

**Porta**  **Switch**™



**PortaSwitch Handbook:  
SIP Services**

Maintenance Release 15

Part I

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**PortaSwitch Handbook: SIP Services, July 2007**  
**V.1.15.2**

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# 1. Setting up Standard SIP Services

PortaBilling Web Reference Guide

## Basic SIP service

### Checklist

<b>Operation</b>	<b>Done</b>
<b>General configuration</b>	

**Network configuration**

**Rating configuration**

**Routing**

**Account provisioning**

**Testing**

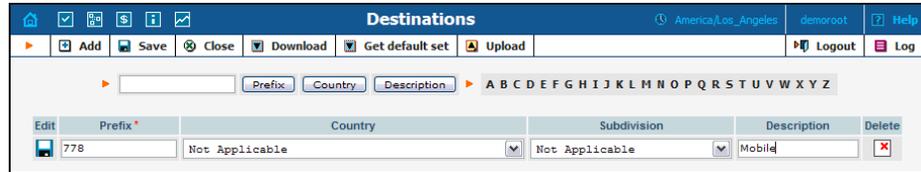


Creating destinations "one-by-one":

**Destinations**

 Add

**Description**



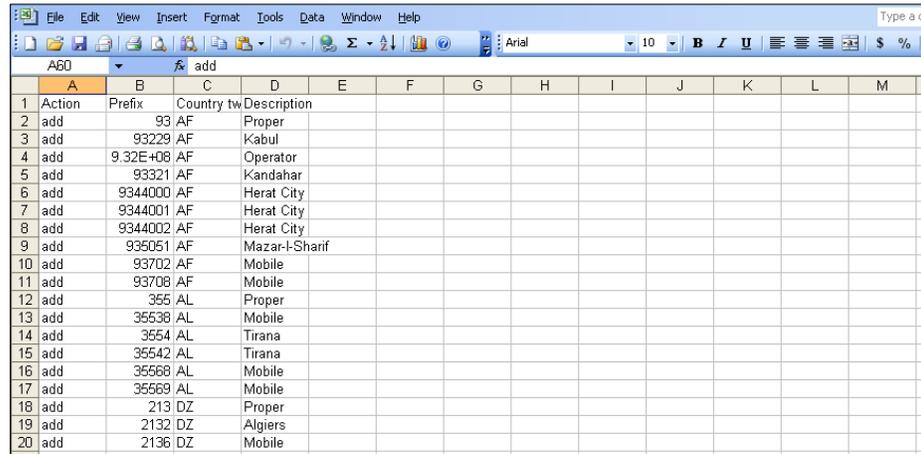
Prefix*	Country	Subdivision	Description	Delete
778	Not Applicable	Not Applicable	Mobile	

Save

Uploading a set of destinations from a file:

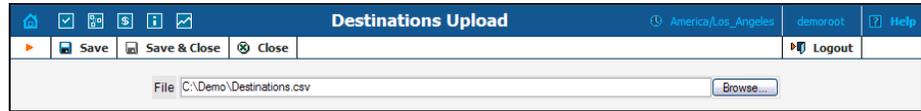
**Destinations**

**Default set**



	A	B	C	D	E	F	G	H	I	J	K	L	M
1	Action	Prefix	Country tw	Description									
2	add	93	AF	Proper									
3	add	93229	AF	Kabul									
4	add	9.32E+08	AF	Operator									
5	add	93321	AF	Kandahar									
6	add	9344000	AF	Herat City									
7	add	9344001	AF	Herat City									
8	add	9344002	AF	Herat City									
9	add	935051	AF	Mazar-I-Sharif									
10	add	93702	AF	Mobile									
11	add	93708	AF	Mobile									
12	add	355	AL	Proper									
13	add	35538	AL	Mobile									
14	add	3554	AL	Tirana									
15	add	35542	AL	Tirana									
16	add	35568	AL	Mobile									
17	add	35569	AL	Mobile									
18	add	213	DZ	Proper									
19	add	2132	DZ	Algiers									
20	add	2136	DZ	Mobile									

Upload 



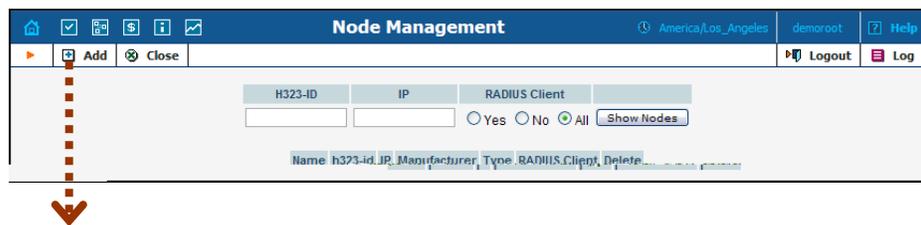
Browse...  
 Save & Close.

Destinations for SIP phones

1202781

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

Create Nodes





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  
Radius source IP      [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

- 
- 

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



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

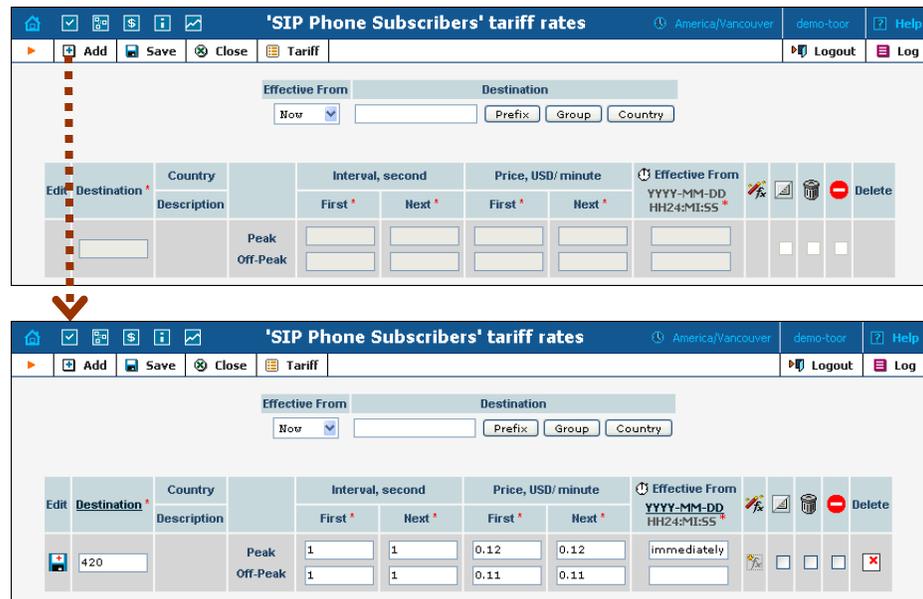
Vendor      Applied To  
Routing

# Enter Rates

*Call Billing Parameters*

## PortaBilling Administrator Guide

Managing rates online



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



Edit	Destination *	Country	Routing			Interval, second		Price, USD/minute		Effective From YYYY-MM-DD HHMMSS *				Delete
			Route Category	Preference	Huntstop	First *	Next *	First *	Next *					
	1604	British Columbia	Low-cost	6		1	1	0.60000	0.60000	2007-06-21 06:41:24				
	1	Not Applicable North America	Default	5		1	1	0.50000	0.50000	2007-06-21 06:42:07				
	44	UNITED KINGDOM Proper	Premium	3		1	1	0.18000	0.18000	2007-07-24 20:39:53				

- Route category

Default

- Preference

- Huntstop

Managing rates offline

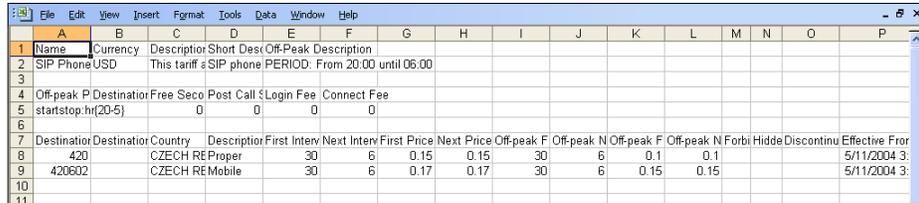
**NOTE: Templates** are available in PortaBilling, a powerful tool for uploading rates from custom format data files. However, in this particular example we assume that you will enter data using the PortaBilling default format.

Tariffs

Download

Now

File download



	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	h(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	Forbil	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:hf(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	Forbid	Hidden	Discontin	Effective Fror
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**



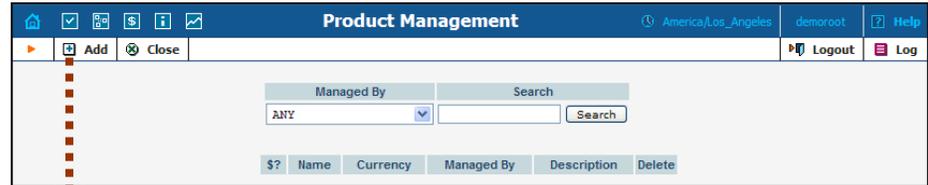
*Create Tariff    Enter Rates*

- 
- 
- 

*Managed by NNN*

*NNN*

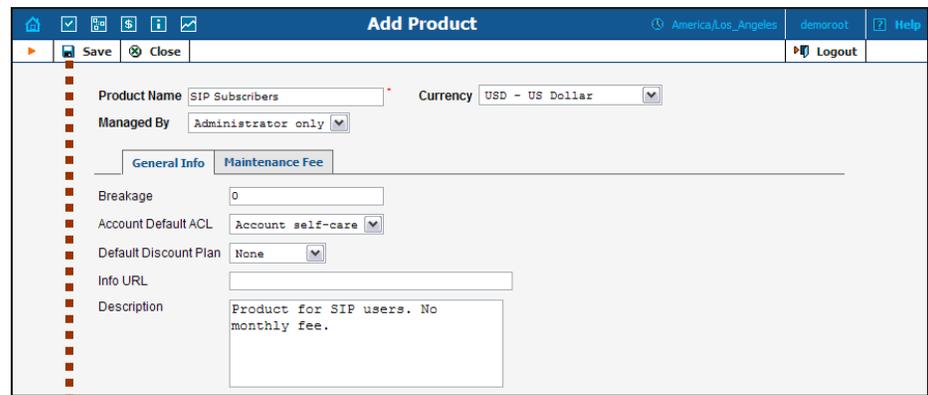
## Create Product



Product Management

Managed By: ANY Search

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



Add Product

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

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

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
  - None
- Breakage
- 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

		 Add
Voice Calls	Service Type	
Access code	Info Digits	
 Save		

Create Vendors

Vendors

 Add

**Add Vendor**

Main form (top)

- **Vendor name**
- **Currency**
- **Opening Balance**

Additional info:

- **Billing Period**

User Interface:

- o Time Zone



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	X

**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

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

Close

Vendors

Define Connections

Vendors

Connections

**Vendor Management** America/Vancouver demo-tool Help

Add Close

Search [ ] Search

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



PSTN to Vendor VoIP to Vendor

Add

Description

Capacity

Save

Close

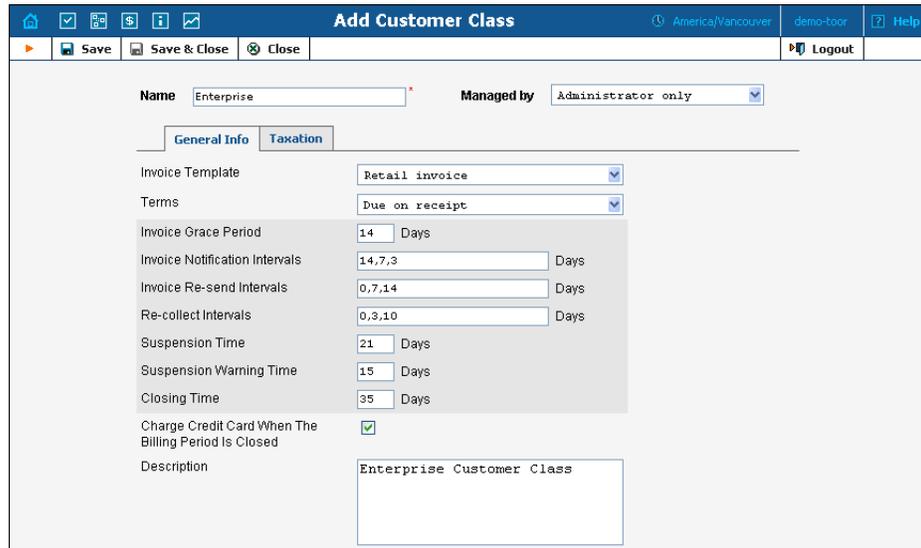
Vendor Management

## Create a Customer Class

Customer

Classes

Add



- Name
- Invoice Template

Do not create invoice

- Terms

[PortaBilling Web Reference Guide](#)

- Description

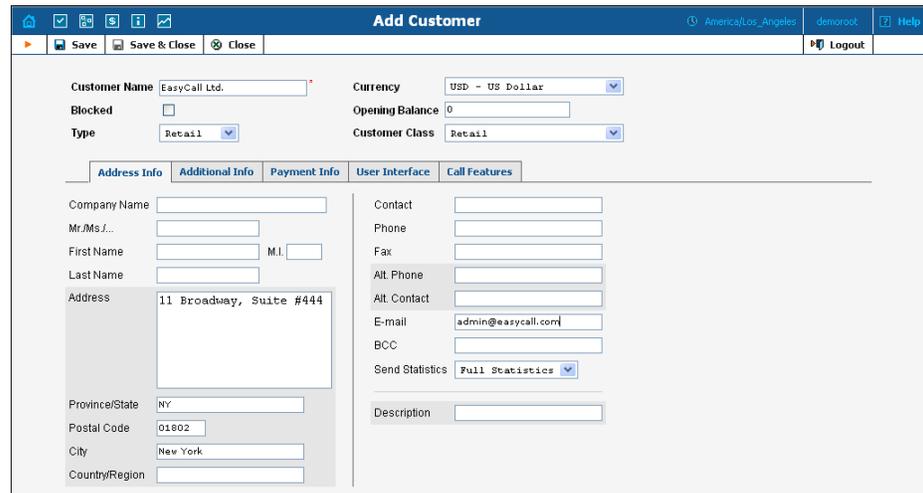
 Save&Close

## Create a Customer

### Customers

 Add

### New Customer



**Add Customer**

America/Los\_Angeles demoroot Help

Save Save & Close Close Logout

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./J...:   
 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

**Customer Management** America/Vancouver demo-root Help

Type: Direct Customers Representative: ANY Search:   [Advanced search](#)

xDRs	Name	Accounts / Subcustomers	Currency	Type	Credit limit	Balance	E-mail	Status	Delete
	EasyCall Ltd.		USD	Retail	0.00000		admin@easycall.com		

**Accounts of Retail Customer 'EasyCall Ltd.'** America/Los\_Angeles demoroot Help

Account ID:  Batch: ANY Ctrl #:  SIP Status: ANY  [Advanced search](#)

**Add Account for Retail Customer 'EasyCall Ltd.'** America/Los\_Angeles demoroot Help

Account ID: 16041234567 Product: USD - SIP Subscribers  
 Blocked:  Opening Balance: 10

[Account Info](#) [Subscriber](#) [Additional Info](#) [Life Cycle](#) [User Interface](#) [Call Features](#)

Type:  Debit  Credit  Voucher

VoIP Password: scho7rtw   
 E-mail:   
 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**



**Edit Customer** America/Vancouver demo-tour Help

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

Limit simultaneous calls: No  
 Set CLI to Account ID: No  
 Set CLI To Centrix:   
 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 Columbia (c) 2001 Revin

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

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**

E.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

**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 0 425 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



## # 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

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

Abbreviated Number Length 3

Abbreviated # *	# To Dial	Description	SIP	Delete
102				<input type="checkbox"/>
123	16049994321	Joe Brown		<input type="checkbox"/>

**Select Account**

Account ID	Batch	Ctrl #
<input type="text"/>	Unknown Batch	<input type="text"/>

Account ID	Batch	Status	SIP
16041234567	easycall		

## Testing the Whole System

○

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

Account ID	Batch	Ctrl #	SIP Status	Advanced search
<input type="text"/>	ANY	<input type="text"/>	ANY	<input type="button" value="Show Accounts"/>

xDRs	Account ID	Idle, days	Currency	Balance	Credit Limit	Type	Product	Batch	Status	SIP
	16041234567		USD	10.00000		Debit	SIP Subscribers	easycall		<input checked="" type="checkbox"/>
	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									
<input type="button" value="Save"/> <input type="button" value="Save &amp; Close"/> <input type="button" value="Close"/> <input type="button" value="xDRs"/> <input type="button" value="Rate Lookup"/> <input type="button" value="E-Payments Log"/>		<input type="button" value="Logout"/> <input type="button" value="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										
<table border="1"> <tr> <th>Account Info</th> <th>Maintenance</th> <th>Subscriber</th> <th>Additional Info</th> <th>Life Cycle</th> <th>User Interface</th> <th>Call Features</th> <th>Subscriptions</th> <th>Notepad</th> </tr> </table>					Account Info	Maintenance	Subscriber	Additional Info	Life Cycle	User Interface	Call Features	Subscriptions	Notepad
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 <input type="button" value="Auto"/>		Non call related charges	0 USD									
E-mail	<input type="text"/>												
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 in log file for: --SELECT--

Include log file

Only text only

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

Logout Close

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 17A91CC0 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 1<- (A? 101/I) 100 trying --0|
22:09:43 1 0-> (A? 101/I) INVITE -----|
22:09:43 1 1<- (A? 101/I) 401 Unauthorized --0|
22:09:43 1 0-> (A? 101/A) ACK -----|
22:09:43 1 0-> (A? 102/I) INVITE -----|
22:09:43 1 1<- (A? 102/I) 100 trying --0|
22:09:43 1 0-> (A? 102/I) INVITE -----|
22:09:43 1 1<- (A? 102/I) 100 Trying ---0|
22:09:44 1 1<- Auth request accepted ---0|
    
```

```

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=873d0427882f87o
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
a=ptime: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=873d0427882f8700;lr>
Authorization: Digest
username="16041234568",realm="193.28.87.106",nonce="3051864d7d0c6578460cc4b0e28ad4
3b448c6a77",uri="sip:
193.28.87.106:5061",algorithm=MD5,response="4e3308a385aeff5159ad03f738f9bd31"
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=z9hG4bK13db39161f94a7d3e3eabcf209661a93;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=z9hG4bK13db39161f94a7d3e3eabcf209661a93;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.



- `less /var/log/porta-billing.log`
- **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;rtpp=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-timepoint     = '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.'										
										Advanced search
xDRs	Account ID	Batch	Idle, days	Currency	Balance	Credit Limit	Type	Product	Batch	Status
	16041234567	ANY		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			
From Date	2007-06-25	YYYY-MM-DD	07:45:00 HH24:MI:SS
To Date	Now	YYYY-MM-DD	HH24:MI:SS
Service Type	All		
Show Unsuccessful Attempts	<input type="checkbox"/>		
<a href="#">Show xDRs</a>			



xDR History									
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			<b>Total</b>	<b>0.14000 USD</b>				
Total Transactions	3								
<a href="#">Show Totals By Service Types</a>									
<b>Voice Calls</b>									
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	
<b>Subtotal</b>								<b>6:24</b>	<b>0.14000</b>

Show xDR

Download .csv

## Check the Call History

### Trace Call

Call Trace
America/Los\_Angeles democroot Help

Close
Logout

H323-conf-id

Trace a call

Destination %  10 min

From 2006-06-11 YYYY-MM-DD 00:00:00 HH24:MI:SS

To 2006-06-12 YYYY-MM-DD 00:00:00 HH24:MI:SS

Trace a call

View	Error Report	CL(ani)	CLD(dnis)	Country	Description	Connect Time	Disconnect Time	Duration, min:sec	Amount	Account	Customer	Vendor	Disconnect Reason
	E	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
	E	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 Vendor	Recovery on timer expiry
	E	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
	E	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
	E	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
	E	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	
	<i>Basic SIP service</i>
	<i>Basic SIP service</i>
Network configuration	
Rating configuration	
	<b>Routing</b>
Account provisioning	
	<i>Basic SIP service</i>
<b>Testing</b>	

## 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 &amp; dial-peers

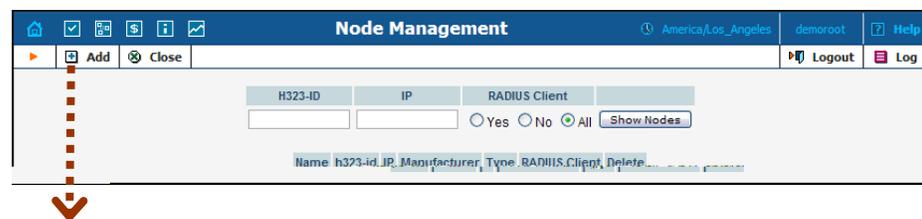
```
call application voice pstn2sip tftp://your-server/pstn2sip.tcl
call application voice pstn2sip authenticate-by dnis
call application voice pstn2sip skip-password yes
call application voice pstn2sip authorize yes
call application voice pstn2sip dial-account-id yes
```

```
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

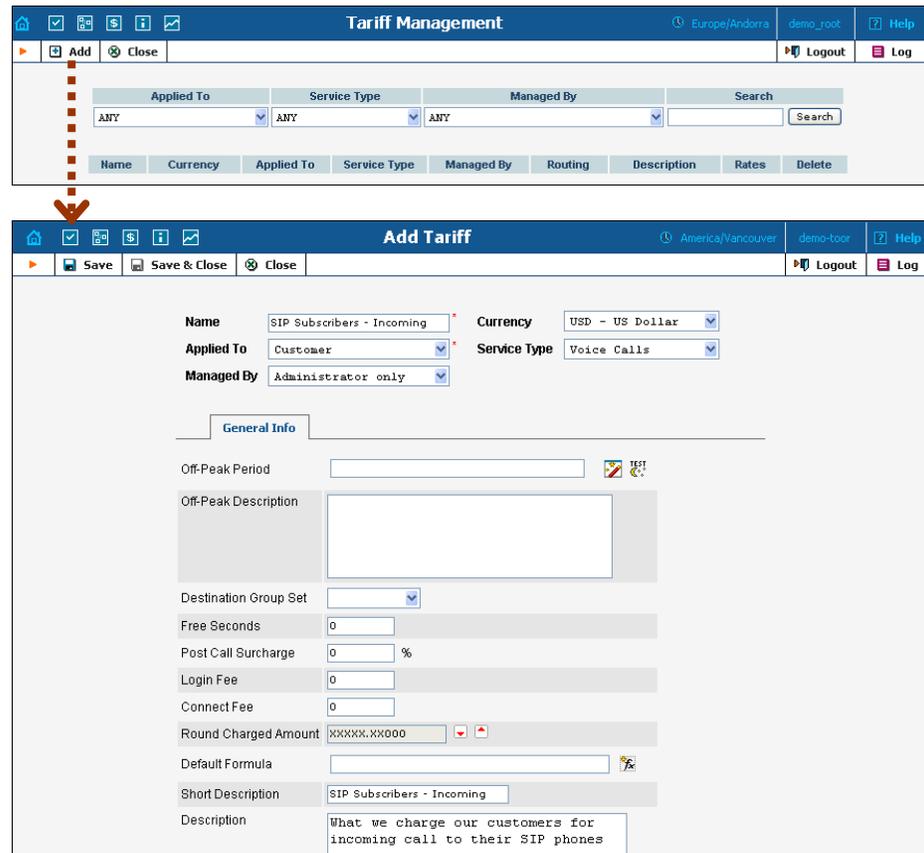




 Save & Close

**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 the 'Tariff Management' interface. At the top, there are navigation icons and a header with 'Tariff Management', 'Europe/Andorra', 'demo\_root', and 'Help'. Below the header, there are 'Add' and 'Close' buttons. A table is displayed with columns: Name, Currency, Applied To, Service Type, Managed By, Routing, Description, Rates, and Delete. The table contains one row with 'ANY' in the first three columns. Below the table, there is a red dashed arrow pointing down to the 'Add Tariff' form.

The 'Add Tariff' form has a header with 'Add Tariff', 'America/Vancouver', 'demo-toor', and 'Help'. Below the header, there are 'Save', 'Save & Close', and 'Close' buttons. The form contains the following fields:

- Name: SIP Subscribers - Incoming
- Currency: USD - US Dollar
- Applied To: Customer
- Service Type: Voice Calls
- Managed By: Administrator only
- General Info tab
- Off-Peak Period: [text input]
- Off-Peak Description: [text area]
- Destination Group Set: [dropdown]
- Free Seconds: 0
- Post Call Surcharge: 0 %
- Login Fee: 0
- Connect Fee: 0
- Round Charged Amount: XXXXXX.XX000
- Default Formula: [text input]
- Short Description: SIP Subscribers - Incoming
- 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

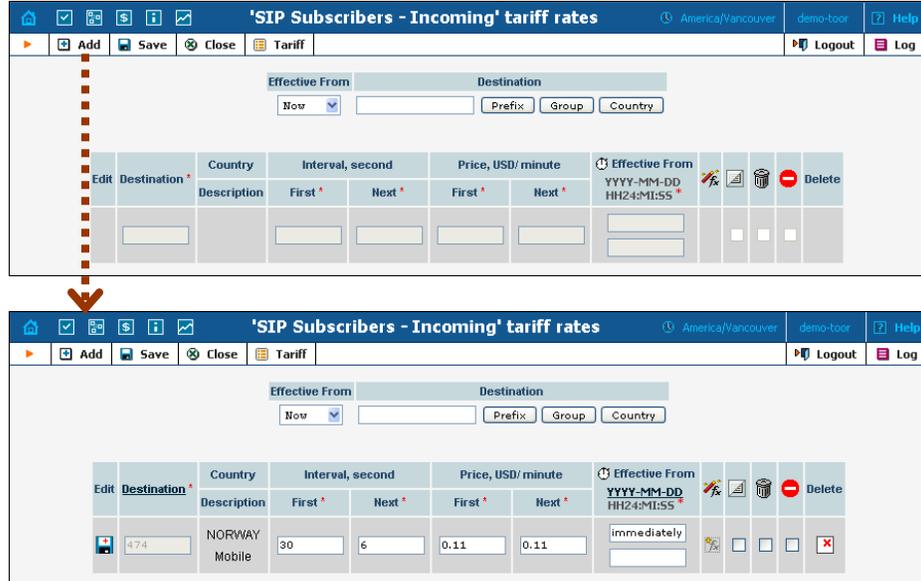
- Off-peak Description
- Destination Group Set
- Free Seconds
- Post-Call Surcharge
- Login Fee
- Connect Fee
- Round Charged Amount
- Default Formula
- Short Description
- Description

 Save

Enter Rates

*System Concepts*

Managing rates online



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

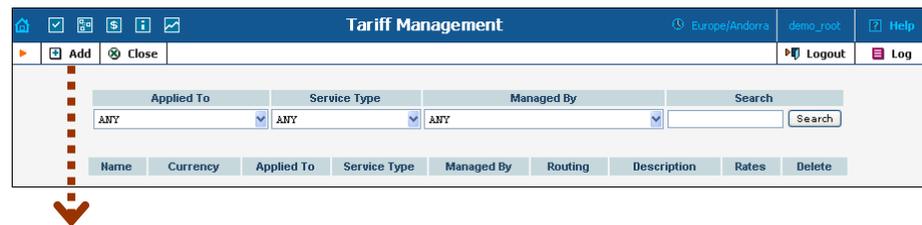
**NOTE:** Off-peak fields appear only if an **off-peak period** has been defined for the tariff.

- 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



Tariffs

Add

Add Tariff

Applied To  
Routing

Vendor

Save

Enter Rates

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

Save



## Modify the Accessibility for the Product

**Product Management** America/Los\_Angeles demoot Help

Managed By: ANY Search

Item	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 demoot Help

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 demoot Help

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 demoot Help

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**

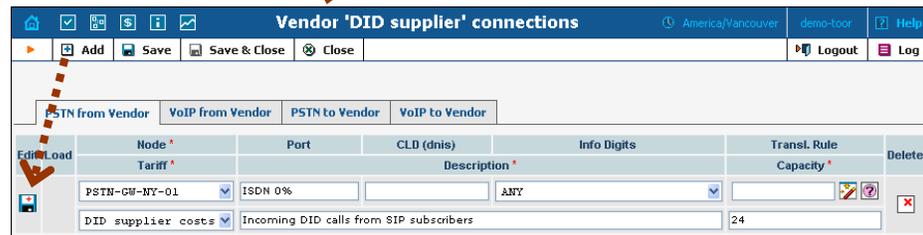
## Define Connections

### Connections

### Vendors



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	
	Teleglobe		USD	0.00000		
	X-Telecom		USD	374.37843	voip@x-telecom.com	



Vendor 'DID supplier' connections						
PSTN from Vendor						
Node *	Port	CLD (dnis)	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

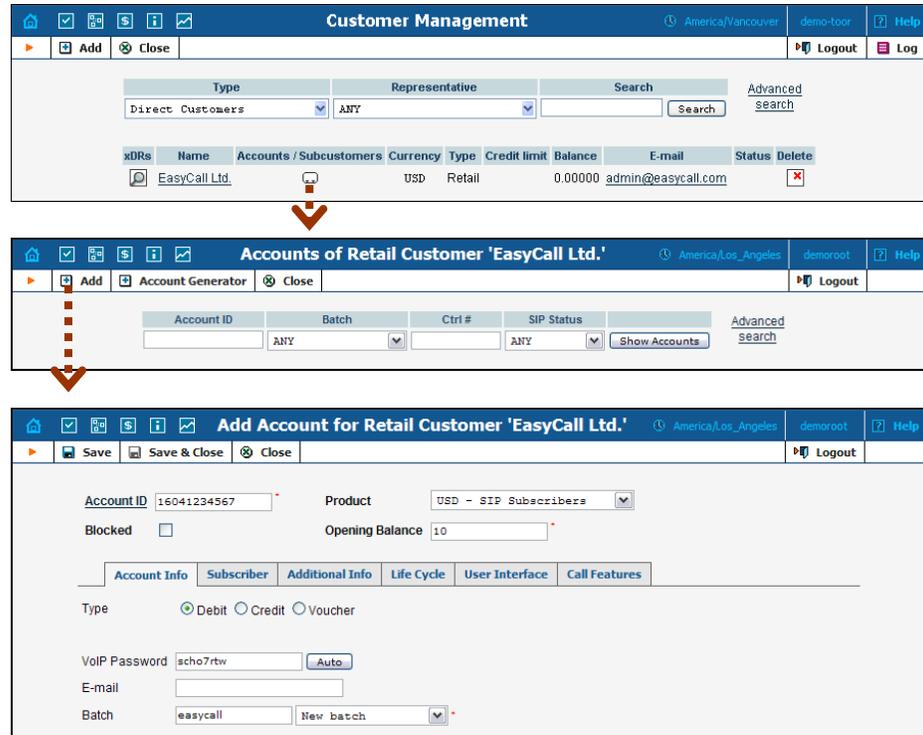
Description

Capacity

 Save

## Create Accounts

### Customers



## 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
- 
- o Timeout, sec

 Save & Close

## Incoming DID calls (from a VoIP vendor)

### Checklist



Operation		Done
<b>Initial configuration</b>		
	<i>Basic SIP service</i>	
	<i>Basic SIP service</i>	
<b>Rating configuration</b>		
		<b>Routing</b>
<b>Account provisioning</b>		

*Basic SIP service*

Testing

Set up a PSTN-to-SIP Service



Create a Tariff to Charge Your Subscribers

**Tariff Management**

Europe/Andorra demo\_root Help

Add Close Logout Log

Applied To	Service Type	Managed By	Search
ANY	ANY	ANY	Search

Name Currency Applied To Service Type Managed By Routing Description Rates Delete

**Add Tariff**

America/Vancouver demo-root Help

Save Save & Close Close Logout Log

Name: SIP Subscribers - Incoming Currency: USD - US Dollar

Applied To: Customer Service Type: Voice Calls

Managed By: Administrator only

**General Info**

Off-Peak Period: [ ]

Off-Peak Description: [ ]

Destination Group Set: [ ]

Free Seconds: 0

Post Call Surcharge: 0 %

Login Fee: 0

Connect Fee: 0

Round Charged Amount: XXXXX.XX000

Default Formula: [ ]

Short Description: SIP Subscribers - Incoming

Description: What we charge our customers for incoming call to their SIP phones

Tariffs



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




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

Uploading a rate list from a file

### PortaBilling Templates Guide

## Create a Tariff for Incoming DID Costs



Tariffs

Add

Add Tariff

Applied To  
Routing

Vendor

Save

Enter Rates

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

Save



## Modify the Accessibility for the Product

Product Management

Managed By: ANY Search

Item	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	
	Smarnet Termination	USD	Administrator only	What we charge Customer Smarnet for calls they terminate to us	

Edit 'SIP Subscribers' Product

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

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	INCOMING	ANY	SIP Subscribers -	
	Voice Calls	DemoSIP			SIP Phone Subscribers	

Edit 'SIP Subscribers' Product

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      Access Code  
**Info Digits**

 **Save**

### Create a DID Supplier Vendor

**Vendors**

 **Add**

America/Los\_Angeles    demoroot    ? Help **Add Vendor**

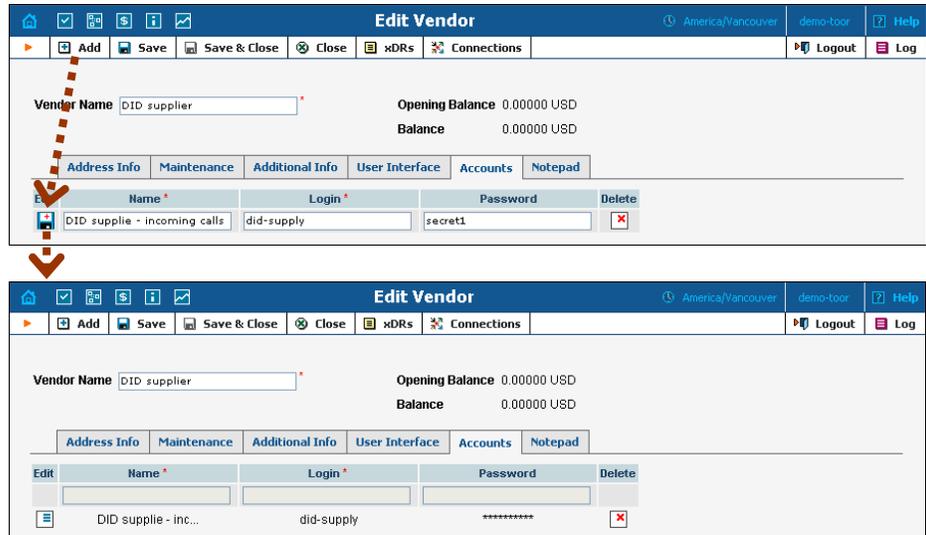
Logout Save Save & Close Close

<p>USD - US Dollar</p> <p>0</p>	<p><b>Vendor Name</b> DID supplier      <b>Currency</b></p> <p style="text-align: right;"><b>Opening Balance</b></p> <p><b>Address Info</b>    <b>Additional Info</b>    <b>User Interface</b></p> <p>Company Name: DID supplier      Contact: <input type="checkbox"/></p> <p>Mr./Ms./...: <input type="text"/>      Phone: <input type="text"/></p> <p>First Name: <input type="text"/>      M.I.: <input type="text"/>      Fax: <input type="text"/></p> <p>Last Name: <input type="text"/>      Alt. Phone: <input type="text"/></p> <p>Address: 11 Broadway, Suite 444      Alt. Contact: <input type="text"/></p> <p>Province/State: NY      E-mail: <input type="text"/></p> <p>Postal Code: 01501      BCC: <input type="text"/></p> <p>City: New York      Send Statistics: <input type="checkbox"/> Fu</p> <p>Country/Region: USA      Description: <input type="text"/></p>
---------------------------------	-----------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------------

**Add Vendor**  
*Basic SIP service*

 **Save**

 **Add**



- Name
- Login

cisco

- Password

 Save



Close

Vendors

Define Connections

Connections

Vendors

**Vendor Management** America/Vancouver demo-tool Help

Add Close Logout Log

Search  Search

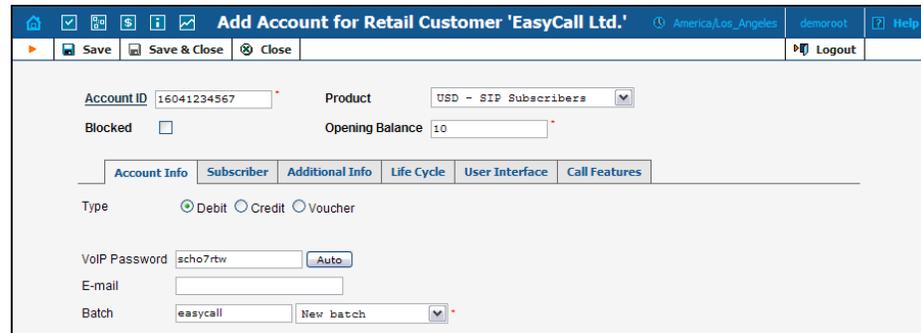
xDRs	Name	Connections	Currency	Balance	E-mail	Delete
	ABC		USD	0.00000		
	<b>DID supplier</b>		USD	<b>0.00000</b>		
	GlobalNet		USD	0.06986	info@globalnet.com	
	MCI Vendor		USD	18.58369		
	SmartNetwork		USD	0.00000		
	SPT Telecom		USD	0.00000	info@spt.cz	
	Telelobe		USD	0.00000		
	X-Telecom		USD	374.37843	voip@xtelecom.com	

**Vendor 'DID supplier' connections** America/Vancouver demo-tool Help

Add Save Save & Close Close Logout Log

PSTN from Vendor VoIP from Vendor PSTN to Vendor VoIP to Vendor

Edit Load	Node *	RTP Proxying	Transl. Rule	Account *	Delete
	Tariff *	Description *		Capacity *	
	DemoSIP	Optimal		DID supplie - i	
	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

<b>Operation</b>	<b>Done</b>
General configuration	

IP phone inventory

Assigning a phone number to the IP phone

IP phone settings

Testing

## Create an IP Phone Profile

Phone Profiles.

IP

Add

- Name
- Managed By

Administrator Only

- Type
- As Copy Of
- Effective From  
immediately

None

Save.

## IP Phone Profile Settings

Save



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

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

**Edit 'Sipura - Standard' IP Phone Profile**

Name: Sipura - Standard  
 Type: Sipura 2000  
 Description: Standard profile for residential SIP  
 Managed By: Administrator only

Effective From: Date: Immediately, Time: HH:MM:SS

System | Provisioning | Regional | Line 1 | Line 2

Line Enable: Yes  
 Proxy: sip.smartcall.com  
 Use Outbound Proxy: No  
 Outbound Proxy:   
 Register: Yes  
 Use DNS SRV: No  
 DNS SRV Auto Prefix: No  
 Register Expires: 3600  
 Display Name: \$firstname \$midinit \$lastname  
 User ID: \$id  
 Auth ID:   
 Use Auth ID: No  
 Preferred Codec: G729a  
 Use Pref Codec Only: No  
 DTMF Tx Method: Auto  
 Dial Plan: (\*xx)[3469]11|0|00|[2-9]xxxxxx1xxx[2-9]xxxxxS0|xxxxxxxxxxxxxx  
 SIP Port: 5060  
 SIP Debug Option: none  
 NAT Mapping Enable: No  
 NAT Keep Alive Enable: No  
 NAT Keep Alive Msg: \$NOTIFY  
 NAT Keep Alive Dest: \$PROXY

Line 1    Line 2

Field	Description
Proxy	
Outbound Proxy	
Use Outbound Proxy	
Register	

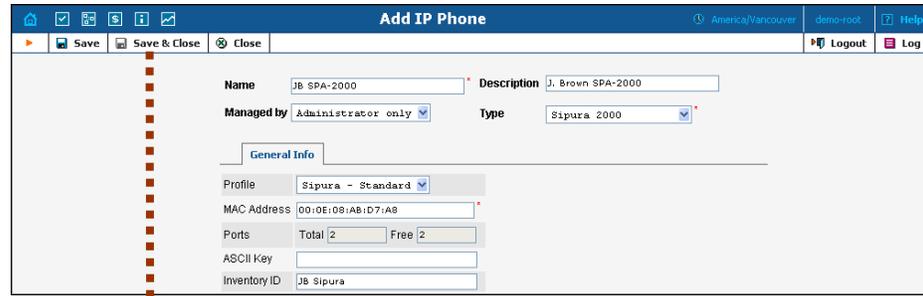
Register Expires	
Display Name	
User ID	
Auth ID	
Use Auth ID	
Preferred Codec	
Use Pref Codec Only	
DTMF Tx Method	
Dial Plan	

### Create an IP Phone Entry



The screenshot shows the 'IP Phone Inventory' management page. At the top, there are navigation icons and a header with 'IP Phone Inventory', 'America/Vancouver', 'demo-root', and 'Help'. Below the header, there are 'Add' and 'Close' buttons. A search section contains dropdown menus for 'Type' (set to 'ANY') and 'Status' (set to 'ANY'), along with a search input field and a 'Search' button. The main area is a table with the following data:

Name	Type	Profile	Managed By	MAC Address	Description	Ports free total	Inventory ID	Delete
<a href="#">My SPA-941</a>	Linksys SPA-941	SPA-941	Administrator only	0:E:8:DB:F8:CE	My office Linksys SPA-941	2 2	KZ Linksys	
<a href="#">SPA 2000_117</a>	Slipura 2000	Slipura 2000	Administrator only	0:50:56:C0:0:8		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	

Phone Inventory.

IP

 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

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  Resync On Reset:  yes

Resync Random Delay:  Resync Periodic:

Resync Error Retry Delay:  Forced Resync Delay:

Resync From SIP:  yes  Resync After Upgrade Attempt:  yes

Resync Trigger 1:

Resync Trigger 2:

Resync Fails On FNF:  yes

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  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.htpd.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	
RxCodec	
TxCodec	
NumTxFrames	
CallFeatures	
PaidFeatures	
CallerIdMethod	
FeatureTimer	
Polarity	
ConnectMode	
AuthMethod	
TimeZone	
NTPIP	

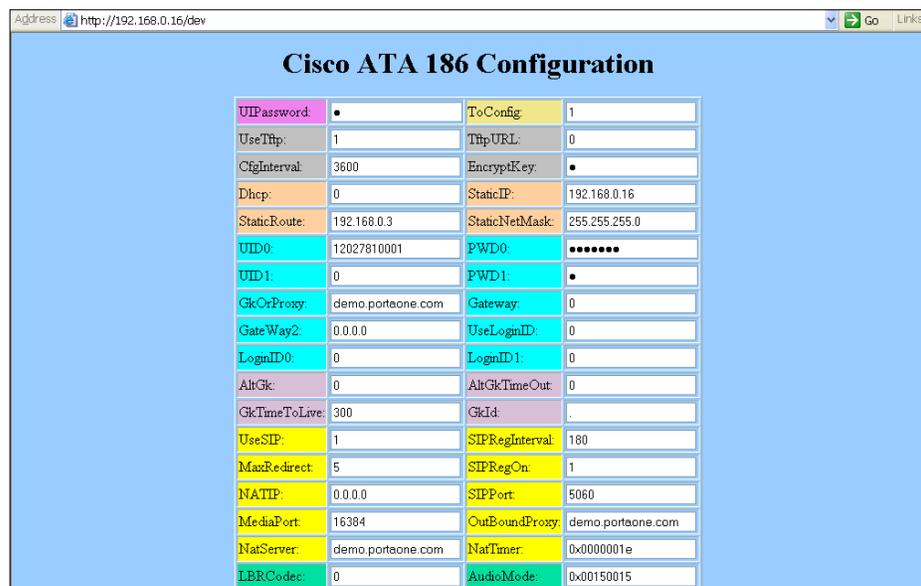
AltNTPIP	
DNS1IP	
DNS2IP	
UDPTOS	
SigTimer	
OpFlags	
VLANSettings	
NPrintf	
TraceFlags	

## 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](http://www.cisco.com/en/US/products/hw/gatecont/ps514/products_configuration_example09186a00800c3a50.shtml)



UIPassword	•	ToConfig	1
UseThp	1	ThpURL	0
CfgInterval	3600	EncrypKKey	•
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

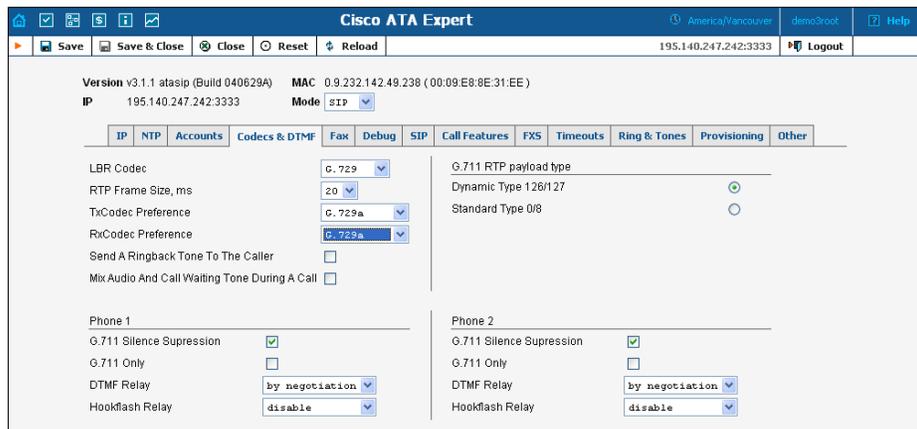
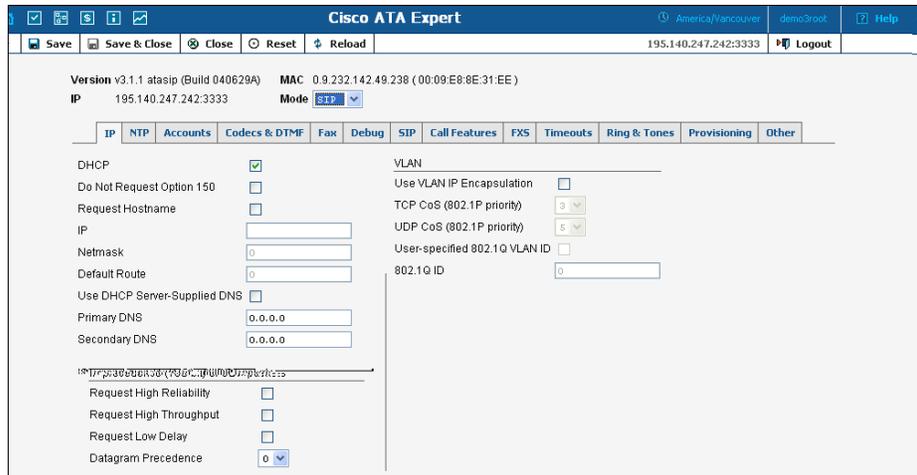
Expert

Cisco

ATA Expert.



**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.



 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:	Enabled	Current IP:	192.168.0.88
Host Name:	SipuraSPA	Domain:	portaone.com
Current Netmask:	255.255.255.0	Current Gateway:	192.168.0.192
Primary DNS:	192.168.0.192		
Secondary DNS:	207.102.99.66 207.102.99.82		

**Product Information**

Product Name:	SPA-2000	Serial Number:	88012BA66086
Software Version:	2.0.10(e)	Hardware Version:	2.0.1(0905)
MAC Address:	000E08AB4638	Client Certificate:	Installed

**System Status**

Current Time:	1/8/2003 14:17:56	Elapsed Time:	4 days and 02:23:13
Broadcast Pkts Sent:	0	Broadcast Bytes Sent:	0
Broadcast Pkts Recv:	560688	Broadcast Bytes Recv:	34980083
Broadcast Pkts Dropped:	0	Broadcast Bytes Dropped:	0
RTP Packets Sent:	3074	RTP Bytes Sent:	120568
RTP Packets Recv:	2341	RTP Bytes Recv:	54292
SIP Messages Sent:	1724	SIP Bytes Sent:	1167889
SIP Messages Recv:	362	SIP Bytes Recv:	166405
External IP:			

**Line 1 Status**

Hook State:	On	Registration State:	Registered
Last Registration At:	1/8/2003 14:07:33	Next Registration In:	2947 s
Message Waiting:	No	Call Back Active:	No
Last Called Number:	16044680035	Last Caller Number:	
Mapped SIP Port:			
Call 1 State:	Idle	Call 2 State:	Idle
Call 1 Tone:	None	Call 2 Tone:	None
Call 1 Encoder:		Call 2 Encoder:	
Call 1 Decoder:		Call 2 Decoder:	
Call 1 FAX:		Call 2 FAX:	
Call 1 Type:		Call 2 Type:	
Call 1 Remote Hold:		Call 2 Remote Hold:	
Call 1 Callback:		Call 2 Callback:	
Call 1 Peer Name:		Call 2 Peer Name:	
Call 1 Peer Phone:		Call 2 Peer Phone:	

<b>Network Settings</b>	
SIP TOS/DiffServ Value:	0x68
RTP TOS/DiffServ Value:	0xb8
Network Jitter Level:	high
<b>SIP Settings</b>	
SIP Port:	5060
EXT SIP Port:	
SIP Debug Option:	none
SIP 100REL Enable:	no
Auth Resync-Reboot:	yes
<b>Call Feature Settings</b>	
Blind Attn-Xfer Enable:	no
Xfer When Hangup Conf:	yes
MOH Server:	
<b>Proxy and Registration</b>	
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
<b>Subscriber Information</b>	
Display Name:	
Password:	*****
Auth ID:	
Mini Certificate:	
S RTP Private Key:	
User ID:	1206001236
Use Auth ID:	no
<b>Supplementary Service Subscription</b>	
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:

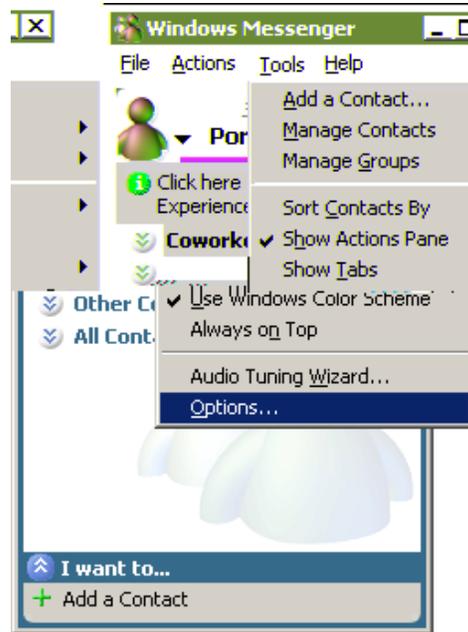
Password:

Save service information permanently

OK Cancel Help

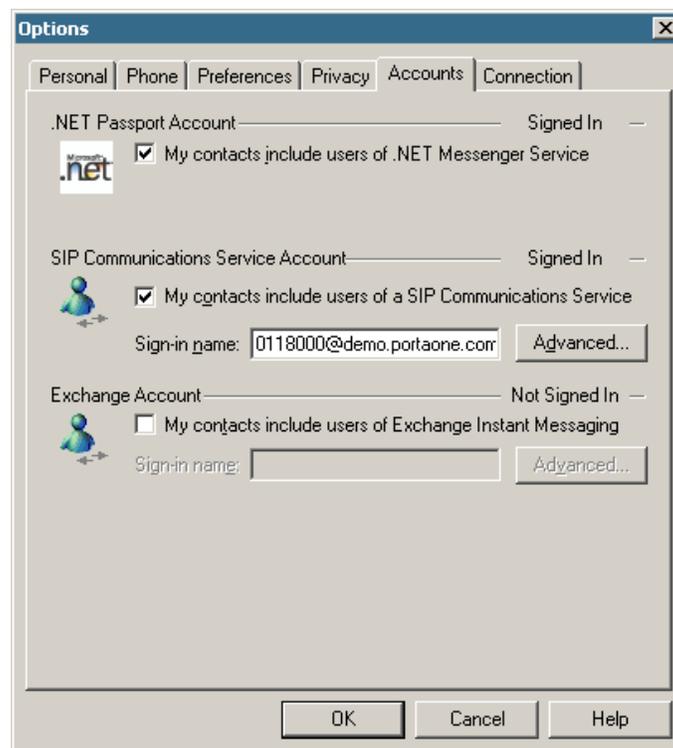


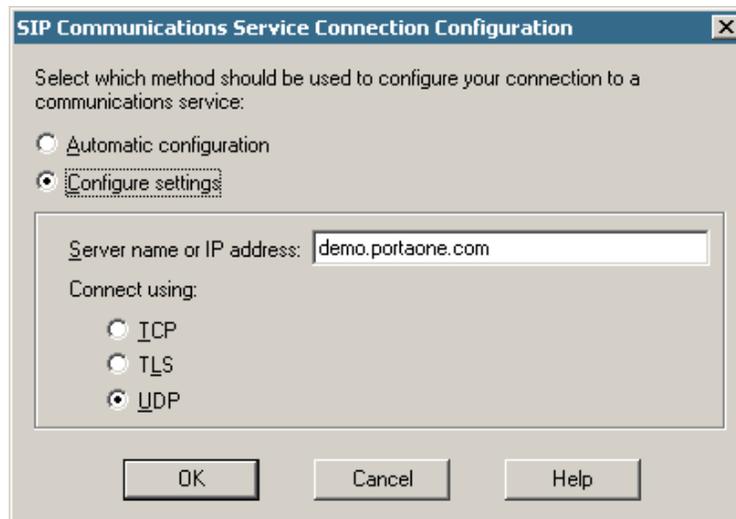
## APPENDIX E. Configuring Windows Messenger for Use as a SIP User Agent



*username@address*

*username  
address*





*username@address      username  
                                         address*



