with a response to the challenge.

```
11 Jun 22:09:43/GLOBAL/ser[98219]: RECEIVED message from 216.231.44.168:9062:
INVITE sip:6831234@193.28.87.106 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-1628d42b
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 102 INVITE
Max-Forwards: 70
Authorization: Digest
username="16041234568",realm="193.28.87.106",nonce="3051864d7d0c6578460cc4b0e28ad4
3b448c6a77",uri="sip:
6831234@193.28.87.106",algorithm=MD5,response="54e0b42337ace33edf36d004f1037ebd"
Contact: John Doe <sip:16041234568@192.168.0.250:9062>
Expires: 240
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 428
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura
Content-Type: application/sdp
```

```
v=0
```

```

o=- 16430652 16430652 IN IP4 192.168.0.250
s=-
c=IN IP4 192.168.0.250
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv

11 Jun 22:09:43/98dda488-69c74dd3@192.168.0.250/ser[98219]: processing INVITE
received from 216.231.44.168
11 Jun 22:09:43/GLOBAL/ser[98219]: SENDING message to 216.231.44.168:9062:

```

SIP UA is informed that the request has been received and is being processed (100 Trying responses will be omitted in the rest of the document).

```

SIP/2.0 100 trying -- your call is important to us
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-
1628d42b;rport=9062;received=216.231.44.168
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 102 INVITE
Server: Sip EXpress router (0.9.4 (i386/freebsd))
Content-Length: 0

```

This request is resent to B2BUA with several modifications (in particular, a PortaBilling-Notify:NAT flag is added to inform B2BUA of the NAT status of the device).

```

11 Jun 22:09:43/GLOBAL/ser[98219]: SENDING message to 193.28.87.106:5061:
INVITE sip:6831234@193.28.87.106:5061 SIP/2.0
Record-Route: <sip:193.28.87.106;ftag=873d0427882f87o0;lr>
Via: SIP/2.0/UDP
193.28.87.106;branch=z9hG4bK22d.30fef1504a66ee2bd3c0b9cfa4e4e09b.0
Via: SIP/2.0/UDP
192.168.0.250:9062;rport=9062;received=216.231.44.168;branch=z9hG4bK-1628d42b
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 102 INVITE
Max-Forwards: 16
Authorization: Digest
username="16041234568",realm="193.28.87.106",nonce="3051864d7d0c6578460cc4b0e28ad4
3b448c6a77",uri="sip:
6831234@193.28.87.106",algorithm=MD5,response="54e0b42337ace33edf36d004f1037ebd"
Contact: John Doe <sip:16041234568@216.231.44.168:9062>
Expires: 240
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 477
Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, REFER
Supported: x-sipura
Content-Type: application/sdp
PortaBilling-Notify: NAT

v=0
o=- 16430652 16430652 IN IP4 192.168.0.250
s=-
c=IN IP4 216.231.44.168

```

```
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250
```

B2BUA sends an authorization request to the billing.

```
11 Jun 22:09:43/98dda488-69c74dd3@192.168.0.250/b2bua: sending AAA request:
User-Name = '16041234568'
Digest-Realm = '193.28.87.106'
Digest-Nonce = '3051864d7d0c6578460cc4b0e28ad43b448c6a77'
Digest-Method = 'INVITE'
Digest-URI = 'sip:6831234@193.28.87.106'
Digest-Algorithm = 'MD5'
Digest-User-Name = '16041234568'
Digest-Response = '54e0b42337ace33edf36d004f1037ebd'
Calling-Station-Id = '16041234568'
Called-Station-Id = '6831234'
h323-conf-id = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id = '98dda488-69c74dd3@192.168.0.250'
h323-remote-address = '216.231.44.168'
h323-session-protocol = 'sipv2'
h323-ivr-out = 'PortaBilling_Routing:SIP'
h323-ivr-out = 'PortaBilling_AuthMethod:INVITE'
h323-ivr-out = 'PortaBilling_Notify:NAT'
h323-ivr-out = 'PortaBilling_Seed:144514807'
```

```
11 Jun 22:09:44/98dda488-69c74dd3@192.168.0.250/b2bua: AAA request accepted,
processing response:
```

Billing authorizes the call and provides information about call routing (5 possible routes are returned).

```
Cisco-AVPair = 'h323-ivr-in=PortaBilling_Routing:@;g-
hunt=seq;expires=300;credit-time=29460;patience=20'
Cisco-AVPair = 'h323-ivr-in=PortaBilling_Routing: 16046831234@193.28.187.3'
Cisco-AVPair = 'h323-ivr-in=PortaBilling_Routing:
16046831234@70.68.128.186;auth=ipcall-test:test123'
Cisco-AVPair = 'h323-ivr-in=PortaBilling_Routing:
16046831234@69.104.30.123;auth=PortaSoftware:PortaSoftware;rtp=1'
Cisco-AVPair = 'h323-ivr-in=PortaBilling_Routing: 16046831234@192.168.0.66'
Cisco-AVPair = 'h323-ivr-in=PortaBilling_Routing: 16046831234@192.168.0.30'
h323-billing-model = 'h323-billing-model=0'
Cisco-AVPair = 'h323-ivr-in=Tariff:SIP Phone Subscribers'
Cisco-AVPair = 'h323-ivr-in=PortaBilling_CLI:16041234568'
Cisco-AVPair = 'h323-ivr-in=MOH:1'
Cisco-AVPair = 'h323-ivr-in=PortaBilling_CompleteNumber:16046831234'
Cisco-AVPair = 'h323-ivr-in=DURATION:29460'
h323-return-code = 'h323-return-code=0'
h323-currency = 'h323-currency=USD'
h323-credit-time = 'h323-credit-time=29460'
h323-preferred-lang = 'h323-preferred-lang=en'
```

INVITE is sent to the first gateway/proxy in the route list.

```
11 Jun 22:09:44/GLOBAL/b2bua: SENDING message to 193.28.187.3:5060:
INVITE sip:16046831234@193.28.187.3:5060 SIP/2.0
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK1343caf4b64b53d7a6d0b68a51b554aa;rport
Max-Forwards: 70
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=52d266cda37f42d0c24541d5190acf71
To: <sip:16046831234@193.28.187.3>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 200 INVITE
Contact: Anonymous <sip:193.28.87.106:5061>
Expires: 300
User-Agent: Sippy
cisco-GUID: 2847286574-946067207-396958912-1950566603
h323-conf-id: 2847286574-946067207-396958912-1950566603
Content-Length: 475
Content-Type: application/sdp

v=0
o=Sippy 137112044 0 IN IP4 193.28.87.106
s=-
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
c=IN IP4 216.231.44.168
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250
```

No reply is received from this gateway/proxy, so PortaSIP re-sends the invite message several times.

```
11 Jun 22:10:00/GLOBAL/b2bua: SENDING message to 193.28.187.3:5060:
INVITE sip:16046831234@193.28.187.3:5060 SIP/2.0
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK1343caf4b64b53d7a6d0b68a51b554aa;rport
Max-Forwards: 70
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=52d266cda37f42d0c24541d5190acf71
To: <sip:16046831234@193.28.187.3>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 200 INVITE
Contact: Anonymous <sip:193.28.87.106:5061>
Expires: 300
User-Agent: Sippy
cisco-GUID: 2847286574-946067207-396958912-1950566603
h323-conf-id: 2847286574-946067207-396958912-1950566603
Content-Length: 475
Content-Type: application/sdp

v=0
o=Sippy 137112044 0 IN IP4 193.28.87.106
s=-
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
c=IN IP4 216.231.44.168
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
```

```

a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250

```

Finally, PortaSIP decides that this route is non-functioning, and a failed accounting record is sent to the billing.

```

11 Jun 22:10:04/98dda488-69c74dd3@192.168.0.250/b2bua: sending Acct Stop
(Originate):
h323-call-origin      = 'originate'
h323-call-type        = 'VoIP'
h323-session-protocol = 'sipv2'
h323-setup-time       = '19:09:44.000 GMT Sun Jun 11 2006'
User-Name              = '16041234568'
Calling-Station-Id     = '16041234568'
Called-Station-Id      = '16046831234'
h323-conf-id          = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id                = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id        = '98dda488-69c74dd3@192.168.0.250'
h323-remote-address    = '193.28.187.3'
h323-ivr-out           = 'DURATION:7200'
h323-ivr-out           = 'PortaBilling_Seed:144514807'
h323-disconnect-time   = '19:10:04.000 GMT Sun Jun 11 2006'
h323-connect-time      = '19:10:04.000 GMT Sun Jun 11 2006'
Acct-Session-Time      = '0'
h323-disconnect-cause  = '10'
Acct-Status-Type       = 'Stop'

```

After that, PortaSIP tries the next route in the list.

```

11 Jun 22:10:04/GLOBAL/b2bua: SENDING message to 70.68.128.186:5060:
INVITE sip:16046831234@70.68.128.186:5060 SIP/2.0
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bKade4b51964bd86e3026170c62ce471c6;rport
Max-Forwards: 70
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 200 INVITE
Contact: Anonymous <sip:193.28.87.106:5061>
Expires: 300
User-Agent: Sippy
cisco-GUID: 2847286574-946067207-396958912-1950566603
h323-conf-id: 2847286574-946067207-396958912-1950566603
Content-Length: 475
Content-Type: application/sdp

v=0
o=Sippy 137183084 0 IN IP4 193.28.87.106
s=-
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
c=IN IP4 216.231.44.168
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000

```

```
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=ptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250
```

This gateway is available, so we get a reply stating that it has started to process the call.

```
11 Jun 22:10:04/GLOBAL/b2bua: RECEIVED message from 70.68.128.186:5060:
SIP/2.0 100 trying -- your call is important to us
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bKade4b51964bd86e3026170c62ce471c6;rport=5061
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 200 INVITE
Server: Sip EXpress router (0.9.4 (i386/freebsd))
Content-Length: 0
```

Ringback is transferred to the SIP UA, so the user on the SIP phone will hear ringing.

```
11 Jun 22:10:04/GLOBAL/ser[98217]: SENDING message to 216.231.44.168:9062:
SIP/2.0 180 Ringing
Via: SIP/2.0/UDP
192.168.0.250:9062;received=216.231.44.168;rport=9062;branch=z9hG4bK-1628d42b
Record-Route: <sip:193.28.87.106;ftag=873d0427882f87o0;lr>
From: "John Doe" <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>;tag=b293b17e775f2bdf6d192e545261bb1d
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 102 INVITE
Server: Sippy
```

```
11 Jun 22:10:04/GLOBAL/b2bua: RECEIVED message from 70.68.128.186:5060:
```

The remote SIP proxy requests authorization.

```
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 193.28.87.106:5061;
branch=z9hG4bKade4b51964bd86e3026170c62ce471c6;rport=5061
Record-Route: <sip:70.68.128.186;ftag=f74a7a2b122becfb3bb9ca65f75193f6;lr>
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 200 INVITE
Server: Sippy
WWW-Authenticate: Digest
realm="70.68.128.186",nonce="ec2f8a5c71f14e5a8e08fc77816b3341448c6a8c"
```

```
11 Jun 22:10:04/GLOBAL/b2bua: SENDING message to 70.68.128.186:5060:
```

PortaSIP computes the digest authentication response and sends back a reply.

```
INVITE sip:16046831234@70.68.128.186:5060 SIP/2.0
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK03ab045dae8f997ee66e34049ee07ee8;rport
Max-Forwards: 70
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>
Call-ID: 98dda488-69c74dd3@192.168.0.250
```

```
CSeq: 201 INVITE
Contact: Anonymous <sip:193.28.87.106:5061>
Expires: 300
User-Agent: Sippy
cisco-GUID: 2847286574-946067207-396958912-1950566603
h323-conf-id: 2847286574-946067207-396958912-1950566603
Authorization: Digest username="ipcall-
test",realm="70.68.128.186",nonce="ec2f8a5c71f14e5a8e08fc77816b3341448c6a8c",uri="
sip:
16046831234@70.68.128.186:5060",response="90b1b14cd90e62d2be3b8b6c10ae72f1"
Content-Length: 475
Content-Type: application/sdp
```

```
v=0
o=Sippy 137183084 0 IN IP4 193.28.87.106
s=-
t=0 0
m=audio 16436 RTP/AVP 18 0 2 4 8 96 97 98 100 101
c=IN IP4 216.231.44.168
a=rtpmap:18 G729a/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:2 G726-32/8000
a=rtpmap:4 G723/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:96 G726-40/8000
a=rtpmap:97 G726-24/8000
a=rtpmap:98 G726-16/8000
a=rtpmap:100 NSE/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
aptime:30
a=sendrecv
a=direction:active
a=oldmediaip:192.168.0.250
```

The called party's phone starts ringing.

```
11 Jun 22:10:08/GLOBAL/b2bua: RECEIVED message from 70.68.128.186:5060:
SIP/2.0 183 Session Progress
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK03ab045dae8f997ee66e34049ee07ee8;rport=5061
Record-Route: <sip:70.68.128.186;ftag=f74a7a2b122becfb3bb9ca65f75193f6;lr>
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>;tag=f55ecc2530650faffe5da956658086c9
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 201 INVITE
Server: Sippy
Content-Length: 176
Content-Type: application/sdp
```

```
v=0
o=NexTone-MSW 48600030 0 IN IP4 64.7.121.229
s=sip call
t=0 0
m=audio 35086 RTP/AVP 18 101
c=IN IP4 70.68.128.186
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

The called party answers the call.

```
11 Jun 22:10:09/GLOBAL/b2bua: RECEIVED message from 70.68.128.186:5060:
SIP/2.0 200 OK
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK03ab045dae8f997ee66e34049ee07ee8;rport=5061
Record-Route: <sip:70.68.128.186;ftag=f74a7a2b122becfb3bb9ca65f75193f6;lr>
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
```



```

To: <sip:16046831234@70.68.128.186>;tag=f55ecc2530650faffe5da956658086c9
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 201 INVITE
Server: Sippy
Contact: Anonymous <sip:70.68.128.186:5061>
Content-Length: 176
Content-Type: application/sdp

v=0
o=NexTone-MSW 48600030 0 IN IP4 64.7.121.229
s=sip call
t=0 0
m=audio 35086 RTP/AVP 18 101
c=IN IP4 70.68.128.186
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15

```

Start accounting for the outgoing (originate/VoIP) call leg is sent to the billing.

```

11 Jun 22:10:09/98dda488-69c74dd3@192.168.0.250/b2bua: sending Acct Start
(Originate):
h323-call-origin      = 'originate'
h323-call-type        = 'VoIP'
h323-session-protocol = 'sipv2'
h323-setup-time       = '19:10:04.000 GMT Sun Jun 11 2006'
User-Name              = '16041234568'
Calling-Station-Id     = '16041234568'
Called-Station-Id      = '16046831234'
h323-conf-id          = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id                = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id        = '98dda488-69c74dd3@192.168.0.250'
h323-remote-address    = '70.68.128.186'
h323-ivr-out           = 'DURATION:7200'
h323-ivr-out           = 'PortaBilling_Seed:144514807'
h323-connect-time      = '19:10:09.000 GMT Sun Jun 11 2006'
alert-timepoint        = '19:10:08.000 GMT Sun Jun 11 2006'
Acct-Status-Type       = 'Start'

```

Start accounting for the incoming (answer/VoIP) call leg is sent to the billing.

```

11 Jun 22:10:09/98dda488-69c74dd3@192.168.0.250/b2bua: sending Acct Start
(Answer):
h323-call-origin      = 'answer'
h323-call-type        = 'VoIP'
h323-session-protocol = 'sipv2'
h323-setup-time       = '19:09:44.000 GMT Sun Jun 11 2006'
User-Name              = '16041234568'
Calling-Station-Id     = '16041234568'
Called-Station-Id      = '6831234'
h323-conf-id          = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id                = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id        = '98dda488-69c74dd3@192.168.0.250'
h323-remote-address    = '216.231.44.168'
h323-ivr-out           = 'PortaBilling_Seed:144514807'
h323-connect-time      = '19:10:09.000 GMT Sun Jun 11 2006'
alert-timepoint        = '19:10:09.000 GMT Sun Jun 11 2006'
Acct-Status-Type       = 'Start'

```

One of the parties hangs up; the call termination process is started.

```

11 Jun 22:11:24/GLOBAL/ser[98217]: RECEIVED message from 216.231.44.168:9062:
BYE sip:193.28.87.106:5061 SIP/2.0
Via: SIP/2.0/UDP 192.168.0.250:9062;branch=z9hG4bK-87f40044
From: John Doe <sip:16041234568@193.28.87.106>;tag=873d0427882f87o0
To: <sip:6831234@193.28.87.106>;tag=b293b17e775f2bdf6d192e545261bb1d
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 103 BYE

```

```

Max-Forwards: 70
Route: <sip:193.28.87.106;ftag=873d0427882f87o0;lr>
Authorization: Digest
username="16041234568",realm="193.28.87.106",nonce="3051864d7d0c6578460cc4b0e28ad4
3b448c6a77",uri="sip:
193.28.87.106:5061",algorithm=MD5,response="4e3308a385aef5159ad03f738f9bd31"
User-Agent: Sipura/SPA2000-3.1.5
Content-Length: 0

```

B2BUA sends stop accounting to the billing for the incoming call leg.

```

11 Jun 22:11:24/98dda488-69c74dd3@192.168.0.250/b2bua: sending Acct Stop (Answer):
h323-call-origin      = 'answer'
h323-call-type        = 'VoIP'
h323-session-protocol = 'sipv2'
h323-setup-time       = '19:09:44.000 GMT Sun Jun 11 2006'
User-Name             = '16041234568'
Calling-Station-Id    = '16041234568'
Called-Station-Id     = '6831234'
h323-conf-id          = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id               = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id       = '98dda488-69c74dd3@192.168.0.250'
h323-remote-address   = '216.231.44.168'
h323-ivr-out          = 'PortaBilling_Seed:144514807'
h323-voice-quality    = '0'
Acct-Terminate-Cause  = 'User-Request'
h323-ivr-out          = 'PortaBilling_Session:unlock'
h323-disconnect-time  = '19:11:23.000 GMT Sun Jun 11 2006'
h323-connect-time     = '19:10:09.000 GMT Sun Jun 11 2006'
Acct-Session-Time     = '74'
h323-disconnect-cause = '0'
alert-timepoint       = '19:10:09.000 GMT Sun Jun 11 2006'
Acct-Status-Type      = 'Stop'

```

The outgoing call leg is closed.

```

11 Jun 22:11:24/GLOBAL/b2bua: SENDING message to 70.68.128.186:5060:
BYE sip:70.68.128.186:5061 SIP/2.0
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK13db39161f94a7d3eabcf209661a93;rport
Route: <sip:70.68.128.186;ftag=f74a7a2b122becfb3bb9ca65f75193f6;lr>
Max-Forwards: 70
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>;tag=f55ecc2530650faffe5da956658086c9
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 203 BYE
Contact: Anonymous <sip:193.28.87.106:5061>
Expires: 300
User-Agent: Sippy
cisco-GUID: 2847286574-946067207-396958912-1950566603
h323-conf-id: 2847286574-946067207-396958912-1950566603

```

B2BUA sends stop accounting to the billing for the incoming call leg.

```

11 Jun 22:11:24/98dda488-69c74dd3@192.168.0.250/b2bua: sending Acct Stop
(Originate):
h323-call-origin      = 'originate'
h323-call-type        = 'VoIP'
h323-session-protocol = 'sipv2'
h323-setup-time       = '19:10:04.000 GMT Sun Jun 11 2006'
User-Name             = '16041234568'
Calling-Station-Id    = '16041234568'
Called-Station-Id     = '16046831234'
h323-conf-id          = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id               = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id       = '98dda488-69c74dd3@192.168.0.250'

```

```
h323-remote-address    = '70.68.128.186'
h323-ivr-out           = 'DURATION:7200'
h323-ivr-out           = 'PortaBilling_Seed:144514807'
h323-voice-quality     = '0'
Acct-Terminate-Cause   = 'User-Request'
h323-disconnect-time   = '19:11:23.000 GMT Sun Jun 11 2006'
h323-connect-time      = '19:10:09.000 GMT Sun Jun 11 2006'
Acct-Session-Time      = '74'
h323-disconnect-cause  = '0'
alert-timepoint        = '19:10:08.000 GMT Sun Jun 11 2006'
Acct-Status-Type       = 'Stop'
```

The other party confirms call disconnection.

```
11 Jun 22:11:24/GLOBAL/b2bua: RECEIVED message from 70.68.128.186:5060:
SIP/2.0 200 OK
Via: SIP/2.0/UDP
193.28.87.106:5061;branch=z9hG4bK13db39161f94a7d3eabcf209661a93;rport=5061
From: "John Doe"
<sip:16041234568@193.28.87.106>;tag=f74a7a2b122becfb3bb9ca65f75193f6
To: <sip:16046831234@70.68.128.186>;tag=f55ecc2530650faffe5da956658086c9
Call-ID: 98dda488-69c74dd3@192.168.0.250
CSeq: 203 BYE
Server: Sippy
```

The call is finished.



- `/var/log/porta-billing.log"` less
- **Trace call**



View log

PortaBilling receives the authorization request.

```
Jun 11 12:09:44: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address      = '193.28.87.106'
User-Name           = '16041234568'
Called-Station-Id   = '6831234'
Calling-Station-Id  = '16041234568'
h323-conf-id        = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id             = '98dda488-69c74dd3@192.168.0.250'
Digest-Attributes   = 'Realm = "193.28.87.106"'
Digest-Attributes   = 'Nonce = "3051864d7d0c6578460cc4b0e28ad43b448c6a77"'
Digest-Attributes   = 'Method = "INVITE"'
Digest-Attributes   = 'URI = "sip:6831234@193.28.87.106"'
Digest-Attributes   = 'Algorithm = "MD5"'
Digest-Attributes   = 'User-Name = "16041234568"'
Digest-Response     = '54e0b42337ace33edf36d004f1037ebd'
h323-remote-address = '216.231.44.168'
h323-session-protocol = 'sipv2'
h323-ivr-out        = 'PortaBilling_Routing:SIP'
h323-ivr-out        = 'PortaBilling_AuthMethod:INVITE'
h323-ivr-out        = 'PortaBilling_Notify:NAT'
h323-ivr-out        = 'PortaBilling_Seed:144514807'
NAS-Port            = '5060'
Jun 11 12:09:44: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:09:44: H323/SIP call, use h323-conf-id, but remember call-id
Jun 11 12:09:44: Checking if this call comes through a VoIP from vendor connection
Jun 11 12:09:44: No VoIP from vendor connections were found
Jun 11 12:09:44: PrepareNexecute 'AccountAuth'
Jun 11 12:09:44: Found Account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000]
```

Account information is located in the database.

```
Jun 11 12:09:44: Account 16041234568 is not logged in yet
Jun 11 12:09:44: Verification of password using method 'digest_response': success
```

Password verification is successful.

```
Jun 11 12:09:44: Applying override translation rule on CLD ...
Jun 11 12:09:44: Translation 's/^\*3164\*/' applied: '6831234' unchanged
Jun 11 12:09:44: Applying customer dialing translation rule on CLD ...
Jun 11 12:09:44: Translation
'local_to_e164($_,{cc=>'1',ac=>'604',dp=>'1',ip=>'011',em=>'911',ex=>'411'})';##
cc=1 ac=604 dp=1 ip=011 em=911 ex=411' applied: '6831234' -> '16046831234'
```

The customer's number translation rule is applied, and the phone number is changed from the local format into E.164.

```
Jun 11 12:09:44: PrepareNexecute 'AccountAuth'
Jun 11 12:09:44: CLD '16046831234' is an off-net number
```

The dialed number is an off-net destination.

```
Jun 11 12:09:44: Setting up a new charge with tariff 'SIP Phone Subscribers' ...
Jun 11 12:09:44: Checked 2006-06-11 21:09:44 Europe/Prague against 'hr{20-5}': 1
Jun 11 12:09:44: Start of call is peak level 1
Jun 11 12:09:44: PrepareNexecute 'GetPricePerDestination*'
Jun 11 12:09:44: Maximum call duration: 29460 announced as 29460
(!+1x30x0.02+4905x6x0.02) by rate 101 using 144547388 as seed
```

Maximum call duration is calculated according to the available funds and rate.

```
Jun 11 12:09:44: Remote termination 'PortaSIP': Calculating routing for
16046831234
Jun 11 12:09:44: RTP Proxy with origination preference 'Undetermined', Calling
party behind NAT.
Jun 11 12:09:44: Looking up routes to '16046831234' using '<Default System
Routing>' routing plan
Jun 11 12:09:44: PrepareNexecute 'GetRoutingPerDestination*'
Jun 11 12:09:44: Using peak rate, since no off-peak is defined
Jun 11 12:09:44: Result routes to destination '16046831234':
    16046831234@193.28.187.3, prio = 9, cost = 0.07000, 'test' - to remote GW
    16046831234@70.68.128.186, prio = 7, cost = 0.01000, 'Termination to
Globalnet' - to remote GW
    16046831234@69.104.30.123, prio = 6, cost = 0.55000, 'VoIPio via SIP' - to
remote GW
    16046831234@192.168.0.66, prio = 6, cost = 0.60000, 'Premium VoIP->Vendor'
- to remote GW
    16046831234@192.168.0.30, prio = 6, cost = 0.60000, 'X-Telecom' - to
remote GW
```

There are five possible routes (sorted according to preference and cost).

```
Jun 11 12:09:44: Logging in account '16041234568'(103) to 'A9B6252E 3863D707
17A91CC0 744348CB'
Jun 11 12:09:44: Authentication acknowledge response
```

An authorization response is sent to PortaSIP.

```
Cisco-AVPair      = h323-ivr-in=PortaBilling_Routing:@;g-
hunt=seq;expires=300;credit-time=29460;patience=20
Cisco-AVPair      = h323-ivr-in=PortaBilling_Routing: 16046831234@193.28.187.3
Cisco-AVPair      = h323-ivr-in=PortaBilling_Routing:
16046831234@70.68.128.186;auth=C43B527B8BCCF31A5CB84F49D8D576DE613800D328115690
Cisco-AVPair      = h323-ivr-in=PortaBilling_Routing:
16046831234@69.104.30.123;auth=3741D17EED2A31B7990D8DFEC859D77EA2147BAEE44CD5DBBD9
C1C4D35DE9A99;rtp=1
Cisco-AVPair      = h323-ivr-in=PortaBilling_Routing: 16046831234@192.168.0.66
Cisco-AVPair      = h323-ivr-in=PortaBilling_Routing: 16046831234@192.168.0.30
h323-billing-model = 0
h323-ivr-in        = Tariff:SIP Phone Subscribers
h323-ivr-in        = PortaBilling_CLI:16041234568
h323-ivr-in        = MOH:1
h323-ivr-in        = PortaBilling_CompleteNumber:16046831234
h323-ivr-in        = DURATION:29460
h323-return-code   = 0
h323-currency       = USD
h323-credit-time    = 29460
h323-preferred-lang = en
Jun 11 12:09:44: ...Done.
```

Accounting for the failed outgoing call leg is received.

```
Jun 11 12:10:04: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address     = '193.28.87.106'
User-Name          = '16041234568'
Called-Station-Id  = '16046831234'
```

```
Calling-Station-Id      = '16041234568'
Acct-Status-Type        = 'Stop'
h323-call-origin        = 'originate'
h323-call-type          = 'VoIP'
h323-setup-time         = '19:09:44.000 GMT Sun Jun 11 2006'
h323-connect-time       = '19:10:04.000 GMT Sun Jun 11 2006'
h323-disconnect-time    = '19:10:04.000 GMT Sun Jun 11 2006'
h323-disconnect-cause   = '10'
h323-conf-id            = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id                 = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id         = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Time       = '0'
Acct-Delay-Time         = '0'
h323-session-protocol   = 'sipv2'
h323-remote-address     = '193.28.187.3'
h323-ivr-out            = 'DURATION:7200'
h323-ivr-out            = 'PortaBilling_Seed:144514807'
NAS-Port                = '5060'
Exec-Program-Log        = 'porta-billing.pl'
Jun 11 12:10:04: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:10:04: Found a call in cache with such id
Jun 11 12:10:04: Copied account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000] from '193.28.87.106' into the current
request
```

The billing re-uses information in the call cache to speed up account info lookup.

```
Jun 11 12:10:04: PrepareNexecute 'GetActiveLegIdByAcct'
Jun 11 12:10:04: End of the outgoing failed call for logged in account. Waiting
another outgoing call or hang up
Jun 11 12:10:04: Looking up vendor/connection
Jun 11 12:10:04: Trying to match connection for call
Jun 11 12:10:04: Looking for a connection VoIP/originate
Jun 11 12:10:04: Outgoing VoIP, matching by the remote IP address '193.28.187.3'
(env 4)
Jun 11 12:10:04: Found connection 4 'test' to vendor 'MCI Vendor'
```

Connection matched.

```
Jun 11 12:10:04: Found vendor/connection
Jun 11 12:10:04: Charging call ...
Jun 11 12:10:04: Zero duration call
Jun 11 12:10:04: Checked 2006-06-11 21:10:04 Europe/Prague against 'hr{20-5}': 1
Jun 11 12:10:04: Start of call is peak level 1
Jun 11 12:10:04: End of call is peak level 1
Jun 11 12:10:04: Can reuse the already initialized charge.
Jun 11 12:10:04: Calculating account's charge by tariff 'SIP Phone Subscribers'
Jun 11 12:10:04: Call to '16046831234' with duration 0 seconds will be charged for
0 seconds and cost is 0 (0s<1s) by rate 101 using 144547388 as seed
Jun 11 12:10:04: Setting up a new charge with tariff 'CT Tariff' ...
Jun 11 12:10:04: Using peak rate, since no off-peak is defined
Jun 11 12:10:04: PrepareNexecute 'GetPricePerDestination*'
Jun 11 12:10:04: SQL query 'GetPricePerDestination*' executed in 0.001555 seconds
Jun 11 12:10:04: Calculating vendor's charge by tariff 'CT Tariff'
Jun 11 12:10:04: Call to '16046831234' with duration 0 seconds will be charged for
0 seconds and cost is 0 (0s<1s) by rate 106 using 144547388 as seed
Jun 11 12:10:04: Updating account usage of '16041234568' with 'Sun Jun 11 12:10:04
2006'
Jun 11 12:10:04: PrepareNexecute 'UpdateAccountUsage'
Jun 11 12:10:04: Charging vendor for the call
Jun 11 12:10:04: Inserting fail CDR
Jun 11 12:10:04: PrepareNexecute 'InsertVendorCDRFail'
```

A failed CDR is inserted for the vendor.

```
Jun 11 12:10:04: Accounting response
```

Jun 11 12:10:04: ...Done.

The call is now established, and the start accounting record is sent.

```
Jun 11 12:10:09: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address      = '193.28.87.106'
User-Name           = '16041234568'
Called-Station-Id   = '6831234'
Calling-Station-Id  = '16041234568'
Acct-Status-Type    = 'Start'
h323-call-origin    = 'answer'
h323-call-type      = 'VoIP'
h323-setup-time     = '19:09:44.000 GMT Sun Jun 11 2006'
h323-connect-time   = '19:10:09.000 GMT Sun Jun 11 2006'
h323-conf-id        = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id             = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id     = '98dda488-69c74dd3@192.168.0.250'
Acct-Delay-Time     = '0'
h323-session-protocol = 'sipv2'
h323-remote-address = '216.231.44.168'
h323-ivr-out        = 'PortaBilling_Seed:144514807'
alert-timpoint      = '19:10:09.000 GMT Sun Jun 11 2006'
NAS-Port            = '5060'
Exec-Program-Log    = 'porta-billing.pl'
Jun 11 12:10:09: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:10:09: Found a call in cache with such id
Jun 11 12:10:09: Copied account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000] from '193.28.87.106' into the current
request
```

The billing re-uses information in the call cache to speed up account info lookup.

```
Jun 11 12:10:09: Fixing time with duration 0 seconds from connect time
Jun 11 12:10:09: PrepareNexecute 'GetActiveLegIdByAcct'
Jun 11 12:10:09: Looking up vendor/connection
Jun 11 12:10:09: Trying to match connection for call
Jun 11 12:10:09: Looking for a connection VoIP/answer
Jun 11 12:10:09: VoIP, matching by the node IP '193.28.87.106' and User-Name
'16041234568'
Jun 11 12:10:09: No VoIP from vendor connections were found
Jun 11 12:10:09: Connection to vendor not found
Jun 11 12:10:09: PrepareNexecute 'InsertActiveLeg'
```

The call has not been billed yet, but an entry is made in the table of active calls.

```
Jun 11 12:10:09: Accounting response
Jun 11 12:10:09: ...Done.
```

A start accounting record about the egress call leg is received.

```
Jun 11 12:10:09: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address      = '193.28.87.106'
User-Name           = '16041234568'
Called-Station-Id   = '16046831234'
Calling-Station-Id  = '16041234568'
Acct-Status-Type    = 'Start'
h323-call-origin    = 'originate'
h323-call-type      = 'VoIP'
h323-setup-time     = '19:10:04.000 GMT Sun Jun 11 2006'
h323-connect-time   = '19:10:09.000 GMT Sun Jun 11 2006'
h323-conf-id        = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id             = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id     = '98dda488-69c74dd3@192.168.0.250'
Acct-Delay-Time     = '0'
```

```

h323-session-protocol = 'sipv2'
h323-remote-address   = '70.68.128.186'
h323-ivr-out          = 'DURATION:7200'
h323-ivr-out          = 'PortaBilling_Seed:144514807'
alert-timepoint       = '19:10:08.000 GMT Sun Jun 11 2006'
NAS-Port              = '5060'
Exec-Program-Log      = 'porta-billing.pl'
Jun 11 12:10:09: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:10:09: Found a call in cache with such id
Jun 11 12:10:09: Copied account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000] from '193.28.87.106' into the current
request

```

The billing re-uses information in the call cache to speed up account info lookup.

```

Jun 11 12:10:09: Fixing time with duration 0 seconds from connect time
Jun 11 12:10:09: PrepareNexecute 'GetActiveLegIdByAcct'
Jun 11 12:10:09: SQL query 'GetActiveLegIdByAcct' executed in 0.00115 seconds
Jun 11 12:10:09: Looking up vendor/connection
Jun 11 12:10:09: Trying to match connection for call
Jun 11 12:10:09: Looking for a connection VoIP/originate
Jun 11 12:10:09: Outgoing VoIP, matching by the remote IP address '70.68.128.186'
(env 4)
Jun 11 12:10:09: Found connection 11 'Termination to Globalnet' to vendor
'GlobalNet'
Jun 11 12:10:09: Found vendor/connection
Jun 11 12:10:09: PrepareNexecute 'InsertActiveLeg'

```

The call has not been billed yet, but an entry is made in the table of active calls.

```

Jun 11 12:10:09: Accounting response
Jun 11 12:10:09: ...Done.

```

The call is terminated, and stop accounting for one of the call legs is received.

```

Jun 11 12:11:24: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address       = '193.28.87.106'
User-Name            = '16041234568'
Called-Station-Id    = '6831234'
Calling-Station-Id   = '16041234568'
Acct-Status-Type     = 'Stop'
h323-call-origin     = 'answer'
h323-call-type       = 'VoIP'
h323-setup-time      = '19:09:44.000 GMT Sun Jun 11 2006'
h323-connect-time    = '19:10:09.000 GMT Sun Jun 11 2006'
h323-disconnect-time = '19:11:23.000 GMT Sun Jun 11 2006'
h323-disconnect-cause = '0'
h323-voice-quality    = '0'
h323-conf-id         = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id              = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id      = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Time    = '74'
Acct-Delay-Time      = '0'
h323-session-protocol = 'sipv2'
h323-remote-address   = '216.231.44.168'
h323-ivr-out          = 'PortaBilling_Seed:144514807'
Acct-Terminate-Cause  = 'User-Request'
h323-ivr-out          = 'PortaBilling_Session:unlock'
alert-timepoint       = '19:10:09.000 GMT Sun Jun 11 2006'
NAS-Port              = '5060'
Exec-Program-Log      = 'porta-billing.pl'
Jun 11 12:11:24: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:11:24: Found a call in cache with such id
Jun 11 12:11:24: Copied account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall

```



```
Ltd.[3,balance=0.18000,limit=100.00000] from '193.28.87.106' into the current
request
Jun 11 12:11:24: PrepareNexecute 'GetActiveLegIdByAcct'
Jun 11 12:11:24: PrepareNexecute 'DeleteActiveLeg'
Jun 11 12:11:24: Force unlock requested by NAS
Jun 11 12:11:24: Scheduling 16041234568 for logout, call lifetime reduced to 15
Jun 11 12:11:24: Logging out account '16041234568'(103) from 'A9B6252E 3863D707
17A91CC0 744348CB'
Jun 11 12:11:24: Set lifetime with 15s to Sun Jun 11 12:11:39 2006
Jun 11 12:11:24: Looking up vendor/connection
Jun 11 12:11:24: Trying to match connection for call
Jun 11 12:11:24: Looking for a connection VoIP/answer
Jun 11 12:11:24: VoIP, matching by the node IP '193.28.87.106' and User-Name
'16041234568'
Jun 11 12:11:24: No VoIP from vendor connections were found
Jun 11 12:11:24: Connection to vendor not found
Jun 11 12:11:24: No connection from vendor
```

This is an on-net call leg (while the call is still traveling on the network), so it is ignored.

```
Jun 11 12:11:24: Accounting response
Jun 11 12:11:24: ...Done.
```

Accounting for the second (outgoing) call leg is received.

```
Jun 11 12:11:24: Processing request (BE ver1.245.2.4,pid32021):
NAS-IP-Address      = '193.28.87.106'
User-Name           = '16041234568'
Called-Station-Id   = '16046831234'
Calling-Station-Id  = '16041234568'
Acct-Status-Type    = 'Stop'
h323-call-origin    = 'originate'
h323-call-type      = 'VoIP'
h323-setup-time     = '19:10:04.000 GMT Sun Jun 11 2006'
h323-connect-time   = '19:10:09.000 GMT Sun Jun 11 2006'
h323-disconnect-time = '19:11:23.000 GMT Sun Jun 11 2006'
h323-disconnect-cause = '0'
h323-voice-quality   = '0'
h323-conf-id        = 'A9B6252E 3863D707 17A91CC0 744348CB'
call-id             = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Id     = '98dda488-69c74dd3@192.168.0.250'
Acct-Session-Time    = '74'
Acct-Delay-Time      = '0'
h323-session-protocol = 'sipv2'
h323-remote-address  = '70.68.128.186'
h323-ivr-out         = 'DURATION:7200'
h323-ivr-out         = 'PortaBilling_Seed:144514807'
Acct-Terminate-Cause = 'User-Request'
alert-timepoint      = '19:10:08.000 GMT Sun Jun 11 2006'
NAS-Port             = '5060'
Exec-Program-Log     = 'porta-billing.pl'
Jun 11 12:11:24: h323-conf-id=A9B6252E 3863D707 17A91CC0 744348CB/4, call-
id=98dda488-69c74dd3@192.168.0.250/4
Jun 11 12:11:24: Found a call in cache with such id
Jun 11 12:11:24: Copied account:
16041234568[103,credit,balance=10.18000,limit=20.00000] of customer EasyCall
Ltd.[3,balance=0.18000,limit=100.00000] from '193.28.87.106' into the current
request
Jun 11 12:11:24: PrepareNexecute 'GetActiveLegIdByAcct'
Jun 11 12:11:24: PrepareNexecute 'DeleteActiveLeg'
Jun 11 12:11:24: End of the outgoing call for logged in account. Waiting another
outgoing call or hang up
Jun 11 12:11:24: Set lifetime with 15s to Sun Jun 11 12:11:39 2006
Jun 11 12:11:24: Looking up vendor/connection
Jun 11 12:11:24: Trying to match connection for call
Jun 11 12:11:24: Looking for a connection VoIP/originate
Jun 11 12:11:24: Outgoing VoIP, matching by the remote IP address '70.68.128.186'
(env 4)
```

```
Jun 11 12:11:24: Found connection 11 'Termination to Globalnet' to vendor  
'GlobalNet'
```

This call leg crosses a connection to the vendor.

```
Jun 11 12:11:24: Found vendor/connection  
Jun 11 12:11:24: Charging call ...  
Jun 11 12:11:24: Checked 2006-06-11 21:10:09 Europe/Prague against 'hr{20-5}': 1  
Jun 11 12:11:24: Checked 2006-06-11 21:11:23 Europe/Prague against 'hr{20-5}': 1  
Jun 11 12:11:24: Start of call is peak level 1  
Jun 11 12:11:24: End of call is peak level 1  
Jun 11 12:11:24: Can reuse the already initialized charge.  
Jun 11 12:11:24: Calculating account's charge by tariff 'SIP Phone Subscribers'  
Jun 11 12:11:24: Call to '16046831234' with duration 74 seconds will be charged  
for 78 seconds and cost is 0.03 (1x30x0.02+8x6x0.02^2) by rate 101 using 144547388  
as seed
```

PortaBilling calculates how much the account should be charged for this call.

```
Jun 11 12:11:24: Setting up a new charge with tariff 'GlobalNet Termination' ...  
Jun 11 12:11:24: Using peak rate, since no off-peak is defined  
Jun 11 12:11:24: PrepareNexecute 'GetPricePerDestination*'  
Jun 11 12:11:24: Calculating vendor's charge by tariff 'GlobalNet Termination'  
Jun 11 12:11:24: Call to '16046831234' with duration 74 seconds will be charged  
for 74 seconds and cost is 0.01234 (1x1x0.01+73x1x0.01) by rate 104 using  
144547388 as seed
```

PortaBilling also calculates the termination costs for this call.

```
Jun 11 12:11:24: Charging account for the call  
Jun 11 12:11:24: Inserting CDR  
Jun 11 12:11:24: PrepareNexecute 'InsertAccountCDR'  
Jun 11 12:11:24: Charging credit account 16041234568 0.03  
Jun 11 12:11:24: PrepareNexecute 'UpdateAccountBalance'  
Jun 11 12:11:24: Charging account's owner for the call  
Jun 11 12:11:24: Charging customer 3 'EasyCall Ltd.' 0.03  
Jun 11 12:11:24: PrepareNexecute 'UpdateCustomerBalance'  
Jun 11 12:11:24: Charging vendor for the call  
Jun 11 12:11:24: Charging vendor 9 'GlobalNet' 0.01234  
Jun 11 12:11:24: Inserting CDR  
Jun 11 12:11:24: PrepareNexecute 'InsertVendorCDR'  
Jun 11 12:11:24: PrepareNexecute 'UpdateVendorBalance'
```

CDRs are inserted and balances are modified.

```
Jun 11 12:11:24: Accounting response  
Jun 11 12:11:24: ...Done.
```

Verify Call History for the Account






Accounts

Home

Accounts of Retail Customer 'EasyCall Ltd.' America/Vancouver demo-tool Help

[Add](#) [Account Generator](#) [Close](#) [Logout](#)

Account ID	Batch	Ctrl #	SIP Status	Advanced search
	ANY		ANY	Show Accounts

xDRs	Account ID	Idle, days	Currency	Balance	Credit Limit	Type	Product	Batch	Status	SIP
	16041234567		USD	9.91000		Debit	SIP Subscribers	easycall		
	16041234568	0	USD	3.98356	20.00000	Credit	SIP Subscribers	easycall		
	16041234569		USD	9.98000		Debit	SIP Subscribers	easycall		
	16041234570		USD	0.00000	10.00000	Credit	SIP Subscribers	easycall		

xDR History America/Vancouver demo-tool Help

[Close](#)

[From Date](#) YYYY-MM-DD HH24:MI:SS

[To Date](#) YYYY-MM-DD HH24:MI:SS

Service Type

Show Unsuccessful Attempts ☐

[Show xDRs](#)







xDR History America/Vancouver demo-tool Help

[Close](#) [Download](#) [xDRs](#)

Account	16041234568	Credits/Refunds	0.00000 USD
From	2007-06-25 07:45:00	Payments	0.00000 USD
To	2007-06-25 07:59:32	Subscriptions Charged	0.00000 USD
Charged by	'SIP Subscribers' product	Services Charged	0.14000 USD
Type	Credit	Total	0.14000 USD
Total Transactions	3		

[Show Totals By Service Types](#)

Voice Calls

View	Account	From	To	Country	Description	Date/Time	Charged time, min:sec	Amount, USD	Refund
	16041234568	16041234568	16046831234	CANADA	British Columbia	2007-06-25 07:57:15	1:12	0.03000	
	16041234568	16041234568	16046282508	CANADA	British Columbia	2007-06-25 07:51:30	4:24	0.09000	
	16041234568	16041234568	16043102255	CANADA	British Columbia	2007-06-25 07:49:20	0:48	0.02000	
Subtotal							6:24	0.14000	

Show xDR

 [Download .csv](#)

Check the Call History

Trace Call

Call Trace

America/Los_Angeles

demoroot

Help

Close

Logout

H323-conf-id

Trace a call

Destination %

10 min

From

2006-06-11

YYYY-MM-DD

00:00:00

HH:MM:SS

To

2006-06-12

YYYY-MM-DD

00:00:00

HH:MM:SS

Trace a call

View	Error Report	CLI(ani)	CLD(dnis)	Country	Description	Connect Time	Disconnect Time	Duration, min:sec	Amount	Account	Customer	Vendor	Disconnect Reason
		16041234568	16046831234	CANADA	British Columbia	2006-06-11 12:10:09	2006-06-11 12:11:23	1:14	0.01234 USD	16041234568	EasyCall Ltd.	GlobalNet	Normal call clearing
		16041234568	16046831234	CANADA	British Columbia	2006-06-11 12:10:04	2006-06-11 12:10:04	0:00	0 USD	16041234568	EasyCall Ltd.	MCI	Recovery on timer expiry
		16041234568	16043102255	CANADA	British Columbia	2006-06-11 12:05:35	2006-06-11 12:05:55	0:20	0.00334 USD	16041234568	EasyCall Ltd.	GlobalNet	Normal call clearing
		16041234568	16043102255	CANADA	British Columbia	2006-06-11 12:05:30	2006-06-11 12:05:30	0:00	0 USD	16041234568	EasyCall Ltd.	MCI Vendor	Bearer service not implemented
		16041234568	16046282508	CANADA	British Columbia	2006-06-11 12:03:59	2006-06-11 12:04:05	0:06	0.00100 USD	16041234568	EasyCall Ltd.	GlobalNet	Normal call clearing
		16041234568	16046282508	CANADA	British Columbia	2006-06-11 12:03:54	2006-06-11 12:03:54	0:00	0 USD	16041234568	EasyCall Ltd.	MCI Vendor	Recovery on timer expiry

List of possible Disconnect reasons:

Normal completed call

Normal uncompleted call

Call progress code

Calling side error

Called side error

Network error

Trace a Call

-
- - h323-conf-id
 - Destination
 - 380%
 - From, To Date –
- Trace a Call.

The advantage of this method:

xDR

2. Setting up PSTN-to-SIP Services

PortaBilling100 Web Reference Guide

Incoming DID calls (from PSTN)

Checklist



Operation Done

Initial configuration

Basic SIP service

Basic SIP service

Network configuration

Rating configuration

Routing

Account provisioning

Basic SIP service

Testing

Cisco gateway configuration guidelines

Obtain the PSTN2SIP script

```
/tftpboot
```

Basic router configuration

```
hostname <h323_id>  
ip domain name <default domain>
```

NOTE: VSA h323-gw-id="hostname.domain"

NTP

NOTE: It is very important to have reliable time services. Also make sure that the time zone abbreviation is one of the standard ones supported by PortaBilling.

```
ntp server <name/IP>  
.....  
ntp server <name/IP>  
ntp master 5  
clock timezone <your time zone> 1  
clock summer-time <your summer time zone> recurring <your  
rules>
```

AAA

```
aaa new-model  
aaa authentication login h323 group radius  
aaa authorization exec h323 group radius  
aaa accounting connection h323 stop-only group radius
```

VoIP interface

```
interface <your interface to the world>  
h323-gateway voip interface  
h323-gateway voip h323-id <h323_id>
```

NOTE: If you want to use a virtual interface then add the line:

```
h323-gateway voip bind srcaddr <IP>
```

Outgoing SIP server

```
sip-ua
  aaa username proxy-auth
  sip-server dns:<hostname-of-your-PortaSIP-server>
```

Enable gateway functionality

```
gateway
```

Enable gateway accounting

```
gw-accounting h323 vsa

gw-accounting aaa
  acct-template callhistory-detail
```

NOTE: VSA does not work for all platforms.

Radius

IMPORTANT NOTE: Ports 1645/1646 are the traditional Radius ports used by many vendors, without obtaining an official IANA assignment. The official assignment is now ports 1812/1813, and users are encouraged to migrate to these new ports when possible.

Cisco notes:

- “radius-server” commands will be available only after issuing the “aaa new-model” command
- UDP port for RADIUS accounting server – the default is 1646 (see note above)
- UDP port for RADIUS authentication server – the default is 1645 (see note above)

Remember:

- The default ports for Cisco are 1645/1646
- The defaults in /etc/ services are 1812/1813

```
radius-server host <name/IP> auth-port 1812 acct-port 1813
radius-server key <key>
radius-server vsa send accounting
radius-server vsa send authentication
```

voice-card

controller

voice-port

call application voice & dial-peers

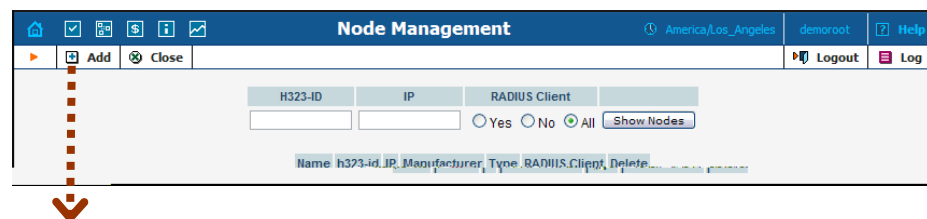
```
call application voice pstn2sip tftp://your-server/pstn2sip.tcl
call application voice pstn2sip authenticate-by dnis
call application voice pstn2sip skip-password yes
call application voice pstn2sip authorize yes
call application voice pstn2sip dial-account-id yes
```

```
dial-peer voice 100 pots
  incoming called-number .T
  application pstn2sip
  voice-port 0:d
!
dial-peer voice 60 voip
  destination-pattern .T
  session protocol sipv2
  session target sip-server
!
```

```
call application voice pstn2sip translate "/^/1/"
```

Set up a PSTN-to-SIP Service

Create a PSTN Gateway Node



Add Node
America/Vancouver
demo-tool
Help
Save Save & Close Close Logout

Node Name PSTN-GW-NY-01

Node info

h323-id PSTN-GW-NY-01.mydomain.com

VoIP Password aml6th Auto

NAS-IP-Address 43.170.34.100

Hostname

Domain

Auth. Transl. Rule ?

Manufacturer Cisco

Type VOIP-GW

RTP Proxying Optimal

RADIUS Client

RADIUS Key ss3ihiso Auto

RADIUS Source IP 43.170.34.100

RADIUS Dictionary Cisco

Node Management
America/Vancouver
demo-tool
Help
Add Close Logout Log

H323-ID IP RADIUS Client

Name h323-id IP Manufacturer Type RADIUS Client Delete

cisco-gw cisco-gw.mydomain.com 192.168.0.99 Cisco VOIP-GW

DemoSIP demosip 193.28.87.41 PortaOne PortaSIP

PSTN-GW-NY-01 PSTN-GW-NY-01.mydomain.com 43.170.34.100 Cisco VOIP-GW

The page at https://demo.portaone.com says:

Submitted information is being cached in the billing engine and will not take effect immediately. Default caching time is 10 minutes. Please contact your system administrator for more information.

OK

Nodes.

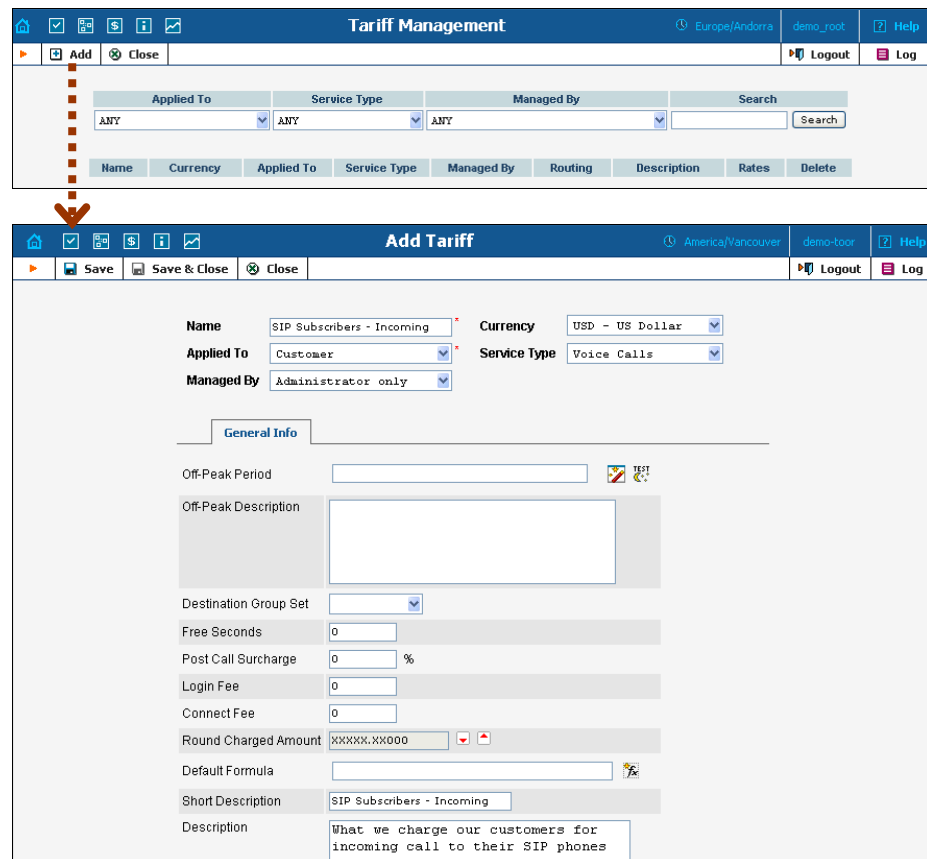
Add

- Node name
- h323-ID hostname.domainname
- VoIP Password cisco
- NAS-IP-Address
- Auth. Translation rule
- Manufacturer Cisco
- Type VOIP-GW
- Radius Client
- Radius Key key
- Radius Source IP Node ID, NAS IP address and PortaBilling Administrator Guide

 Save & Close

NOTE: There is some propagation delay between the database and the Radius server configuration file; however, it is no more than 15 minutes.

Create a Tariff to Charge Your Subscribers



The screenshot shows two parts of the application. The top part is the 'Tariff Management' table with columns: Name, Currency, Applied To, Service Type, Managed By, Routing, Description, Rates, and Delete. It has filters for Applied To (ANY), Service Type (ANY), and Managed By (ANY), along with a search bar. The bottom part is the 'Add Tariff' form. It has a 'General Info' tab. The form fields are: Name (SIP Subscribers - Incoming), Currency (USD - US Dollar), Applied To (Customer), Service Type (Voice Calls), Managed By (Administrator only), Off-Peak Period (empty), Off-Peak Description (empty), Destination Group Set (empty), Free Seconds (0), Post Call Surcharge (0 %), Login Fee (0), Connect Fee (0), Round Charged Amount (XXXXXX.XX000), Default Formula (empty), Short Description (SIP Subscribers - Incoming), and Description (What we charge our customers for incoming call to their SIP phones).

Tariffs

 Add

Add Tariff

- Name
- Currency

NOTE: The currency for the tariff may be chosen only once, and cannot be changed later.

- Applied To Customer
- Managed By Administrator Only
Applied to: Customer
- Service Type Voice Calls
- Off-peak Period



Help

- Off-peak Description
- Destination Group Set
- Free Seconds
- Post-Call Surcharge
- Login Fee
- Connect Fee
- Round Charged Amount

- Default Formula
- Short Description
- Description

 Save

Enter Rates

System Concepts

Managing rates online

'SIP Subscribers - Incoming' tariff rates

Effective From: Now | Destination: Prefix, Group, Country

Edit	Destination *	Country	Interval, second		Price, USD/minute		Effective From				
	Description		First *	Next *	First *	Next *	YYYY-MM-DD HH24:MI:SS *				

'SIP Subscribers - Incoming' tariff rates

Effective From: Now | Destination: Prefix, Group, Country

Edit	Destination *	Country	Interval, second		Price, USD/minute		Effective From				
	Description		First *	Next *	First *	Next *	YYYY-MM-DD HH24:MI:SS *				
	474	NORWAY Mobile	30	6	0.11	0.11	immediately				

Rates

Rates

Edit Rates

Add

- Destination
- 47


Destination

NOTE: The phone prefix you are trying to create a rate for must already exist in Destinations.

- Interval First
- Interval Next
- Price First
- Price Next
- Off-peak Interval First
- Off-peak Interval Next
- Off-peak Price First

- Off-peak Price Next

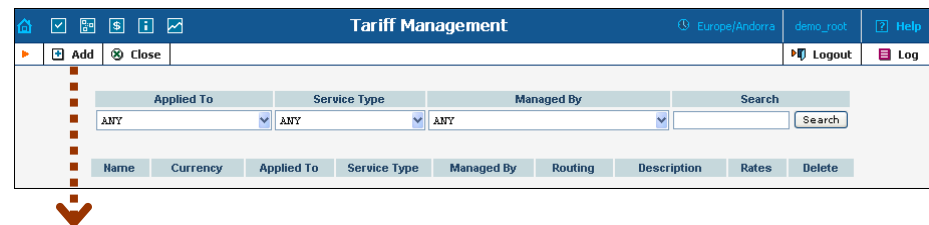
NOTE: Off-peak fields appear only if an **off-peak period** has been defined for the tariff.





- Rating Formula Wizard 
- Effective from

NOTE: When using the calendar, you can specify that the date you are entering is in a different time zone than your present one. PortaBilling will then automatically adjust the time.

- Hidden Forbidden Discontinued
 Save 

Create a Tariff for Incoming DID Costs



Tariff Management								Europe/Andorra	demo_root	Help	
	Add		Close						Logout		Log
Applied To		Service Type		Managed By		Search					
ANY		ANY		ANY		<input type="text"/> <input type="button" value="Search"/>					
Name	Currency	Applied To	Service Type	Managed By	Routing	Description	Rates	Delete			

Add Tariff

Name: DID supplier costs * Currency: USD - US Dollar

Applied To: Vendor * Service Type: Voice Calls

Routing: ☐

General Info

Off-Peak Period:

Off-Peak Description:

Destination Group Set:

Free Seconds:

Post Call Surcharge: %

Login Fee:

Connect Fee:

Round Charged Amount:

Default Formula:

Short Description: DID Supplier

Description: What we are being charged by our DID supplier for incoming calls

Tariffs

Add

Add Tariff

Applied To
Routing

Vendor

Save

Enter Rates

'DID supplier costs' tariff rates

Effective From: Now

Destination:

Edit	Destination *	Country	Interval, second		Price, USD/minute		Effective From	Delete
			First *	Next *	First *	Next *		
	4722	NORWAY Oslo	60	60	0.005	0.005	immediately	

Rates

Rates

Edit Rates

Add



Modify the Accessibility for the Product

Product Management America/Los_Angeles demoproot Help

Add Close Logout Log

Managed By: ANY Search: Search

Icon	Name	Currency	Managed By	Description	Delete
	CT Product	USD	Administrator only	Cisco Test Product	
	PortaOne	USD	Administrator only		
	Prepaid Cards	USD	Administrator only	Prepaid Cards	
	SIP Subscribers	USD	Administrator only	Product for SIP users. No monthly fee.	
	SmartCall	USD	Administrator only	SmartCall Prepaid cards	
	Smartnet Termination	USD	Administrator only	What we charge Customer Smartnet for calls they terminate to us	

Edit 'SIP Subscribers' Product America/Vancouver demo-toor Help

Add Save Save & Close Close Rate Lookup Logout Log

Product Name: SIP Subscribers Currency: USD
Managed By: Administrator only

General Info Maintenance Fee Online Signup Accessibility Subscriptions Notepad

Edit	Service Type	Node	Access Code	Info Digits	Tariff	Delete
	NOT SELECTED	ANY		ANY		
	Voice Calls	DemoSIP			SIP Phone Subscribers	

Edit 'SIP Subscribers' Product America/Vancouver demo-toor Help

Add Save Save & Close Close Rate Lookup Logout Log

Product Name: SIP Subscribers Currency: USD
Managed By: Administrator only

General Info Maintenance Fee Online Signup Accessibility Subscriptions Notepad

Edit	Service Type	Node	Access Code	Info Digits	Tariff	Delete
	Voice Calls	PSTN-GW-NY-01		ANY	SIP Subscribers -	
	Voice Calls	DemoSIP			SIP Phone Subscribers	

Edit 'SIP Subscribers' Product America/Vancouver demo-toor Help

Add Save Save & Close Close Rate Lookup Logout Log

Product Name: SIP Subscribers Currency: USD
Managed By: Administrator only

General Info Maintenance Fee Online Signup Accessibility Subscriptions Notepad

Edit	Service Type	Node	Access Code	Info Digits	Tariff	Delete
	NOT SELECTED	ANY		ANY		
	Voice Calls	DemoSIP			SIP Phone Subscribers	
	Voice Calls	PSTN-GW-NY-01			SIP Subscribers - Incoming	

Products

 Add

Voice Calls

Service Type

Access Code

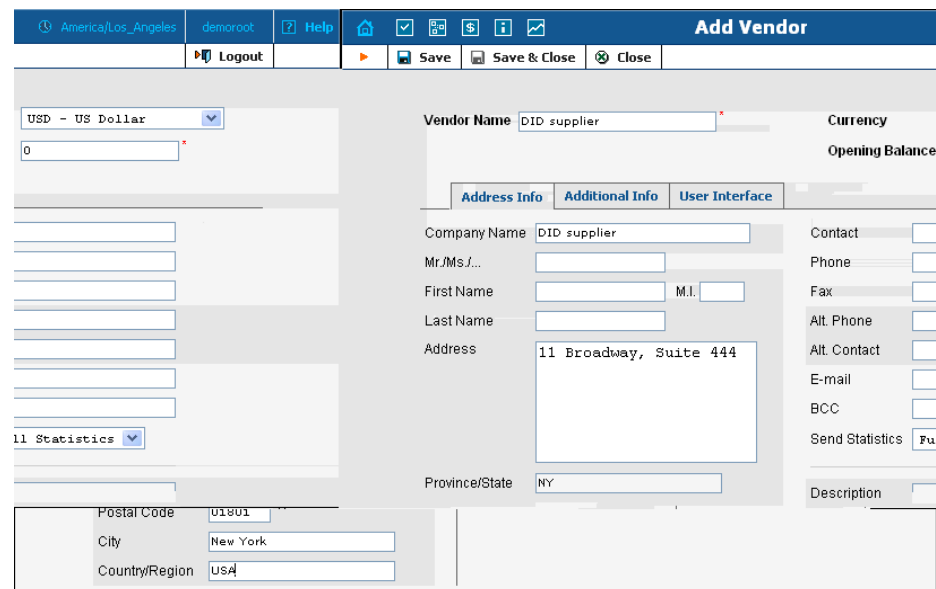
Info Digits

 Save

Create a DID Supplier Vendor

Vendors

 Add



Add Vendor

Basic SIP service

 Save & Close

Vendors

Define Connections

Connections

Vendors

The screenshot shows the 'Vendor Management' interface. The top section displays a list of vendors with columns: xDRs, Name, Connections, Currency, Balance, E-mail, and Delete. The 'DID supplier' vendor is highlighted. A red dashed arrow points from the 'DID supplier' row to the 'Vendor 'DID supplier' connections' interface below.

The 'Vendor 'DID supplier' connections' interface shows a table with columns: Node, Tariff, Port, CLD (dnis), Description, Info Digits, Transl. Rule, Capacity, and Delete. The 'PSTN from Vendor' tab is selected, and the 'DID supplier costs' entry is highlighted. A red dashed arrow points from the 'Add' button to the 'Node' field.

xDRs	Name	Connections	Currency	Balance	E-mail	Delete
ABC	ABC		USD	0.00000		
	DID supplier		USD	0.00000		
	GlobalNet		USD	0.06986	info@globalnet.com	
	MCI Vendor		USD	18.58369		
	SmartNetwork		USD	0.00000		
	SPT Telecom		USD	0.00000	info@spt.cz	
	Tele globe		USD	0.00000		
	X-Telecom		USD	374.37843	voip@x-telecom.com	

Node	Tariff	Port	CLD (dnis)	Description	Info Digits	Transl. Rule	Capacity	Delete
PSTN-CW-NY-01	ISDN 0%			ANY				
DID supplier costs	Incoming DID calls from SIP subscribers						24	

PSTN from Vendor

Add

Node

Port

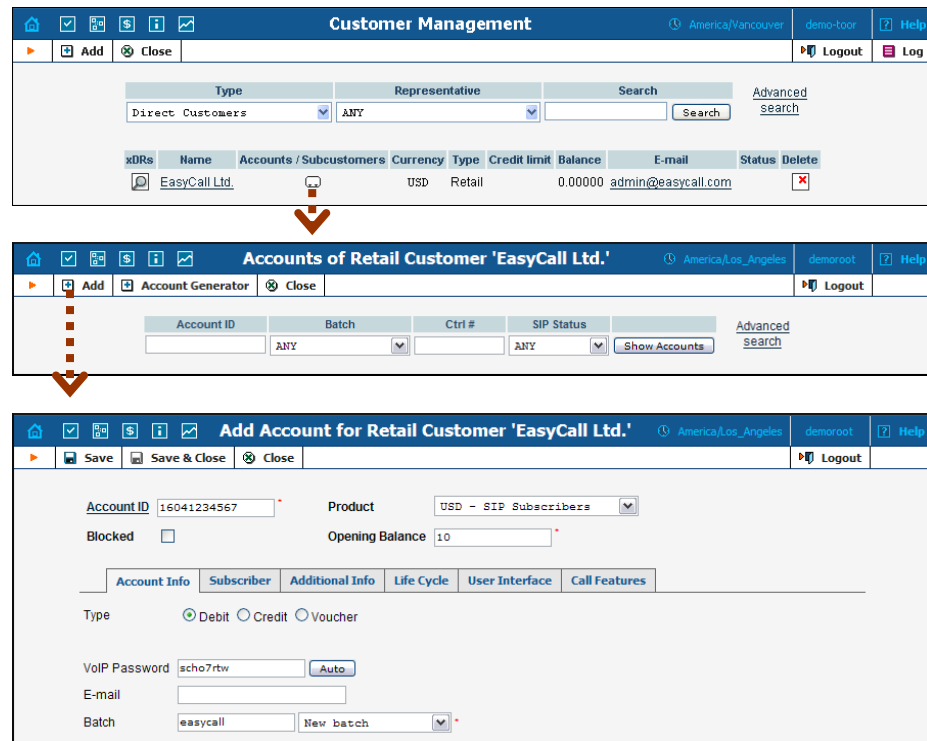
Description

Capacity

Save

Create Accounts

Customers



The first screenshot shows the 'Customer Management' page. It has a table with columns: xDRs, Name, Accounts / Subcustomers, Currency, Type, Credit limit, Balance, E-mail, Status, and Delete. A row for 'EasyCall Ltd.' is visible with a dropdown arrow next to the 'Accounts / Subcustomers' column.

The second screenshot shows the 'Accounts of Retail Customer 'EasyCall Ltd.' page. It has a table with columns: Account ID, Batch, Ctrl #, and SIP Status. A 'Show Accounts' button is visible.

The third screenshot shows the 'Add Account for Retail Customer 'EasyCall Ltd.' page. It has fields for: Account ID (16041234567), Product (USD - SIP Subscribers), Blocked (checkbox), Opening Balance (10), Type (radio buttons for Debit, Credit, Voucher), VoIP Password (scho7rtw), E-mail, and Batch (easycall). There are tabs for Account Info, Subscriber, Additional Info, Life Cycle, User Interface, and Call Features.

Accounts

+ Add

- Account ID
- Product
- Blocked
- Opening Balance

Account Info tab

- Account Type
- Credit Limit
- VoIP Password
- Email

- **Batch**

Additional Info tab

- **IP Phone, IP Phone Port**

Phones

Setting up Auto-provisioning of IP

Life Cycle tab

- **Activation Date**
- **Expiration Date**

- **Lifetime**

User Interface tab

- **Login**

- **Password**
- **Time Zone**

- **Web Interface Language**

Call Features tab

- **Preferred IVR Language**

- **Associated Number**

- **UM Enabled**

- **Forward Mode**

-
-
-

-

Forward

Follow-me

-

○

Timeout, sec

Save&Close

Incoming DID calls (from a VoIP vendor)

Checklist



Operation	Done
Initial configuration	
	Basic SIP service
	Basic SIP service
Rating configuration	
	Routing
Account provisioning	

Basic SIP service

Testing

Set up a PSTN-to-SIP Service



Create a Tariff to Charge Your Subscribers

[illegible]

Tariffs

 **Add**

Add Tariff

- Name
- Currency

NOTE: The currency for the tariff may be chosen only once, and cannot be changed later.

- Applied To Customer
- Managed By Administrator Only
Applied to: Customer
- Service Type Voice Calls
- Off-peak Period



Help

- Off-peak Description
- Destination Group Set
- Free Seconds
- Post-Call Surcharge
- Login Fee
- Connect Fee
- Round Charged Amount
- Default Formula
- Short Description

- **Description**
 **Save**

Enter Rates

System Concepts

Managing rates online

'SIP Subscribers - Incoming' tariff rates

Effective From: Destination:

Now

Edit	Destination *	Country	Interval, second		Price, USD/ minute		Effective From				Delete
	Description		First *	Next *	First *	Next *	YYYY-MM-DD HH24:MI:SS *				
	<input type="text"/>		<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/> <input type="text"/>		<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

 Rates

Rates

Edit Rates

 **Add**

- **Destination**


Destination

NOTE: The phone prefix you are trying to create a rate for must already exist in Destinations.

- **Interval First**
- **Interval Next**
- **Price First**

- Price Next
- Off-peak Interval First
- Off-peak Interval Next
- Off-peak Price First
- Off-peak Price Next

NOTE: Off-peak fields appear only if an **off-peak period** has been defined for the tariff.

- Rating Formula Wizard 
- Effective from

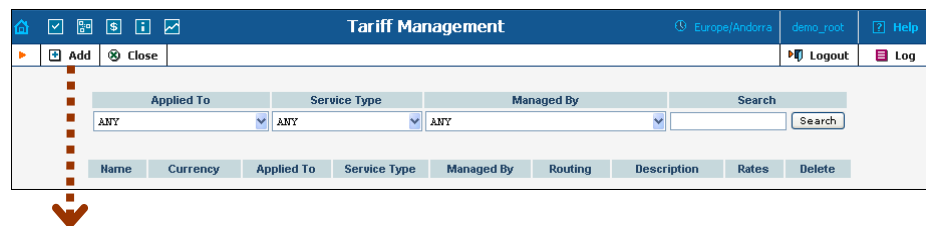
NOTE: When using the calendar, you can specify that the date you are entering is in a different time zone than your present one. PortaBilling will then automatically adjust the time.

- Hidden Forbidden Discontinued 
-  Save

Uploading a rate list from a file

PortaBilling Templates Guide

Create a Tariff for Incoming DID Costs



The screenshot shows the 'Tariff Management' interface. At the top, there's a navigation bar with icons for home, check, add, save, and print. Below this is a header with 'Tariff Management' and user information 'Europe/Andorra demo_root Help'. A sub-header contains 'Add', 'Close', and 'Logout' buttons. The main form has four dropdown menus: 'Applied To' (set to ANY), 'Service Type' (set to ANY), 'Managed By' (set to ANY), and a 'Search' field. Below the form is a table with the following columns: Name, Currency, Applied To, Service Type, Managed By, Routing, Description, Rates, and Delete. A red arrow points to the 'Add' button.

Add Tariff

Name: DID supplier costs * Currency: USD - US Dollar

Applied To: Vendor * Service Type: Voice Calls

Routing: ☐

General Info

Off-Peak Period:

Off-Peak Description:

Destination Group Set:

Free Seconds:

Post Call Surcharge: %

Login Fee:

Connect Fee:

Round Charged Amount:

Default Formula:

Short Description: DID Supplier

Description: What we are being charged by our DID supplier for incoming calls

Tariffs

Add

Add Tariff

Applied To
Routing

Vendor

Save

Enter Rates

'DID supplier costs' tariff rates

Effective From: Now Destination:

Edit	Destination *	Country	Interval, second		Price, USD/minute		Effective From	Delete
			First *	Next *	First *	Next *		
<input type="checkbox"/>	4722	NORWAY Oslo	60	60	0.005	0.005	immediately	<input type="checkbox"/>

Rates

Rates

Edit Rates

Add



Modify the Accessibility for the Product

Product Management America/Los_Angeles demoproot Help

Add Close

Managed By: ANY Search

Icon	Name	Currency	Managed By	Description	Delete
	CT Product	USD	Administrator only	Cisco Test Product	
	PortaOne	USD	Administrator only		
	Prepaid Cards	USD	Administrator only	Prepaid Cards	
	SIP Subscribers	USD	Administrator only	Product for SIP users. No monthly fee.	
	SmartCall	USD	Administrator only	SmartCall Prepaid cards	
	Smartnet Termination	USD	Administrator only	What we charge Customer Smartnet for calls they terminate to us	

Edit 'SIP Subscribers' Product America/Vancouver demo-toor Help

Add Save Save & Close Close Rate Lookup Logout Log

Product Name: SIP Subscribers Currency: USD

Managed By: Administrator only

General Info Maintenance Fee Online Signup Accessibility Subscriptions Notepad

Edit	Service Type *	Node	Access Code	Info Digits	Tariff *	Delete
	NOT SELECTED	ANY		ANY		
	Voice Calls	DemoSIP			SIP Phone Subscribers	

Edit 'SIP Subscribers' Product America/Vancouver demo-toor Help

Add Save Save & Close Close Rate Lookup Logout Log

Product Name: SIP Subscribers Currency: USD

Managed By: Administrator only

General Info Maintenance Fee Online Signup Accessibility Subscriptions Notepad

Edit	Service Type *	Node	Access Code	Info Digits	Tariff *	Delete
	Voice Calls	DemoSIP	INCOMING	ANY	SIP Subscribers -	
	Voice Calls	DemoSIP			SIP Phone Subscribers	

Edit 'SIP Subscribers' Product America/Vancouver demo-toor Help

Add Save Save & Close Close Rate Lookup Logout Log

Product Name: SIP Subscribers Currency: USD

Managed By: Administrator only

General Info Maintenance Fee Online Signup Accessibility Subscriptions Notepad

Edit	Service Type *	Node	Access Code	Info Digits	Tariff *	Delete
	NOT SELECTED	ANY		ANY		
	Voice Calls	DemoSIP			SIP Phone Subscribers	
	Voice Calls	DemoSIP	INCOMING		SIP Subscribers - Incoming	

Products

 Add

Voice Calls

Service Type

INCOMING
Info Digits

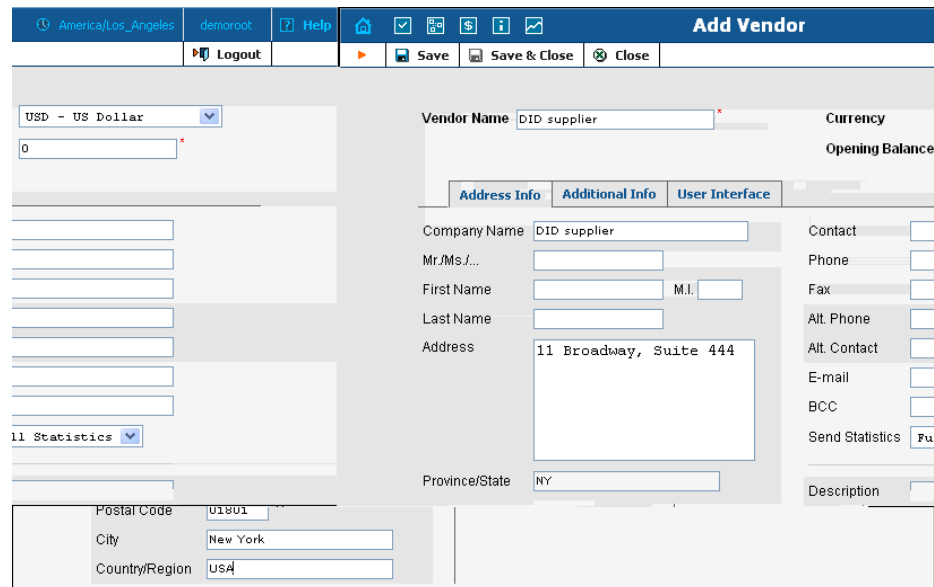
Access Code

 Save

Create a DID Supplier Vendor

Vendors

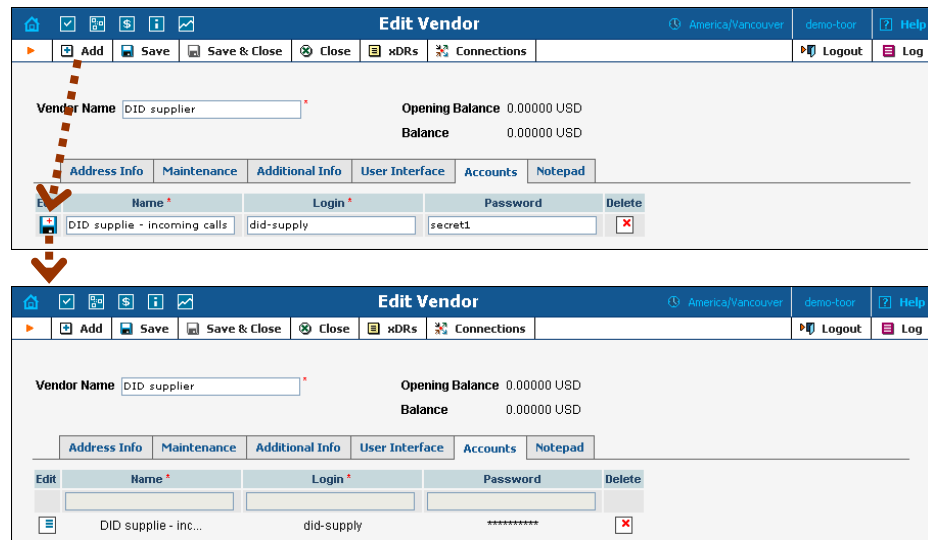
 Add



Add Vendor
Basic SIP service

 Save

 Add



Edit Vendor America/Vancouver demo-tool Help

Vendor Name: DID supplier Opening Balance: 0.00000 USD Balance: 0.00000 USD

Address Info Maintenance Additional Info User Interface Accounts Notepad

Edit	Name *	Login *	Password	Delete
	DID supplie - incoming calls	did-supply	secret1	

Edit Vendor America/Vancouver demo-tool Help

Vendor Name: DID supplier Opening Balance: 0.00000 USD Balance: 0.00000 USD

Address Info Maintenance Additional Info User Interface Accounts Notepad

Edit	Name *	Login *	Password	Delete
	DID supplie - inc...	did-supply	*****	

- Name



- Login

cisco

- Password

Save




















Close




Vendors

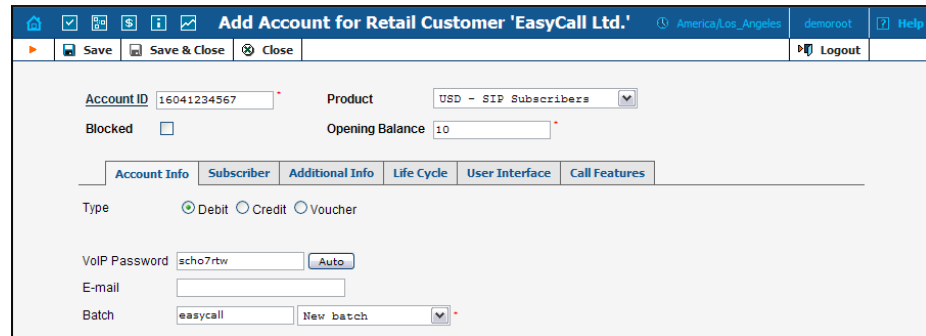
Define Connections

Connections

Vendors

Vendor Management						
<div> <div> <div></div> <div></div> <div></div> <div></div> <div></div> </div> <div> <div></div> <div></div> <div></div> <div></div> <div></div> </div> </div> <div> <div></div> <div></div> <div></div> <div></div> <div></div> </div>					America/Vancouver	
					demo-tool	Help
<div> <div></div> <div>Add</div> <div>Close</div> </div>					Logout	Log
<div> <div>Search</div> <div> <input type="text"/> <div>Search</div> </div> </div>						
xDRs	Name	Connections	Currency	Balance	E-mail	Delete
	ABC		USD	0.00000		
	DID supplier		USD	0.00000		
	GlobalNet		USD	0.06986	info@globalnet.com	
	MCI Vendor		USD	18.58369		
	SmartNetwork		USD	0.00000		
	SPT Telecom		USD	0.00000	info@spt.cz	
	Telelobe		USD	0.00000		
	X-Telecom		USD	374.37843	voip@xtelecom.com	

Vendor 'DID supplier' connections						
<div> <div></div> <div>Add</div> <div>Save</div> <div>Save & Close</div> <div>Close</div> </div>					America/Vancouver	
					demo-tool	Help
					Logout	Log
<div> <div>PSTN from Vendor</div> <div>VoIP from Vendor</div> <div>PSTN to Vendor</div> <div>VoIP to Vendor</div> </div>						
<div> <div>Edit</div> <div>Load</div> </div>	Node *		RTP Proxying	Transl. Rule	Account *	Delete
	Tariff *		Description *		Capacity *	
	DemoSIP		Optimal		DID supplie - is	
	DID supplier costs		Incoming DIDs from supplier		60	




Accounts

+ Add

- Account ID
- Product
- Blocked
- Opening Balance

Account Info tab

- Account Type
- Credit Limit
- VoIP Password
- Email
- Batch

Additional Info tab

- IP Phone, IP Phone Port

Setting up Auto-provisioning of IP

Life Cycle tab

- Activation Date
- Expiration Date

- Lifetime

User Interface tab

- Login

- Password
- Time Zone

- Web Interface Language

Call Features tab

- Preferred IVR Language

- Associated Number
- UM Enabled
- Forward Mode

-
-
-

-

Forward

Follow-me

-

- Timeout, sec

 Save&Close

3. Setting up Auto-provisioning of IP Phones

[PortaBilling Web Reference Guide](#)

Setting up Auto-provisioning of IP Phones

Checklist

Operation	Done
General configuration	

IP phone inventory

Assigning a phone number to the IP phone

IP phone settings

Testing

Create an IP Phone Profile

The screenshot shows the 'IP Phone Profiles' management interface. At the top, there's a toolbar with 'Add' and 'Close' buttons. Below it is a table with columns: Effective From, Name, Type, Managed By, Discontinued, Description, and Delete. A red dashed arrow points from the 'Add' button in the toolbar to the 'Add IP Phone Profile' form below. The form contains the following fields:

- Name: Sipura - Standard
- Managed By: Administrator only
- Type: Sipura 2000
- As Copy Of: None
- Effective From: Date: immediately, Time: HH24:MI:SS

IP
Phone Profiles.

Add

- Name
- Managed By

Administrator Only

- Type
- As Copy Of

None

- Effective From
immediately
- Save.

IP Phone Profile Settings

Save

GPP A, GPP B, GPP C	<div>Rule</div> <div>Profile</div>

There is no need to change any of the values for dynamic variables.

Line 1 Line 2

Field	Description
Proxy	
Outbound Proxy	
Use Outbound Proxy	
Register	

Register Expires	
Display Name	
User ID	
Auth ID	
Use Auth ID	
Preferred Codec	
Use Pref Codec Only	
DTMF Tx Method	
Dial Plan	

Create an IP Phone Entry



The screenshot shows the 'IP Phone Inventory' management interface. At the top, there's a navigation bar with 'America/Vancouver', 'demo-root', and 'Help'. Below it, a toolbar contains 'Add' and 'Close' buttons. The main area features a search bar with 'Type' and 'Status' dropdowns set to 'ANY', and a 'Search' button. Below the search bar is a table with the following columns: Name, Type, Profile, Managed By, MAC Address, Description, Ports free, Ports total, Inventory ID, and Delete. The table contains two entries: 'My SPA-941' (Linksys SPA-941, SPA-941 profile, Administrator only, MAC 0:E:8:DB:F8:CE, Description 'My office Linksys SPA-941', 2 free/2 total ports, Inventory ID 'KZ Linksys', and a delete icon) and 'SPA 2000_117' (Sipura 2000, Sipura 2000 profile, Administrator only, MAC 0:50:56:C0:0:8, Description 'My office Linksys SPA-941', 1 free/2 total ports, Inventory ID '1', and a delete icon). A red dashed arrow points from the 'Add' button in the top toolbar to the 'Add' button in the table's toolbar.

Name	Type	Profile	Managed By	MAC Address	Description	Ports free	Ports total	Inventory ID	Delete
My SPA-941	Linksys SPA-941	SPA-941	Administrator only	0:E:8:DB:F8:CE	My office Linksys SPA-941	2	2	KZ Linksys	
SPA 2000_117	Sipura 2000	Sipura 2000	Administrator only	0:50:56:C0:0:8	My office Linksys SPA-941	1	2	1	

Add IP Phone

Name: JB SPA-2000 Description: J. Brown SPA-2000

Managed by: Administrator only Type: Sipura 2000

General Info

Profile: Sipura - Standard

MAC Address: 00:0E:08:AB:D7:A8

Ports: Total 2 Free 2

ASCII Key:

Inventory ID: JB Sipura

IP Phone Inventory

Type: ANY Status: ANY Search:

Name	Type	Profile	Managed By	MAC Address	Description	Ports free total	Inventory ID	Delete
JB SPA-2000	Sipura 2000	Sipura - Standard	Administrator only	0:E:8:AB:D7:A8	J. Brown SPA-2000	2 2	JB Sipura	<input type="checkbox"/>
My SPA-941	Linksys SPA-941	SPA-941	Administrator only	0:E:8:DB:F8:CE	My office Linksys SPA-941	2 2	KZ Linksys	<input type="checkbox"/>
SPA 2000_117	Sipura 2000	Sipura 2000	Administrator only	0:50:56:C0:0:8		1 2	1	

IP
Phone Inventory.

+ Add

- Name
 - Type
 - Profile
 - MAC Address
- Ports
 - Save & Close

Provisioning an Account on an IP Phone

Additional Info

Account Info / Retail Customer 'EasyCall Ltd.'

Account ID: 16041234570 Product: USD - SIP Subscribers

Blocked: ☐ Balance: 0.00000 USD

User Agent: Contact:

Account Info Maintenance Subscriber Additional Info Life Cycle User Interface Call Features Subscriptions Notepad

IP Phone: JB SPA-2000

IP Phone Port: 1

E-commerce Enabled: ☐

Discount Plan: Product default

IP Phone

IP Phone Port

 Save & Close

Note: The **IP Phone** select field shows a list of phones that have not been used before in other accounts, or phones with available (unused) ports.

admin/apache/htdocs/ */home/porta-*

IP Phone Device Configuration

Admin login
Provisioning

Advanced



Address <http://192.168.0.237/admin/advanced> [Go](#) [Links](#)

technology, inc. **Sipura Phone Adapter Configuration**

[Info](#) [System](#) [SIP](#) **Provisioning** [Regional](#) [Phone](#) [Line 1](#) [Line 2](#) [User 1](#) [User 2](#) [User Login](#) [basic](#) | [advanced](#)

Configuration Profile

Provision Enable: ☐ yes ☐ no Resync On Reset: ☐ yes ☐ no

Resync Random Delay: Resync Periodic:

Resync Error Retry Delay: Forced Resync Delay:

Resync From SIP: ☐ yes ☐ no Resync After Upgrade Attempt: ☐ yes ☐ no

Resync Trigger 1:

Resync Trigger 2:

Resync Fails On FNF: ☐ yes ☐ no

Profile Rule:

Profile Rule B:

Profile Rule C:

Profile Rule D:

Log Resync Request Msg:

Log Resync Success Msg:

Log Resync Failure Msg:

Report Rule:

Firmware Upgrade

Upgrade Enable: ☐ yes ☐ no Upgrade Error Retry Delay:

Downgrade Rev Limit:

Upgrade Rule:

Log Upgrade Request Msg:

Log Upgrade Success Msg:

Log Upgrade Failure Msg:

General Purpose Parameters

GPP A:

GPP B:

- **Profile Rule**

```
[--key $B]http://PB_SLAVE_SERVER/$A/$MA.cfg
```

- **GPP A**

-

GPP B

Submit All Changes

Advanced Provisioning Tips

```

admin/profile                                     /usr/home/porta-

admin/apache                                     /usr/home/porta-
admin/apache/htdocs                             /usr/home/porta-

```

home/porta-admin/apache/htdocs

```

DocumentRoot

/usr/local/etc/apache/porta.httpd.conf:
<VirtualHost _default_:80>
    DocumentRoot "/home/porta-admin/apache/htdocs/"
    Options ExecCGI
    DirectoryIndex index.pl
</VirtualHost>

```

Note: If you change this value you must make corresponding changes in an additional list of configuration files. Do not forget to restart the Apache server afterwards.

```

[UA_Profiles] ResultDir
/usr/home/porta-admin/etc/porta-admin.conf

[UA_Profiles] Dir
/usr/home/porta-admin/etc/porta-admin.conf

/etc/inetd.conf

```

4. Appendices

APPENDIX A. Clients' Cisco ATA 186 Configuration for PortaSIP

UID0	
PWD0	
UID1	
PWD1	
GkOrProxy	
Gateway	
GateWay2	
UseLoginID	
LoginID0	
LoginID1	
AltGK	
AltGKTimeOut	
GkTimeToLive	
GkId	
UseSIP	
SIPRegInterval	
MaxRedirect	
SIPRegOn	
NATIP	
SIPPort	
MediaPort	
OutBoundProxy	
NatServer	
NatTimer	
LBRCodec	
AudioMode	
RxCCodec	
TxCCodec	
NumTxFrames	
CallFeatures	
PaidFeatures	
CallerIdMethod	
FeatureTimer	
Polarity	
ConnectMode	
AuthMethod	
TimeZone	
NTPIP	

AltNTPIP	
DNS1IP	
DNS2IP	
UDPTOS	
SigTimer	
OpFlags	
VLANSettings	
NPrintf	
TraceFlags	

APPENDIX B. Configure Cisco ATA Using ATA Expert

<http://<ata-IP-address>/dev>

0x00150015

http://www.cisco.com/en/US/products/hw/gatecont/ps514/products_configuration_example09186a00800c3a50.shtml

Address <http://192.168.0.16/dev> Go Links

Cisco ATA 186 Configuration

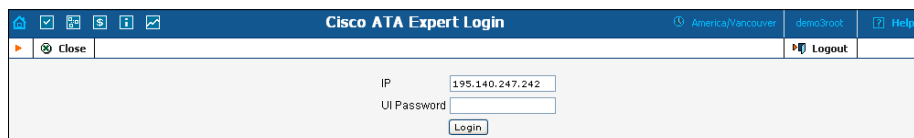
UIPassword:	•	ToConfig:	1
UseTtp:	1	TtpURL:	0
CfgInterval:	3600	EncryptKey:	•
Dhcp:	0	StaticIP:	192.168.0.16
StaticRoute:	192.168.0.3	StaticNetMask:	255.255.255.0
UID0:	12027810001	PWD0:	••••••
UID1:	0	PWD1:	•
GkOrProxy:	demo.portaone.com	Gateway:	0
GateWay2:	0.0.0.0	UseLoginID:	0
LoginID0:	0	LoginID1:	0
AltGk:	0	AltGkTimeOut:	0
GkTimeToLive:	300	GkId:	.
UseSIP:	1	SIPRegInterval:	180
MaxRedirect:	5	SIPRegOn:	1
NATIP:	0.0.0.0	SIPPort:	5060
MediaPort:	16384	OutBoundProxy:	demo.portaone.com
NatServer:	demo.portaone.com	NatTimer:	0x0000001e
LBRCCodec:	0	AudioMode:	0x00150015

Cisco ATA

Expert

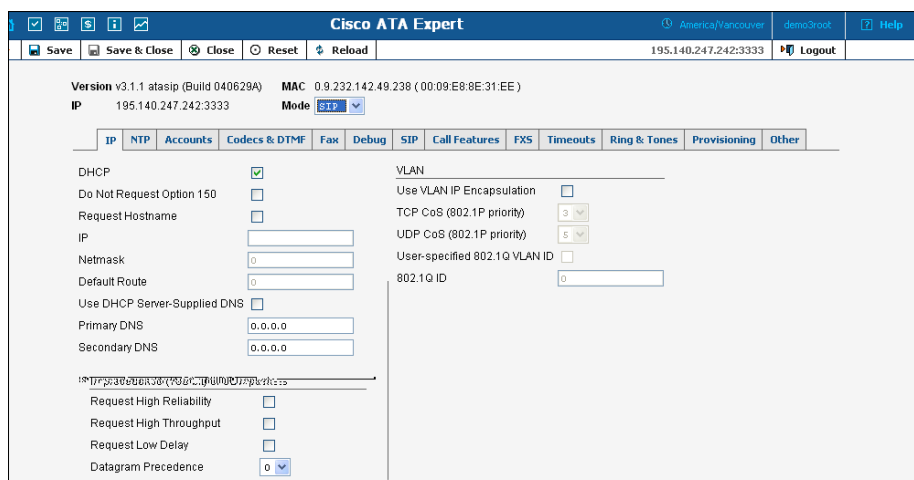
Cisco

ATA Expert.

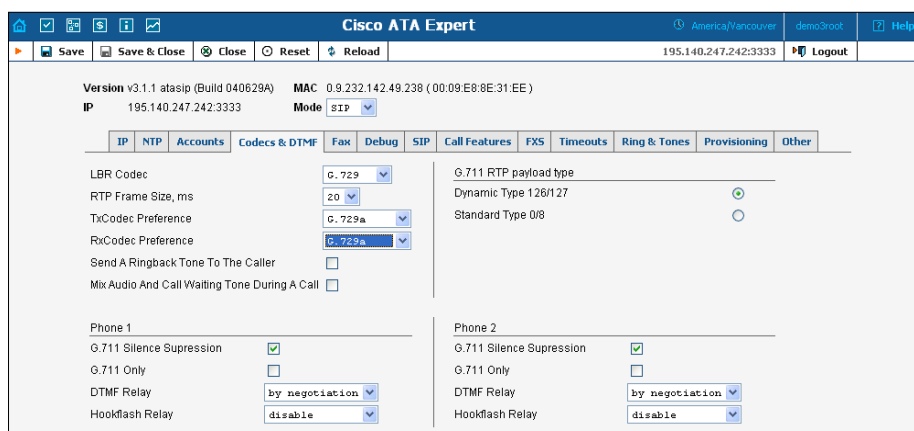


The login screen for the Cisco ATA Expert interface. It features a blue header bar with the title "Cisco ATA Expert Login" and navigation links for "America/Vancouver", "demo3root", and "Help". Below the header, there is a "Close" button on the left and a "Logout" button on the right. The main area contains two input fields: "IP" with the value "195.140.247.242" and "UI Password" which is empty. A "Login" button is positioned below the password field.

NOTE: The PortaBilling ATA Expert needs to communicate directly with the Cisco ATA. So make sure that the ATA is connected to the network and configured with an IP address. This IP address must be either a public IP address (accessible from anywhere on the Internet) or a private IP address (e.g. 192.168.xxx.xxx) which is accessible from the PortaBilling web server.



The configuration screen for the Cisco ATA Expert, showing the "IP" tab. The header bar includes "Cisco ATA Expert" and navigation links. Below the header, there are buttons for "Save", "Save & Close", "Close", "Reset", and "Reload". The main area displays the current IP address "195.140.247.242:3333" and the MAC address "0.9.232.142.49.238 (00:09:E8:8E:31:EE)". The "Mode" is set to "SIP". The "IP" tab is selected, showing various configuration options for DHCP, VLAN, and QoS. The "DHCP" section includes checkboxes for "Do Not Request Option 150", "Request Hostname", "Use DHCP Server-Supplied DNS", and "Request High Reliability", "Request High Throughput", "Request Low Delay", and "Datagram Precedence". The "VLAN" section includes checkboxes for "Use VLAN IP Encapsulation" and "User-specified 802.1Q VLAN ID". The "QoS" section includes dropdowns for "TCP CoS (802.1P priority)", "UDP CoS (802.1P priority)", and "802.1Q ID".



The configuration screen for the Cisco ATA Expert, showing the "SIP" tab. The header bar includes "Cisco ATA Expert" and navigation links. Below the header, there are buttons for "Save", "Save & Close", "Close", "Reset", and "Reload". The main area displays the current IP address "195.140.247.242:3333" and the MAC address "0.9.232.142.49.238 (00:09:E8:8E:31:EE)". The "Mode" is set to "SIP". The "SIP" tab is selected, showing various configuration options for LBR Codec, RTP Frame Size, TxCodec Preference, RxCodec Preference, Send A Ringback Tone To The Caller, Mix Audio And Call Waiting Tone During A Call, and Phone 1 and Phone 2 settings. The "LBR Codec" section includes dropdowns for "LBR Codec", "RTP Frame Size", "TxCodec Preference", and "RxCodec Preference". The "Send A Ringback Tone To The Caller" and "Mix Audio And Call Waiting Tone During A Call" sections include checkboxes. The "Phone 1" and "Phone 2" sections include checkboxes for "G.711 Silence Suppression", "G.711 Only", and "DTMF Relay", and dropdowns for "Hookflash Relay".

 Save

APPENDIX C. Client's Sipura Configuration for PortaSIP

Line 1 Line 2

Proxy and Registration

Proxy

Register

Subscriber

Display Name

User ID

Password

Use Auth ID

SIPURA

technology, inc.

Sipura Phone Adapter Configuration

Info

System

SIP

Provisioning

Regional

Line 1

Line 2

User 1

User 2

[User Login](#)

[basic](#)

[advanced](#)

System Information

DHCP:

Host Name:

Current Netmask:

Primary DNS:

Secondary DNS:

Enabled

SipuraSPA

255.255.255.0

192.168.0.192

207.102.99.66 207.102.99.82

Current IP:

Domain:

Current Gateway:

192.168.0.88

portaone.com

192.168.0.192

Product Information

Product Name:

Software Version:

MAC Address:

SPA-2000

2.0.10(e)

000E08AB4638

Serial Number:

Hardware Version:

Client Certificate:

88012BA66086

2.0.1(0905)

Installed

System Status

Current Time:

Broadcast Pkts Sent:

Broadcast Pkts Recv:

Broadcast Pkts Dropped:

RTP Packets Sent:

RTP Packets Recv:

SIP Messages Sent:

SIP Messages Recv:

External IP:

1/8/2003 14:17:56

0

560688

0

3074

2341

1724

362

Elapsed Time:

Broadcast Bytes Sent:

Broadcast Bytes Recv:

Broadcast Bytes Dropped:

RTP Bytes Sent:

RTP Bytes Recv:

SIP Bytes Sent:

SIP Bytes Recv:

4 days and 02:23:13

0

34980083

0

120568

54292

1167889

166405

Line 1 Status

Hook State:

Last Registration At:

Message Waiting:

Last Called Number:

Mapped SIP Port:

On

1/8/2003 14:07:33

No

16044680035

Registration State:

Next Registration In:

Call Back Active:

Last Caller Number:

Registered

2947 s

No

Call 1 State:

Call 1 Tone:

Call 1 Encoder:

Call 1 Decoder:

Call 1 FAX:

Call 1 Type:

Call 1 Remote Hold:

Call 1 Callback:

Call 1 Peer Name:

Call 1 Peer Phone:

Idle

None

Call 2 State:

Call 2 Tone:

Call 2 Encoder:

Call 2 Decoder:

Call 2 FAX:

Call 2 Type:

Call 2 Remote Hold:

Call 2 Callback:

Call 2 Peer Name:

Call 2 Peer Phone:

Idle

None

Network Settings	
SIP TOS/DiffServ Value:	0x68
RTP TOS/DiffServ Value:	0xb8
Network Jitter Level:	high
SIP Settings	
SIP Port:	5060
EXT SIP Port:	
SIP Debug Option:	none
SIP 100REL Enable:	no
Auth Resync-Reboot:	yes
Call Feature Settings	
Blind Attn-Xfer Enable:	no
Xfer When Hangup Conf:	yes
MOH Server:	
Proxy and Registration	
Proxy:	216.231.44.168
Outbound Proxy:	
Register:	yes
Register Expires:	3600
Use DNS SRV:	no
Proxy Fallback Intvl:	3600
Use Outbound Proxy:	no
Use OB Proxy In Dialog:	yes
Make Call Without Reg:	no
Ans Call Without Reg:	no
DNS SRV Auto Prefix:	no
Subscriber Information	
Display Name:	
Password:	*****
Auth ID:	
Mini Certificate:	
SRTP Private Key:	
User ID:	1206001236
Use Auth ID:	no
Supplementary Service Subscription	
Call Waiting Serv:	yes
Block ANC Serv:	yes
Cfwd All Serv:	yes
Cfwd No Ans Serv:	yes
Cfwd Last Serv:	yes
Accept Last Serv:	yes
CID Serv:	yes
Call Return Serv:	yes
Three Way Call Serv:	yes
Attn Transfer Serv:	yes
Block CID Serv:	yes
Dist Ring Serv:	yes
Cfwd Busy Serv:	yes
Cfwd Sel Serv:	yes
Block Last Serv:	yes
DND Serv:	yes
CWCID Serv:	yes
Call Back Serv:	yes
Three Way Conf Serv:	yes
Unattn Transfer Serv:	yes

APPENDIX D. SJLabs Softphone Configuration for PortaSIP

Service: PortaOne

Please enter this information to initialize the service profile

Account: 123456789

Password: *****

☒ Save service information permanently

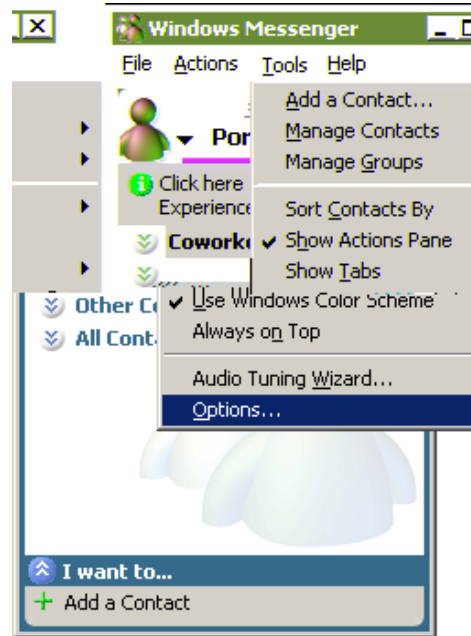
OK

Cancel

Help

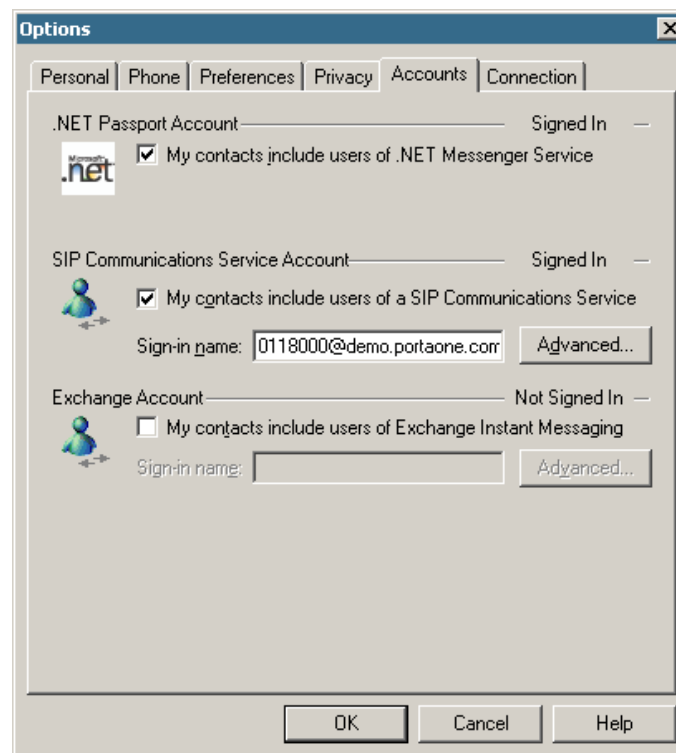


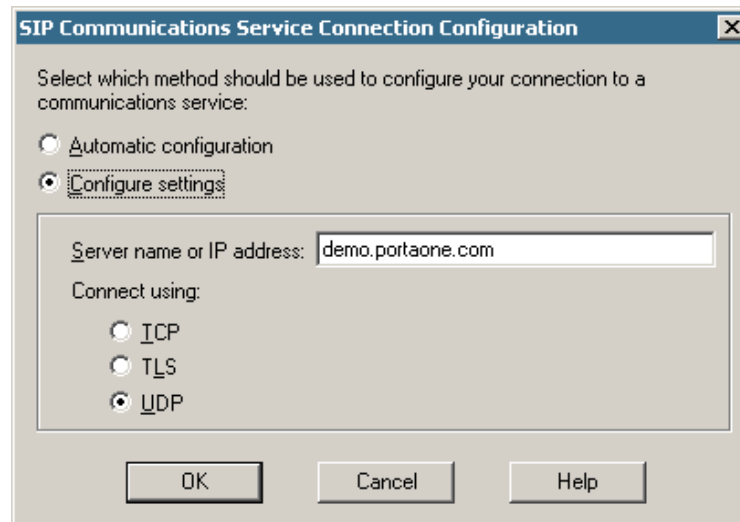
APPENDIX E. Configuring Windows Messenger for Use as a SIP User Agent



username@address

*username
address*





SIP Communications Service Connection Configuration

Select which method should be used to configure your connection to a communications service:

☐ Automatic configuration

☒ **Configure settings**

Server name or IP address:

Connect using:

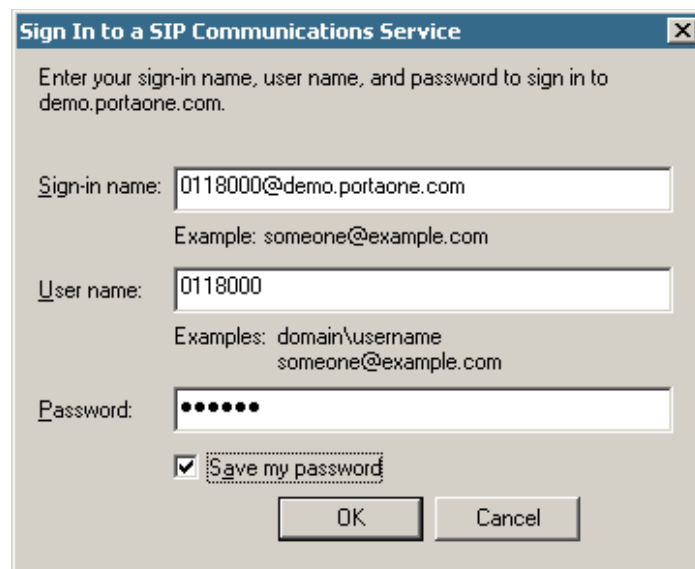
☐ ICP

☐ TLS

☒ **UDP**

OK Cancel Help

*username@address username
address*



Sign In to a SIP Communications Service

Enter your sign-in name, user name, and password to sign in to demo.portaone.com.

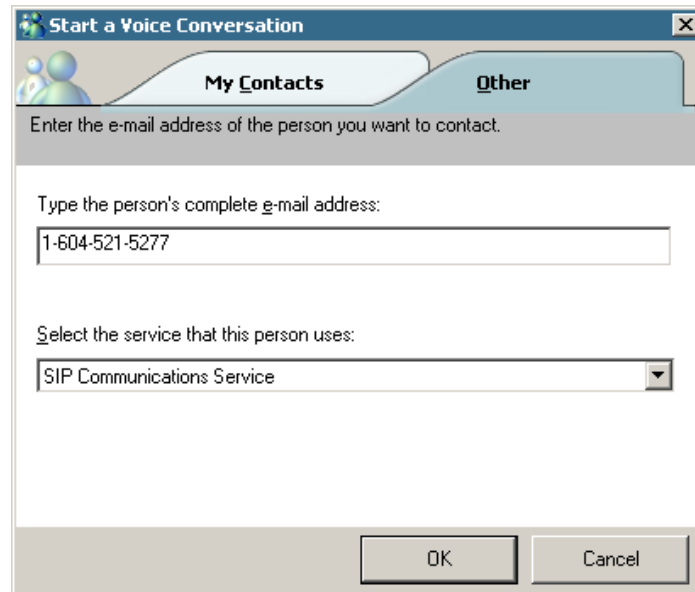
Sign-in name:
Example: someone@example.com

User name:
Examples: domain\username
someone@example.com

Password:

☒ **Save my password**

OK Cancel



APPENDIX F. Auto-provisioned IP Phones and Adapters

-
-
-
-
-
-
-
-
-
-