PORTA ONE

Porta Switch™

PortaSwitch Handbook: SIP Services

Part I

Maintenance Release 15

www.portaone.com

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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

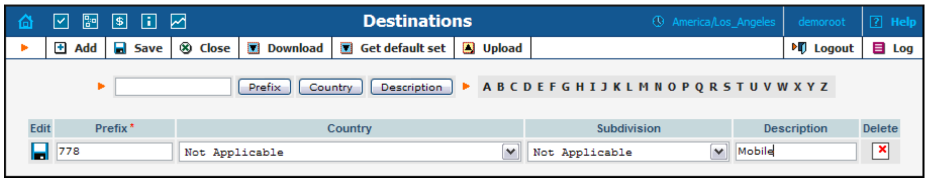
Operation	Done
General configuration	
Network configuration	
Rating configuration	
Routing	
Account provisioning	
Testing	

Creating destinations “one-by-one”:

Destinations

Add

Description

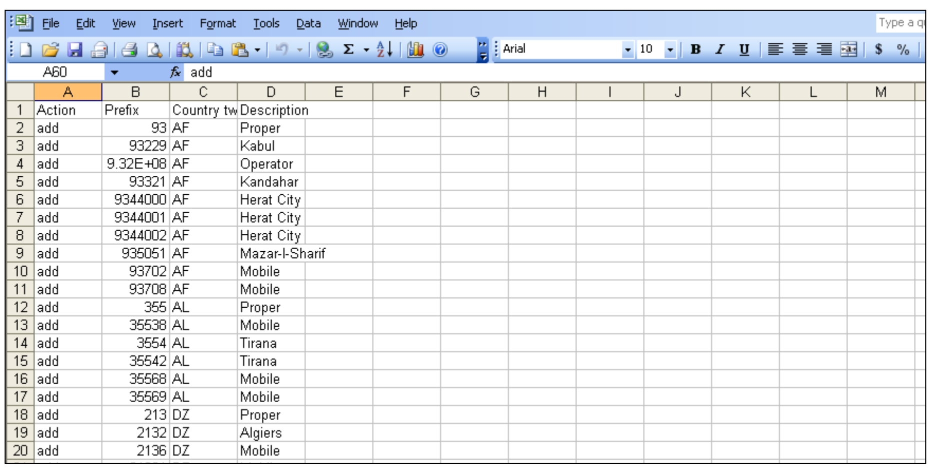


Save

Uploading a set of destinations from a file:

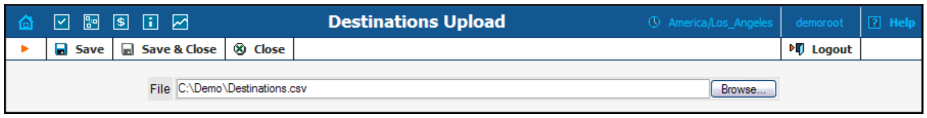
Destinations

Default set



	A	B	C	D
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



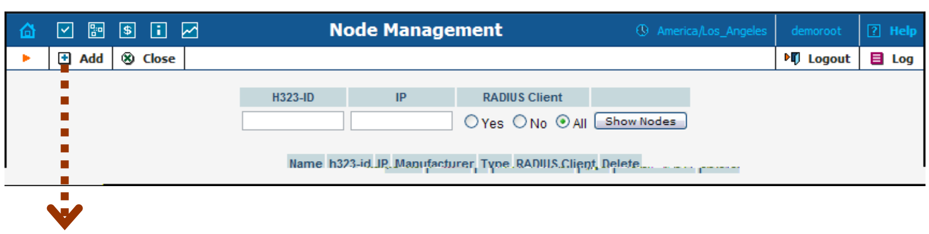
Browse…
Save&Close.

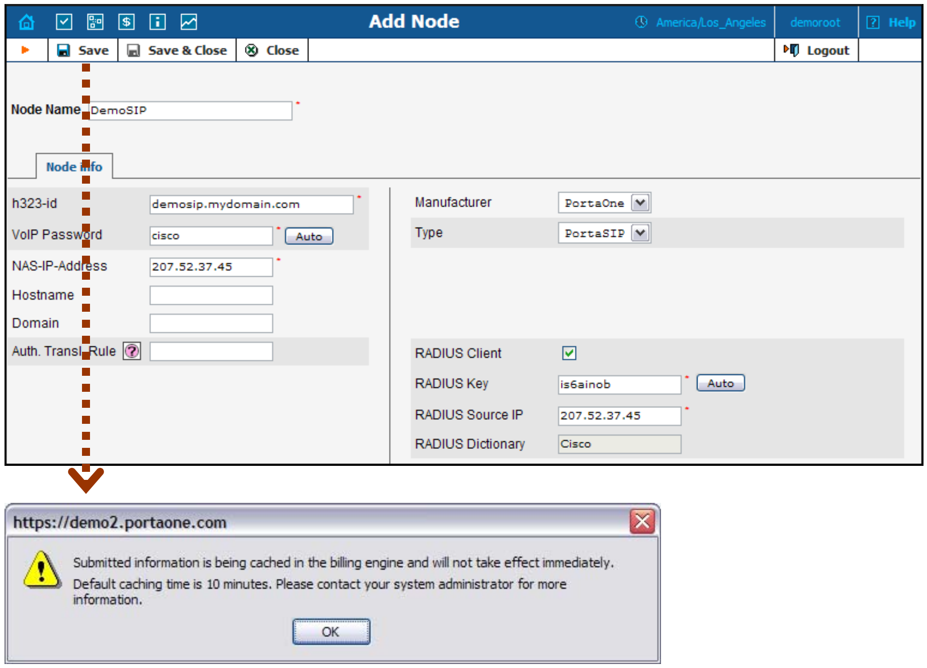
Destinations for SIP phones

1202781

777
N/A **SIP phones**
77700001, 7770002, … 7770999, …

Create Nodes





Nodes.

Add

- Node name
- H323-ID
  `hostname.domainname`
- VoIP Password

  cisco
- NAS-IP-Address
- Auth. Translation rule

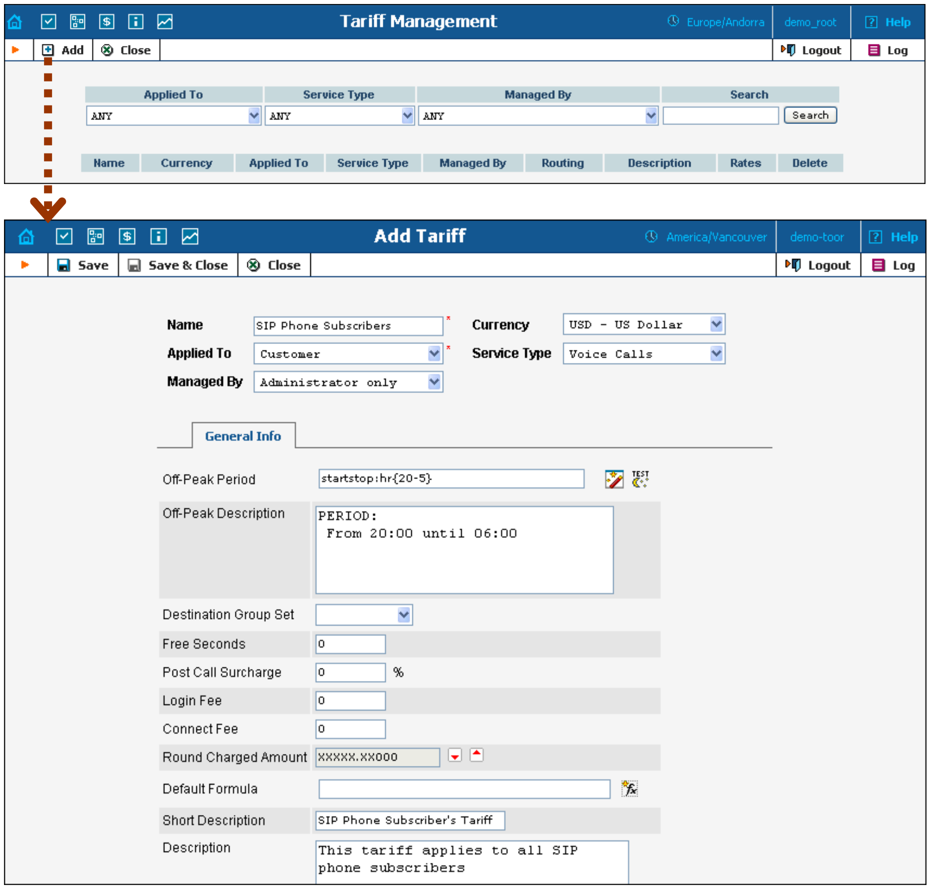
- Manufacturer PortaOne
- Type PortaSIP
- Radius Client

- 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
  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

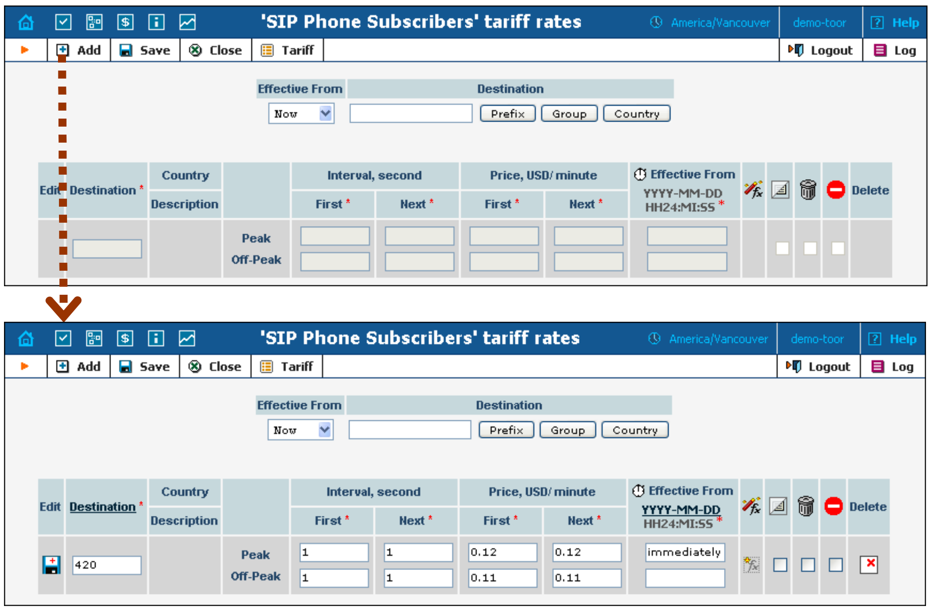
Vendor Applied To
Routing

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

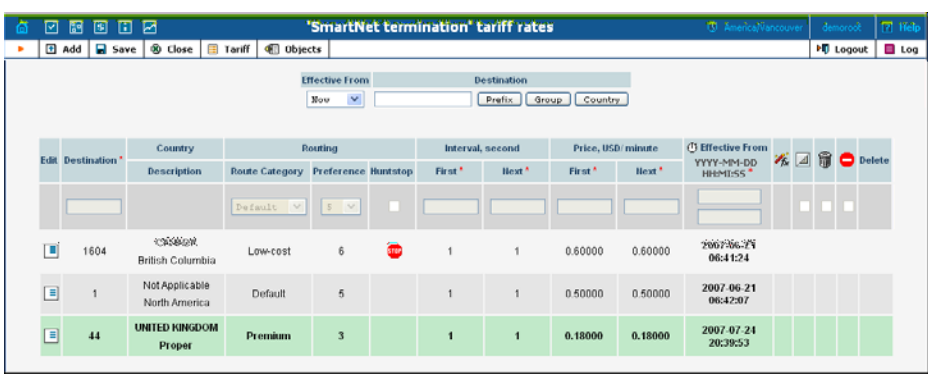
NOTE: Off-peak fields appear only if an **off-peak period** has been defined for the tariff.

- 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



- Route category

Default

- o **Preference**

- o **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

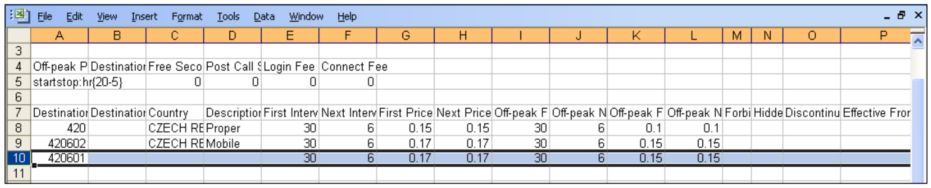
File download

	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P
1	Name	Currency	Descriptior	Short Desc	Off-Peak Description											
2	SIP Phone	USD	This tariff a	SIP phone	PERIOD: From 20:00 until 06:00											
3																
4	Off-peak P	Destinatior	Free Seco	Post Call S	Login Fee	Connect Fee										
5	startstop:hr	{20-5}	0	0	0	0										
6																
7	Destinatior	Destinatior	Country	Descriptior	First Interv	Next Interv	First Price	Next Price	Off-peak F	Off-peak N	Off-peak F	Off-peak N	Forbi	Hidde	Discontinu	Effective Fror
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	Destinatior	Free Seco	Post Call $	Login Fee	Connect Fee										
5	startstop:hr{20-5}		0	0	0	0										
6																
7	Destinatior	Destinatior	Country	Descriptior	First Interv	Next Interv	First Price	Next Price	Off-peak F	Off-peak N	Off-peak F	Off-peak N	Forbi	Hidde	Discontinu	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**

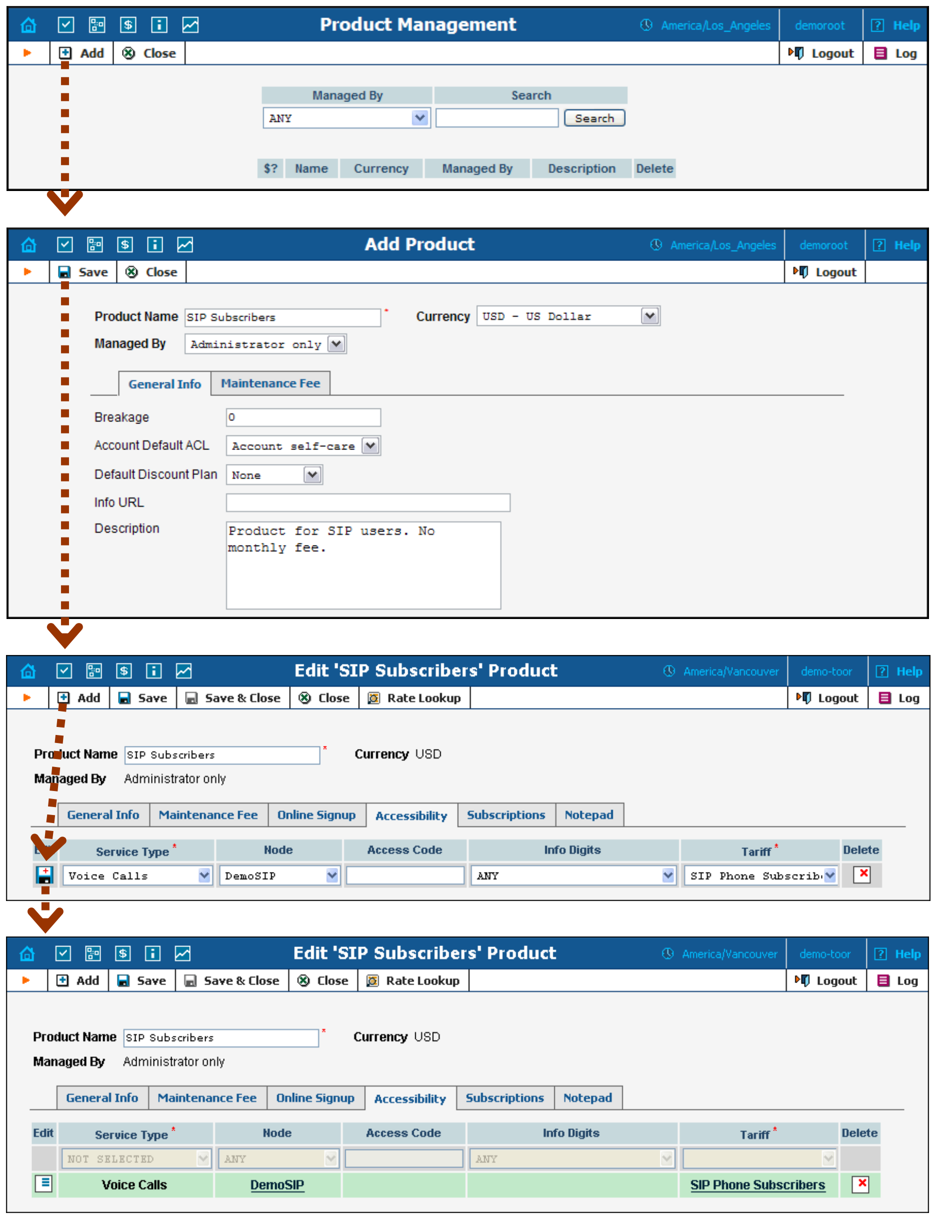
Create All Required Tariffs

Create Tariff *Enter Rates*

-
-
-

Managed by NNN *NNN*

Create Product



Products

Add

o 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**

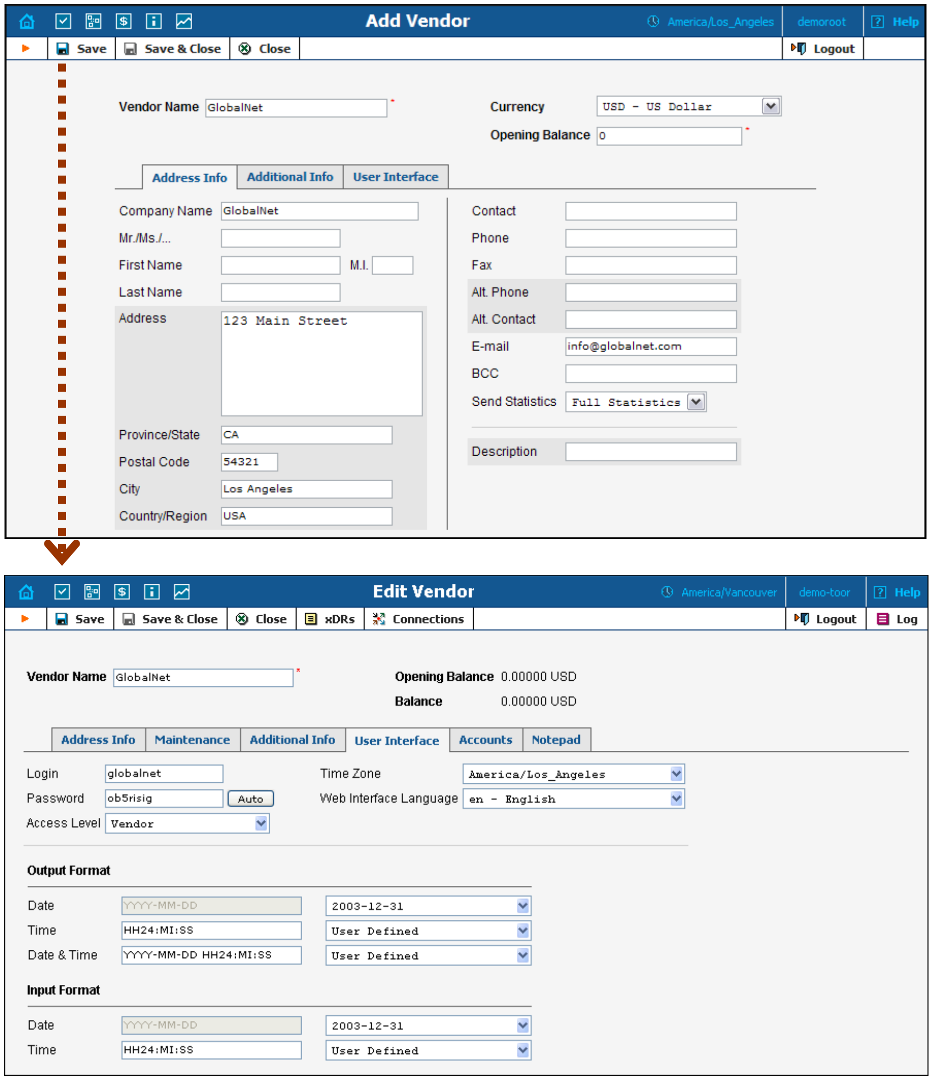
Access code **Info Digits**

Save

Create Vendors

Vendors

Add



Add Vendor

Main form (top)

- Vendor name

- Currency
- Opening Balance

Additional info:

- Billing Period

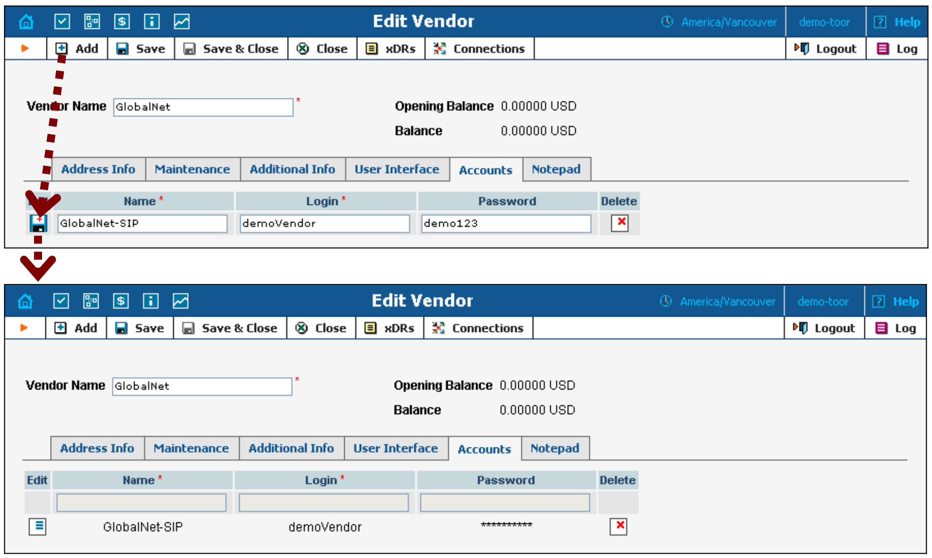
User Interface:

- Time Zone

Save

Vendor

account

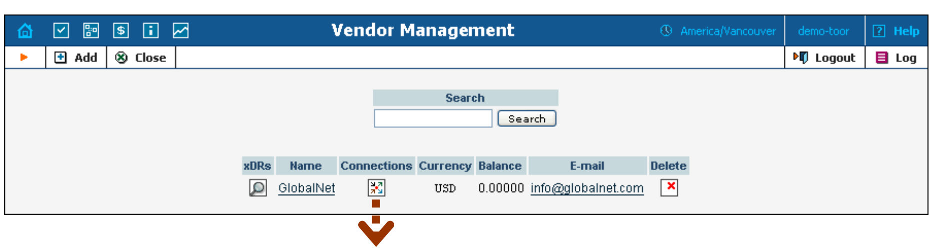


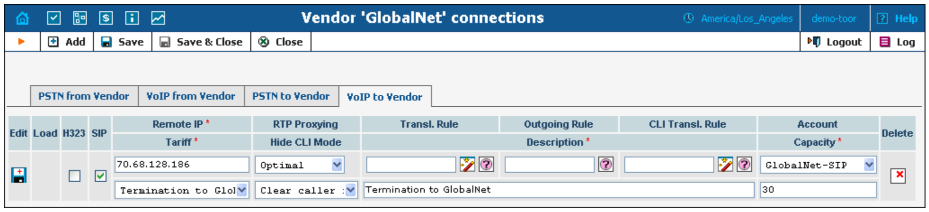
Close Vendors

Define Connections

Vendors

Connections





PSTN to Vendor **VoIP to Vendor**

Add

Description

Capacity

Save

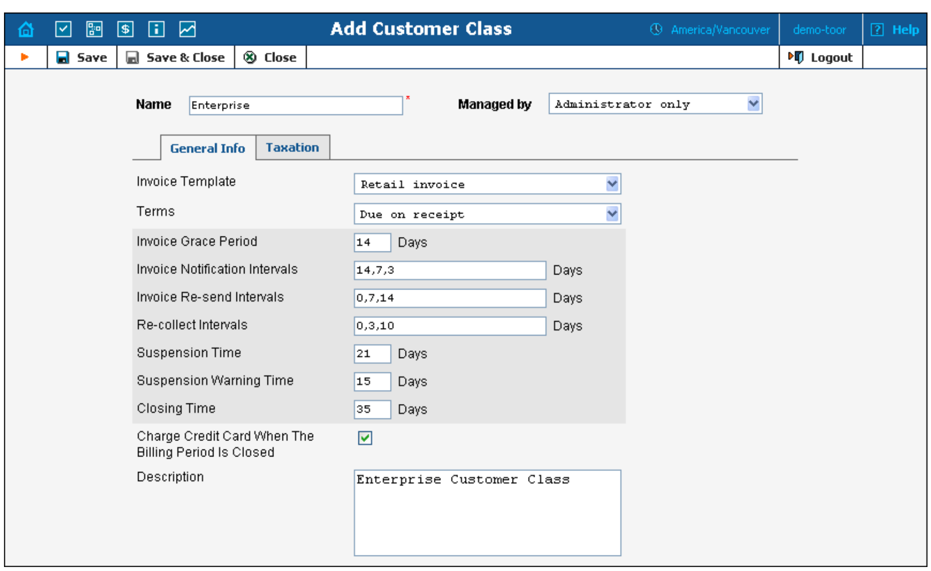
Close **Vendor Management**

Create a Customer Class

Customer

Classes

Add



- Name
- Invoice Template

Do not create invoice

- Terms
- PortaBilling Web Reference Guide

- Description

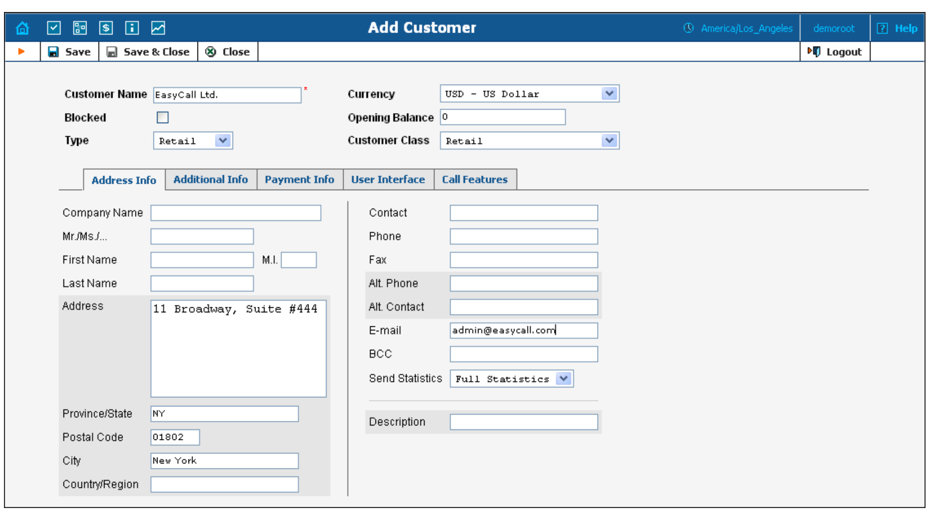
Save&Close

Create a Customer

Customers

Add

New Customer



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

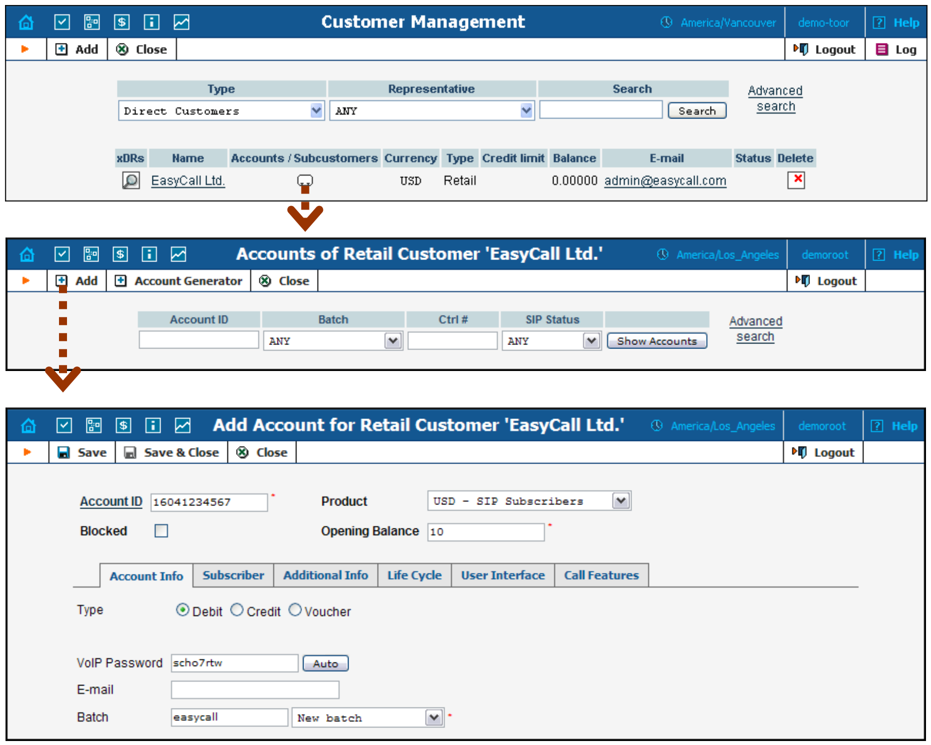
- First login greeting

- Dialing rules

Save&Close

Create Accounts

Customers



Accounts

Add

- Account ID
- Product
- Blocked
- Opening Balance

Account Info tab

- Type
- Credit limit
- VoIP Password

- Email

- Batch

Additional Info tab

- IP Phone

- IP Phone Port

Life Cycle tab

- Activation date
- Expiration date

- Lifetime

User Interface tab

- Login

- Password
- Time Zone

- Web Interface Language

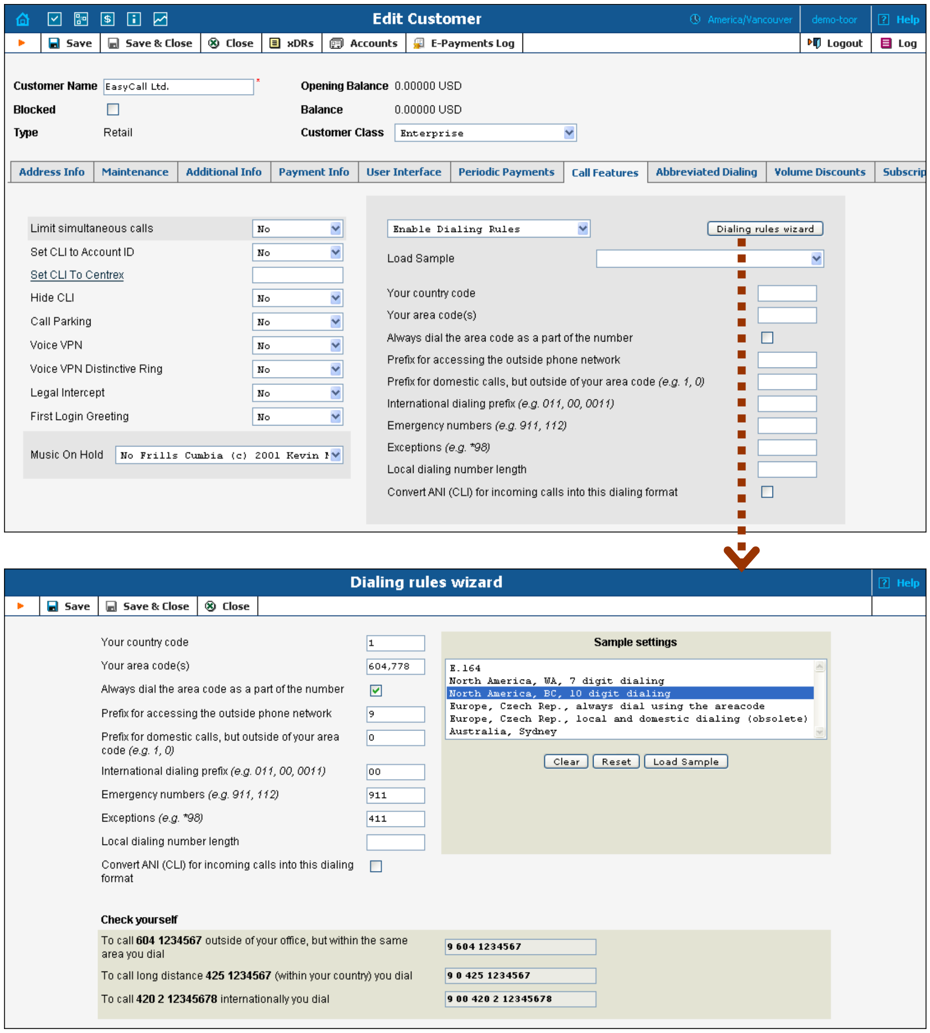
Call Features tab

- Preferred IVR Language

- Associated Number
- UM Enabled
- Forward Mode
  -
  -
  -
  - Follow-me Forward
  -

Save&Close

Set up Dialing Rules for the Customer (optional)



Set up Abbreviated Dialing for the Customer (optional)

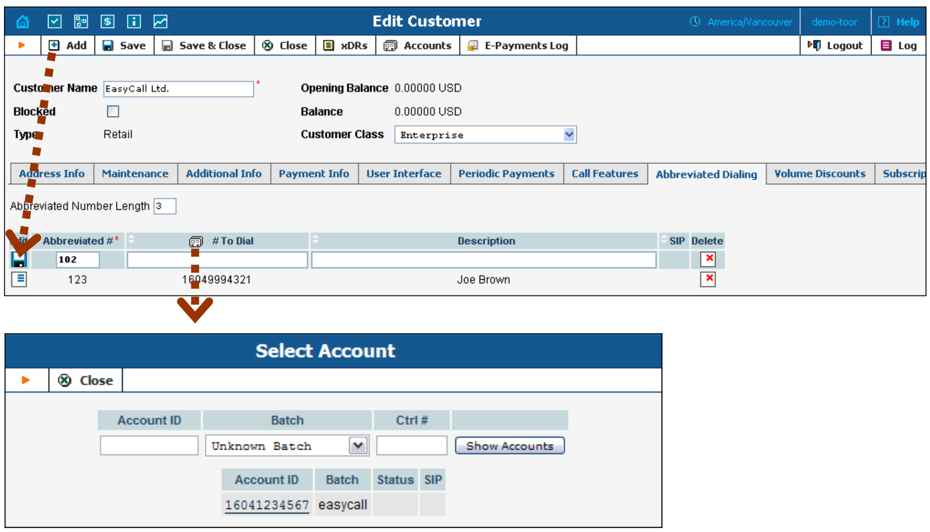
Abbreviated Number Length

Save
Add

 # To Dial

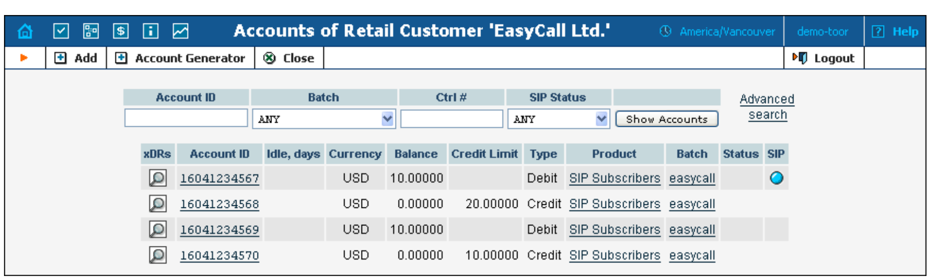
NOTE: If you enter an off-net PSTN number in **# To Dial**; it must be in the E.164 format, i.e. you cannot enter the number in the customer's dialing format.

Save



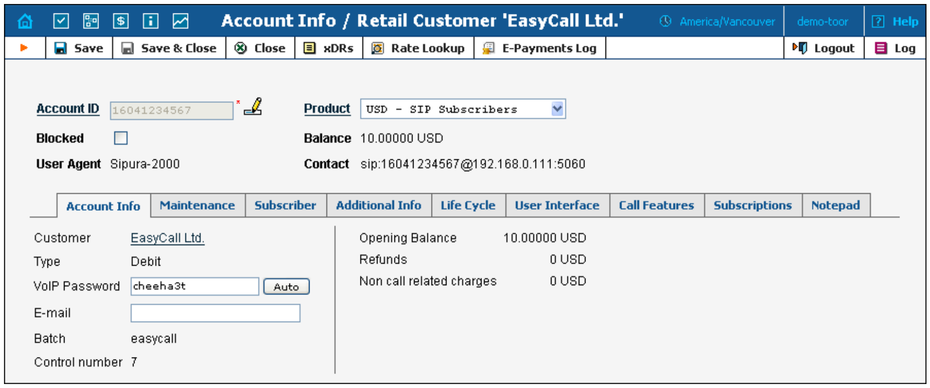
Testing the Whole System

o



o

User Agent **Contact**



Check the Log Files

`/var/log/sipenv-<sipserverIP>/log/sip.log`

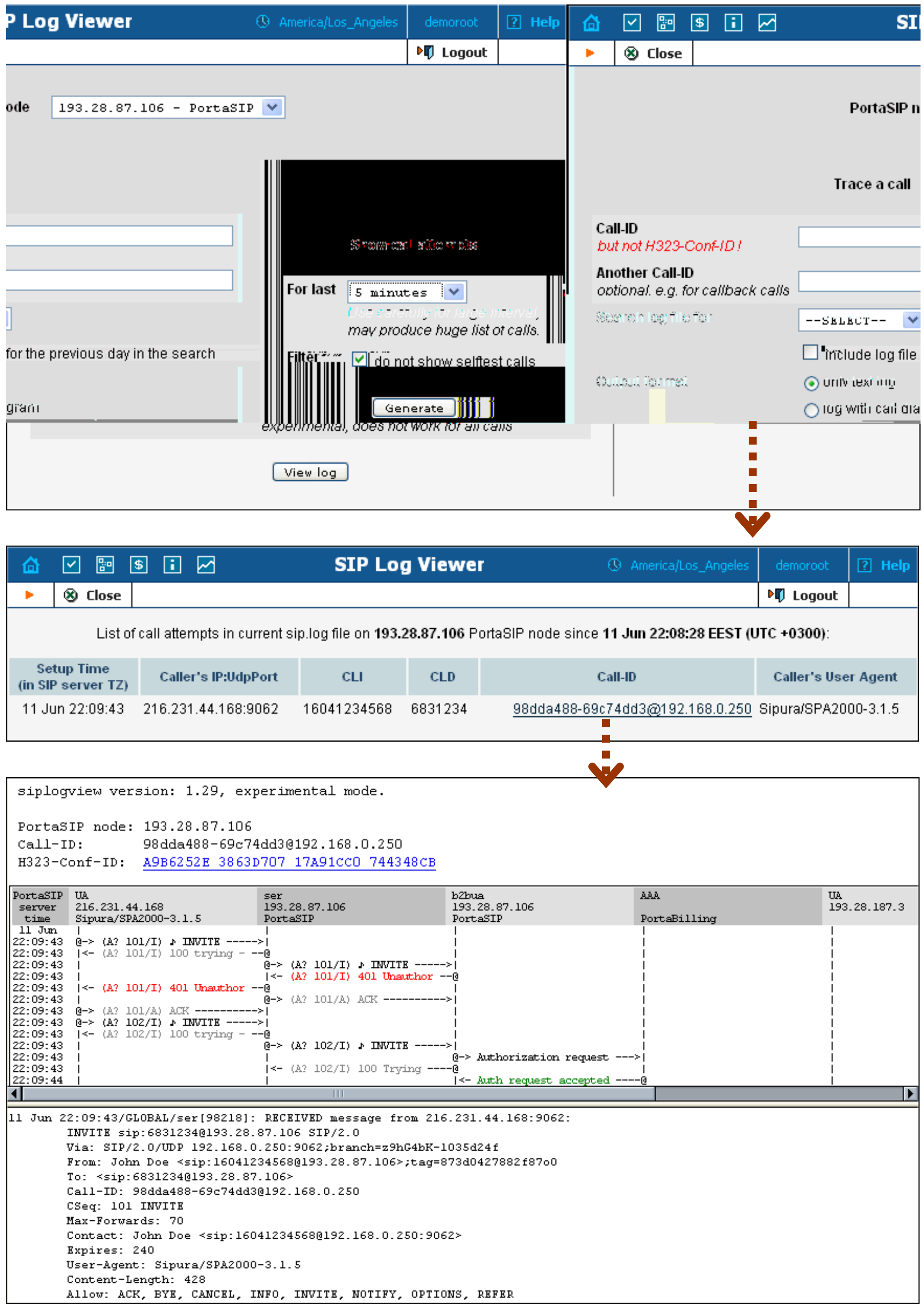
`less` `more`

Using the SIP log viewer

SIP Log Viewer

PortaSIP node

Generate


P Log Viewer
America/Los_Angeles
demoroot
Help
Logout
Close
193.28.87.106 - PortaSIP
For last
5 minutes
may produce huge list of calls.
do not show selftest calls
Generate
for the previous day in the search
experimental, does not work for all calls
View log
PortaSIP n
Trace a call
Call-ID
but not H323-Conf-ID !
Another Call-ID
optional. e.g. for callback calls
--SELECT--
include log file
SIP Log Viewer
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
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

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;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  = '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=873d0427882f87o0;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'
```

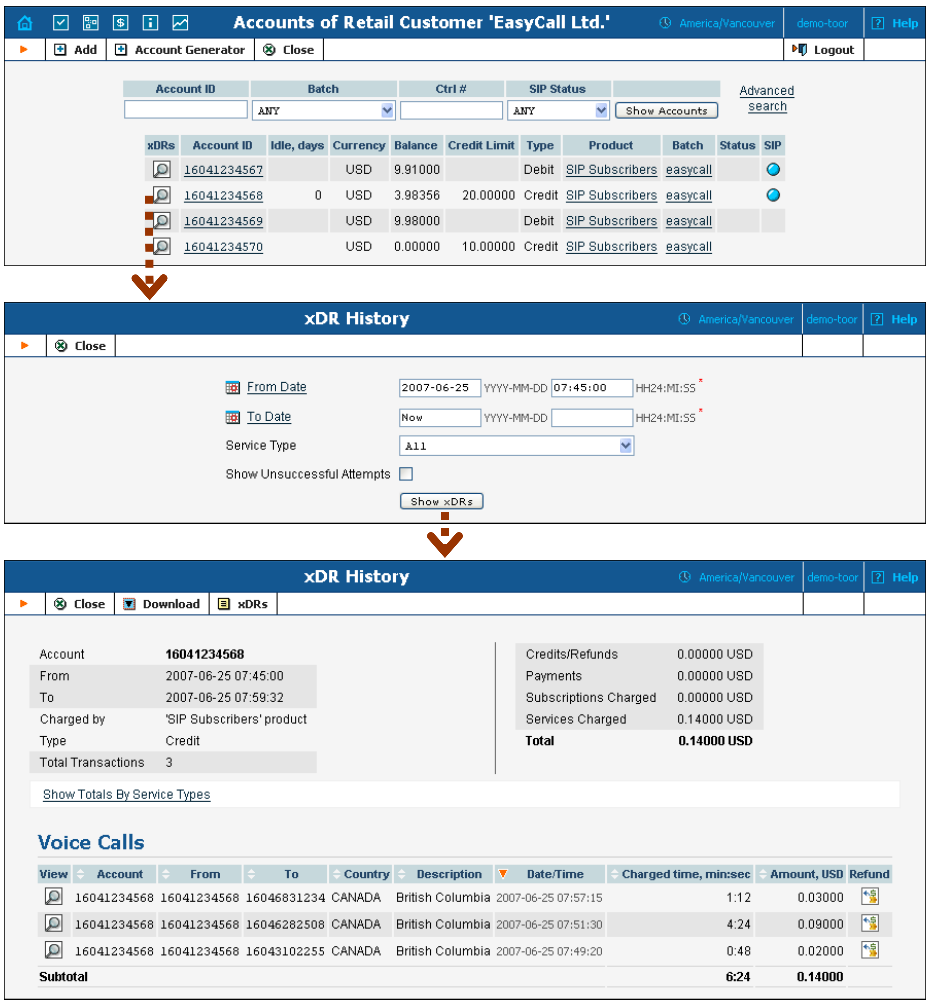
CDRs are inserted and balances are modified.

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

Verify Call History for the Account

Accounts

Home

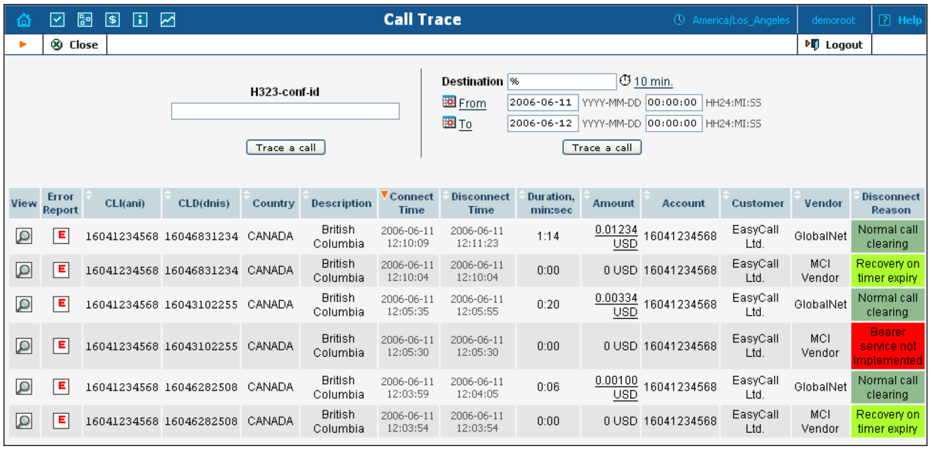


Show xDR

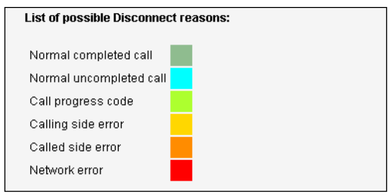
Download .csv

Check the Call History

Trace Call



View	Error Report	CLI(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



- Trace a Call
-
  - h323-conf-id
  - Destination

    380%
  - From, To Date –
- Trace a Call.

The advantage of this method:

xDR

2. Setting up PSTN-to-SIP Services

PortaBilling100 Web Reference Guide

Incoming DID calls (from PSTN)

Checklist

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

Cisco gateway configuration guidelines

Obtain the PSTN2SIP script

```
        /tftpboot
```

Basic router configuration

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

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

NTP

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

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

AAA

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

VoIP interface

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

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

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

Outgoing SIP server

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

Enable gateway functionality

```
gateway
```

Enable gateway accounting

```
gw-accounting h323 vsa

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

NOTE: VSA does not work for all platforms.

Radius

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

Cisco notes:

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

Remember:

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

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

voice-card

controller

voice-port

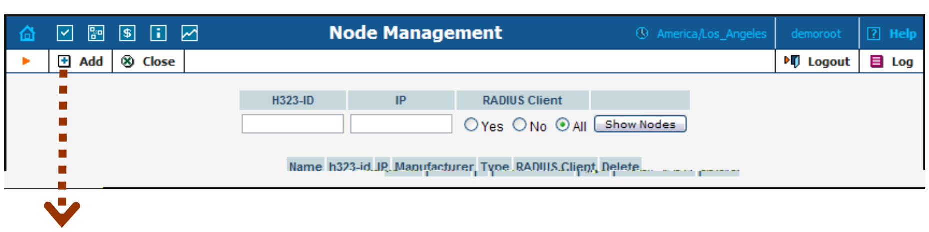
call application voice & dial-peers

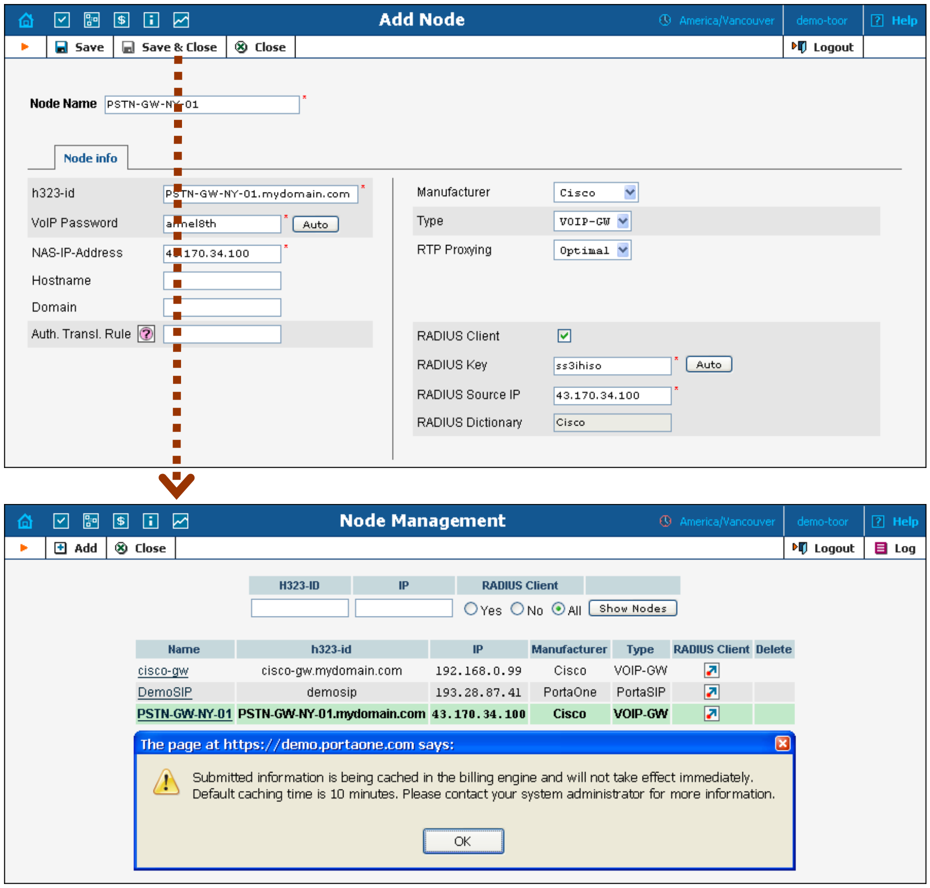
```
call application voice pstn2sip tftp://your-server/pstn2sip.tcl
call application voice pstn2sip authenticate-by dnis
call application voice pstn2sip skip-password yes
call application voice pstn2sip authorize yes
call application voice pstn2sip dial-account-id yes

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

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

Set up a PSTN-to-SIP Service

Create a PSTN Gateway Node





Nodes.

Add

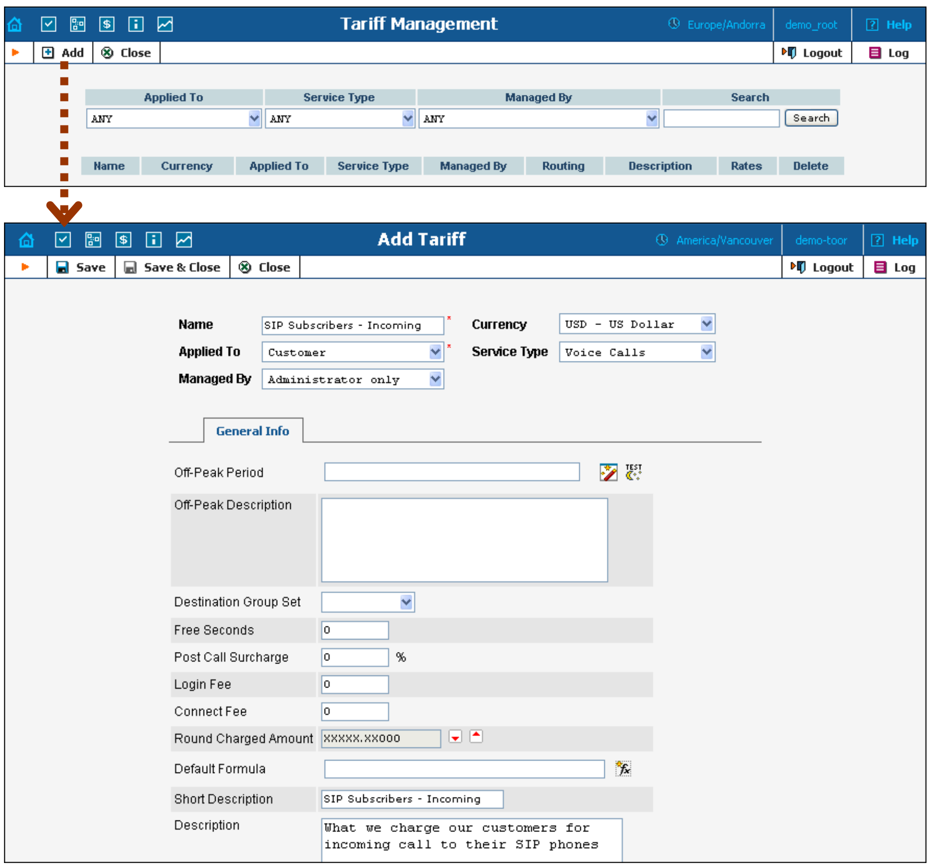
- **Node name**
- **h323-ID** `hostname.domainname`
- **VoIP Password** **`cisco`**
- **NAS-IP-Address**
- **Auth. Translation rule**

- **Manufacturer** **Cisco**
- **Type** **VOIP-GW**
- **Radius Client**
- **Radius Key**
  key
- **Radius Source IP** **Node ID, NAS IP address and 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 a Tariff to Charge Your Subscribers



Tariffs

Add

Add Tariff

- Name

- Currency

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

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

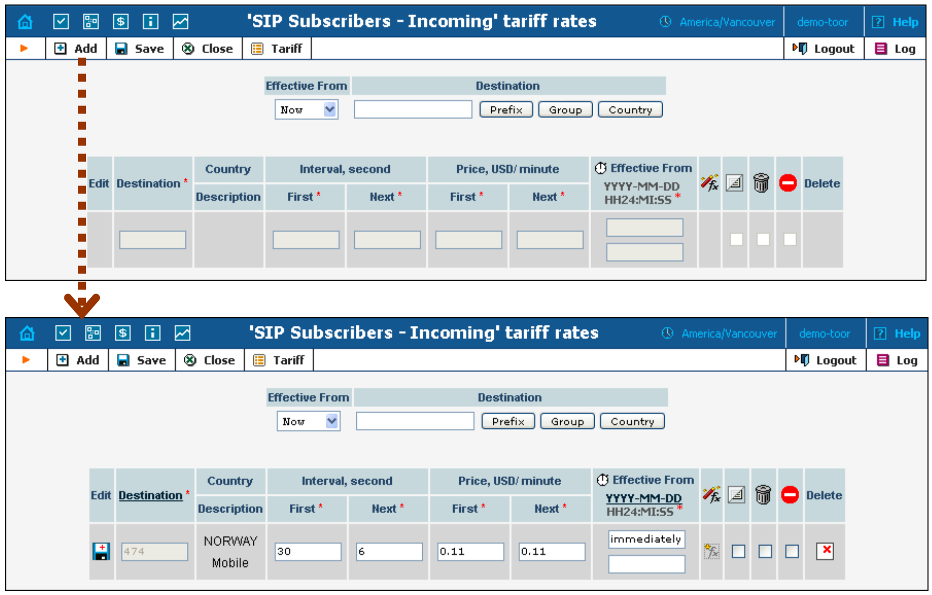
Help

- 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

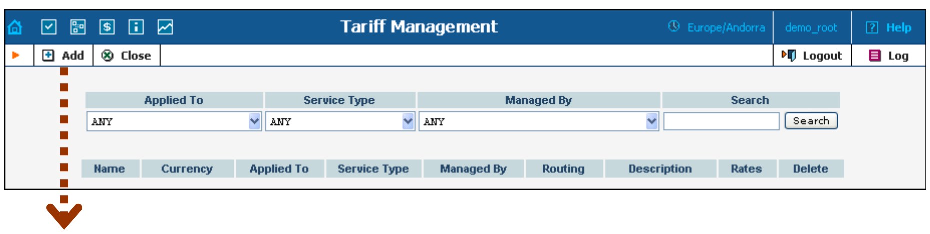
NOTE: Off-peak fields appear only if an **off-peak period** has been defined for the tariff.

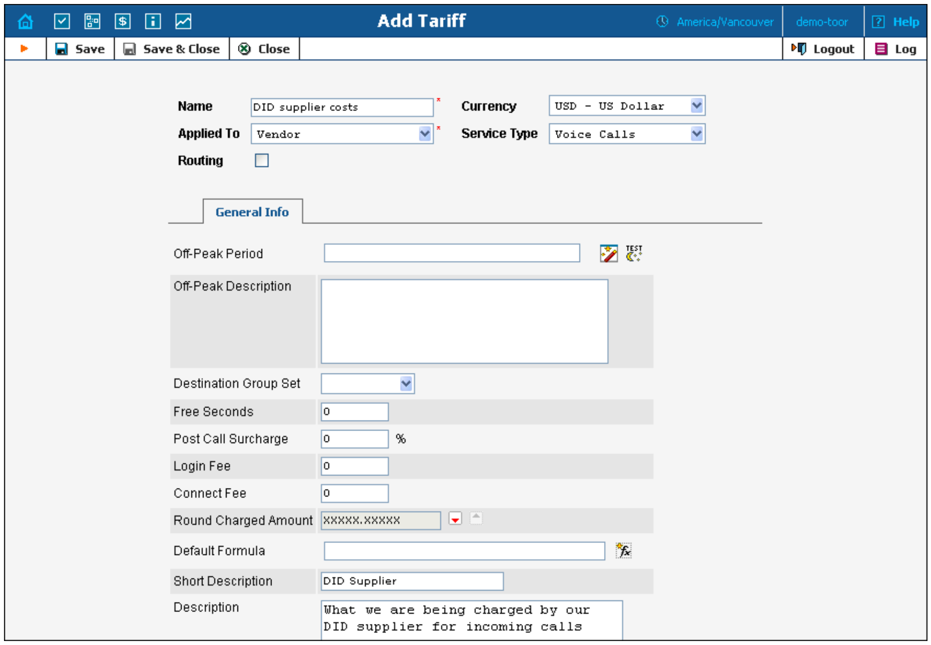
- Rating Formula Wizard
- Effective from

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

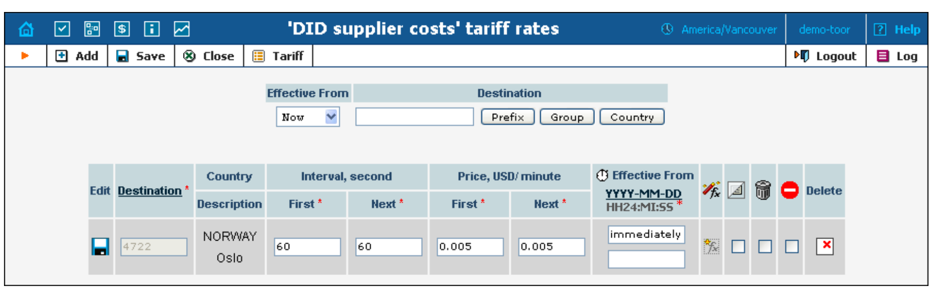
Applied To

Vendor

Routing

Save

Enter Rates



Rates

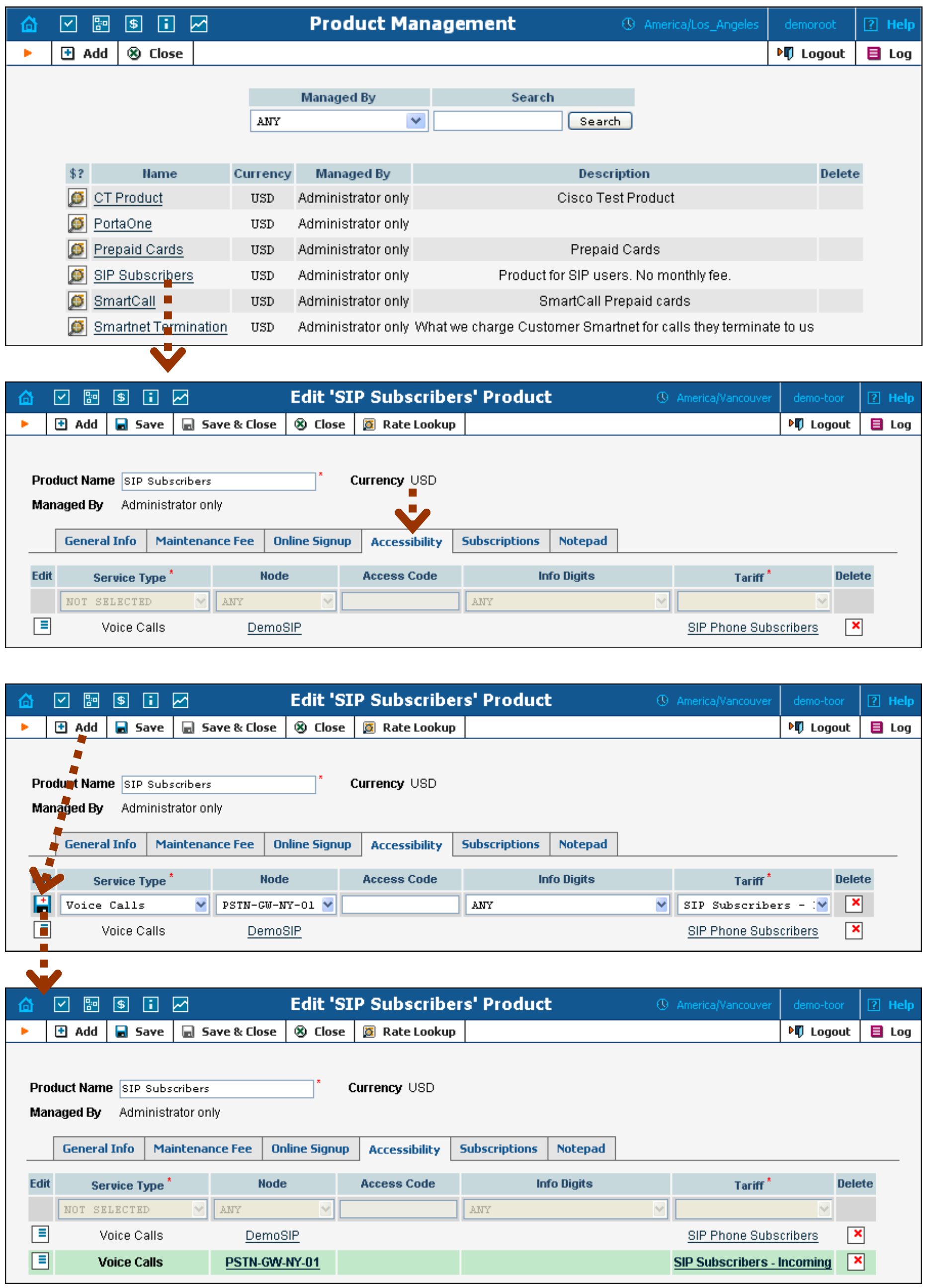
Rates

Edit Rates

Add

Save

Modify the Accessibility for the Product



Products

Add

Voice Calls Service Type

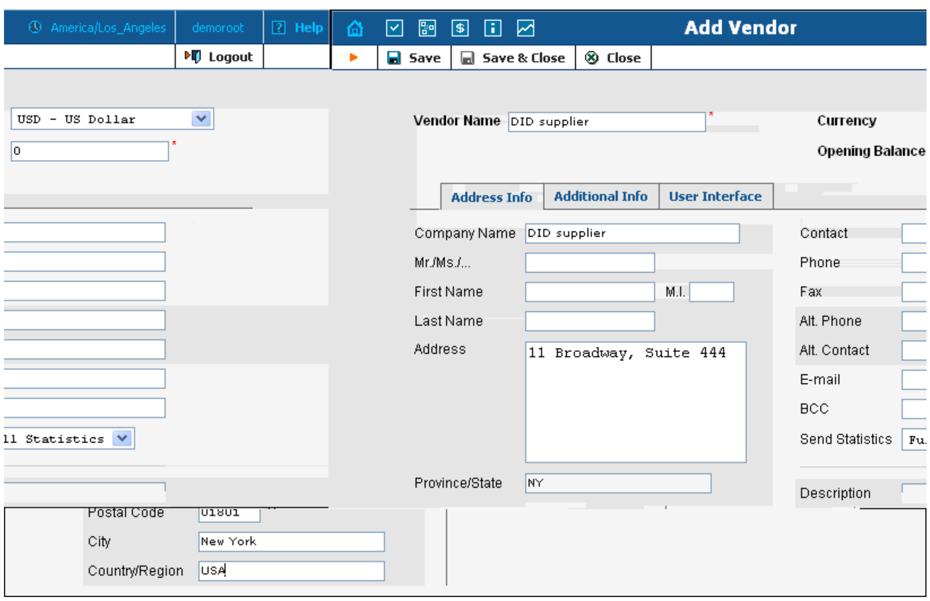
Access Code Info Digits

Save

Create a DID Supplier Vendor

Vendors

Add



Add Vendor

Basic SIP service

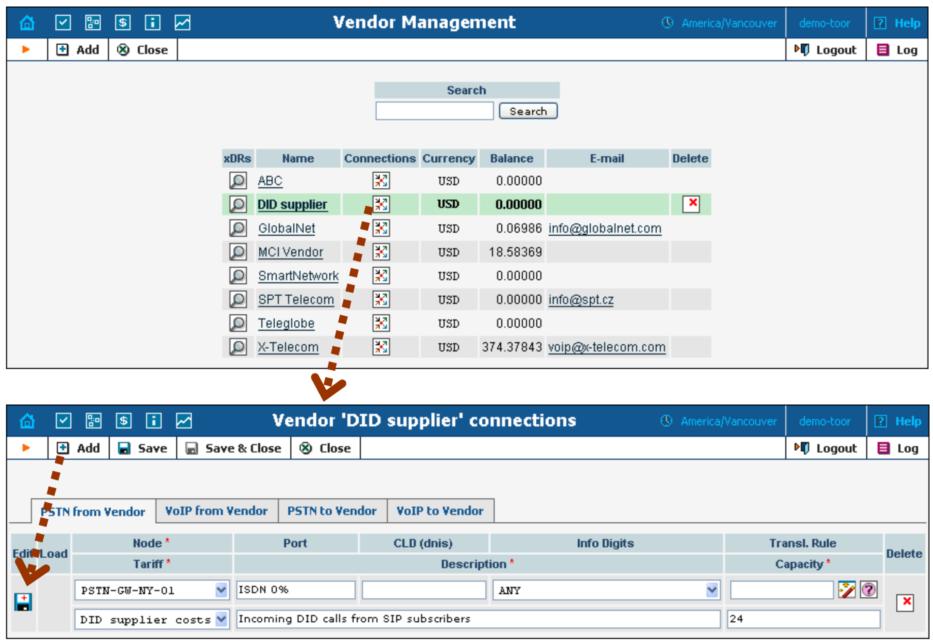
Save&Close

Vendors

Define Connections

Vendors

Connections



PSTN from Vendor

Add

Node

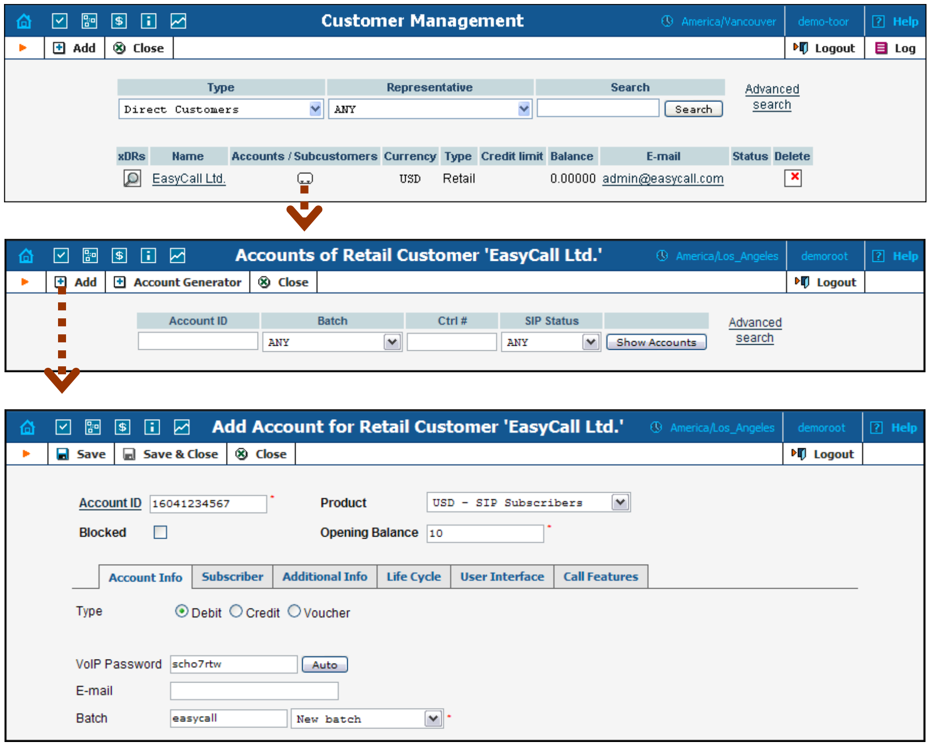
Port

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

Setting up Auto-provisioning of IP Phones

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

  -
  -
  -

- **Follow-me**
Forward

-

  - **Timeout, sec**

Save&Close

Incoming DID calls (from a VoIP vendor)

Checklist

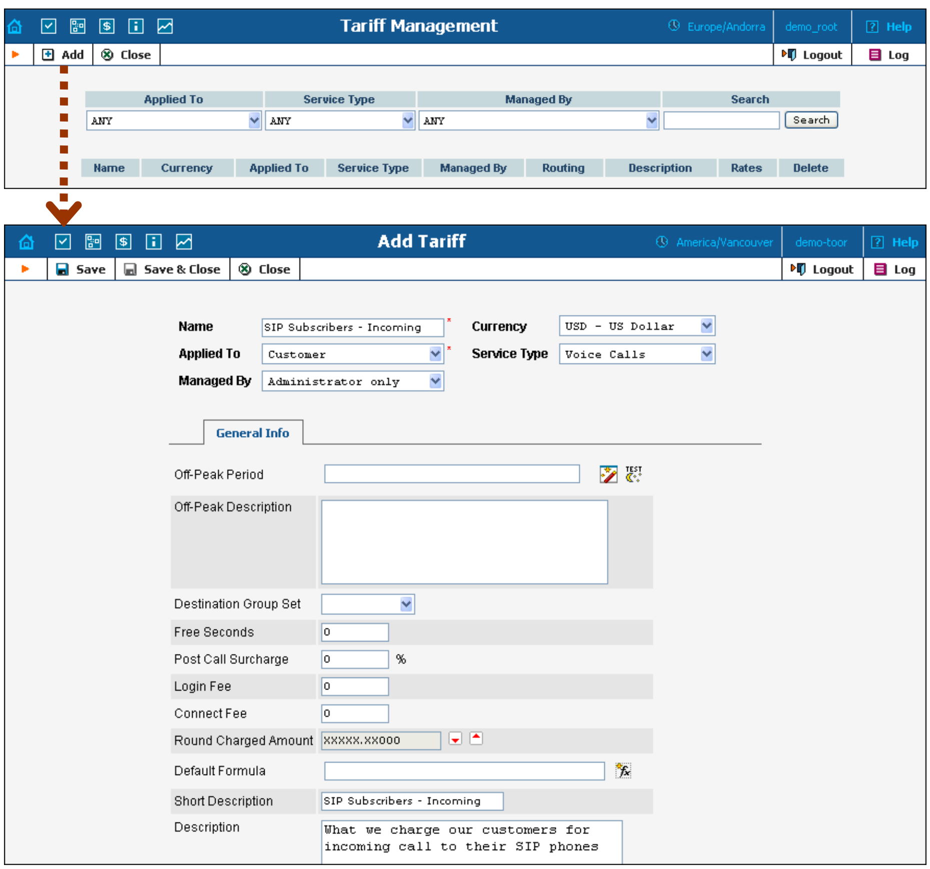
Operation	Done
Initial configuration	
Basic SIP service	
Basic SIP service	
Rating configuration	
Routing	
Account provisioning	

Basic SIP service

Testing

Set up a PSTN-to-SIP Service

Create a Tariff to Charge Your Subscribers



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

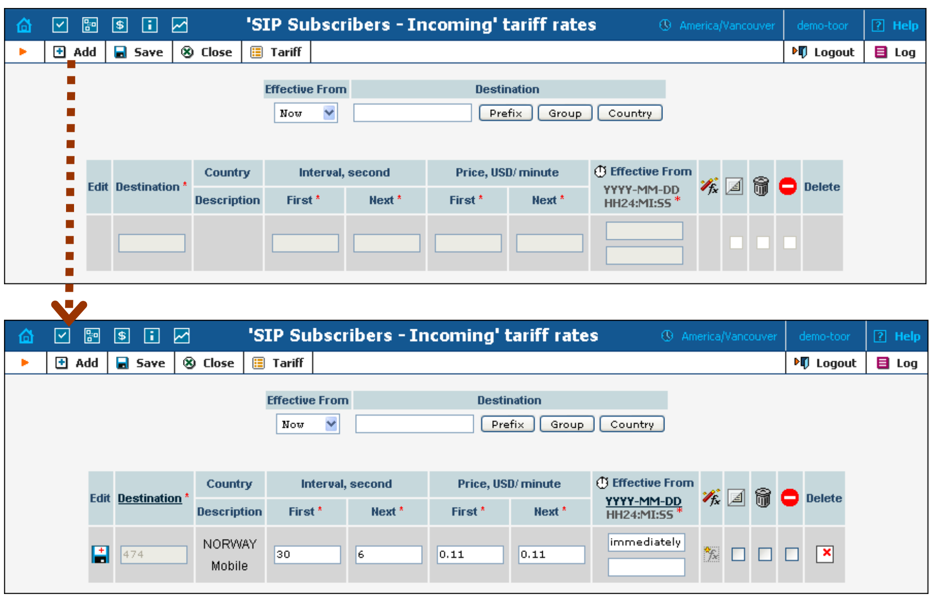
- Off-peak Description
- Destination Group Set
- Free Seconds
- Post-Call Surcharge
- Login Fee
- Connect Fee
- Round Charged Amount
- Default Formula
- Short Description

- o Description
  Save

Enter Rates

System Concepts

Managing rates online



Rates

Rates

Edit Rates **Add**

- o **Destination**
  47
  Destination

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

- o **Interval First**
- o **Interval Next**
- o **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

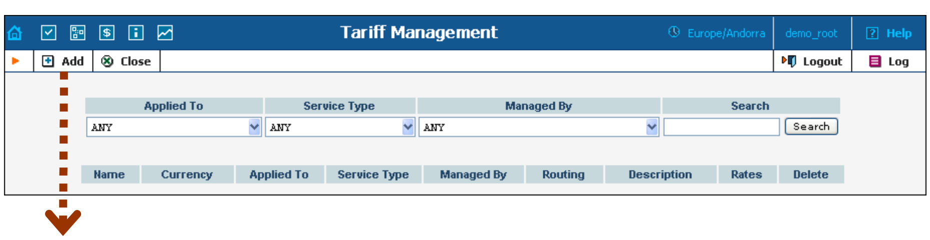
NOTE: When using the calendar, you can specify that the date you are entering is in a different time zone than your present one. PortaBilling will then automatically adjust the time.

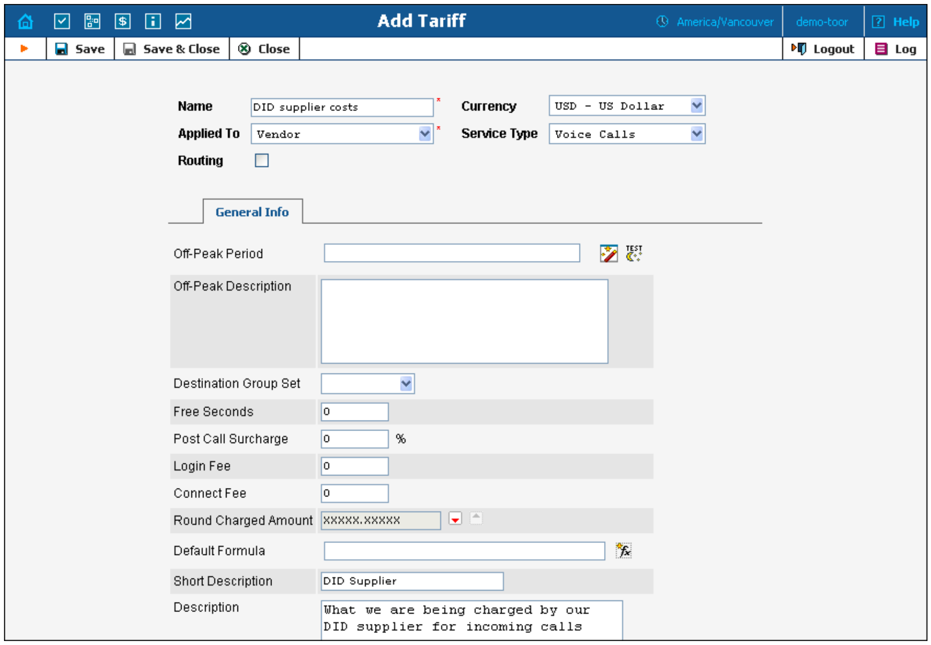
- Hidden Forbidden Discontinued
  Save

Uploading a rate list from a file

PortaBilling Templates Guide

Create a Tariff for Incoming DID Costs





Tariffs

Add

Add Tariff

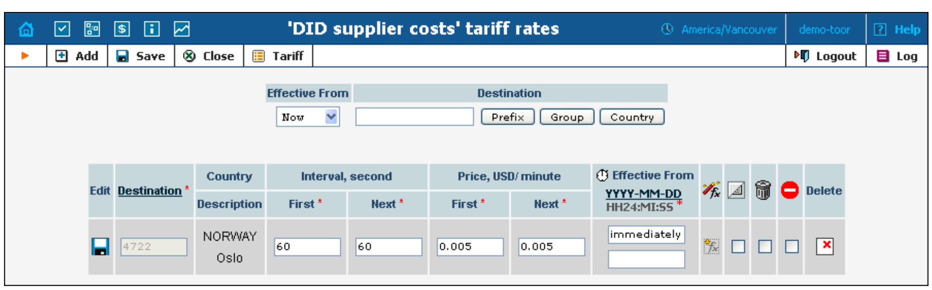
Applied To

Vendor

Routing

Save

Enter Rates



Rates

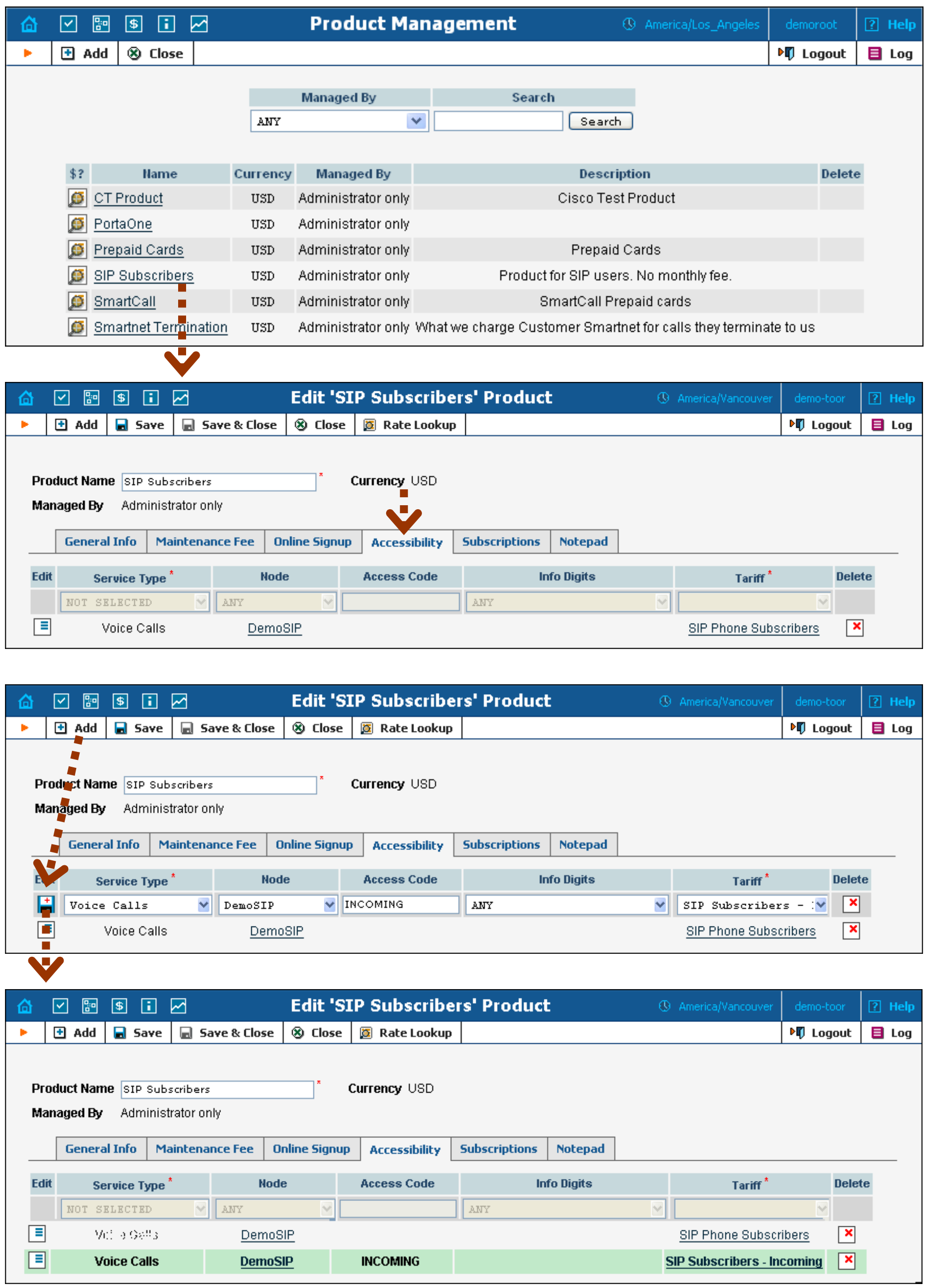
Rates

Edit Rates

Add

Save

Modify the Accessibility for the Product



Products

Add

Voice Calls Service Type

INCOMING Access Code

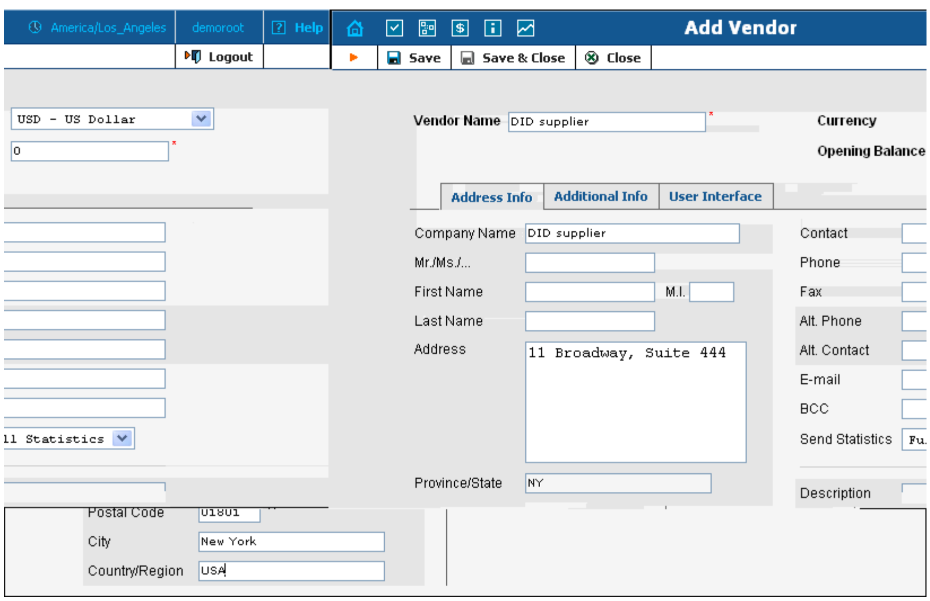
Info Digits

Save

Create a DID Supplier Vendor

Vendors

Add

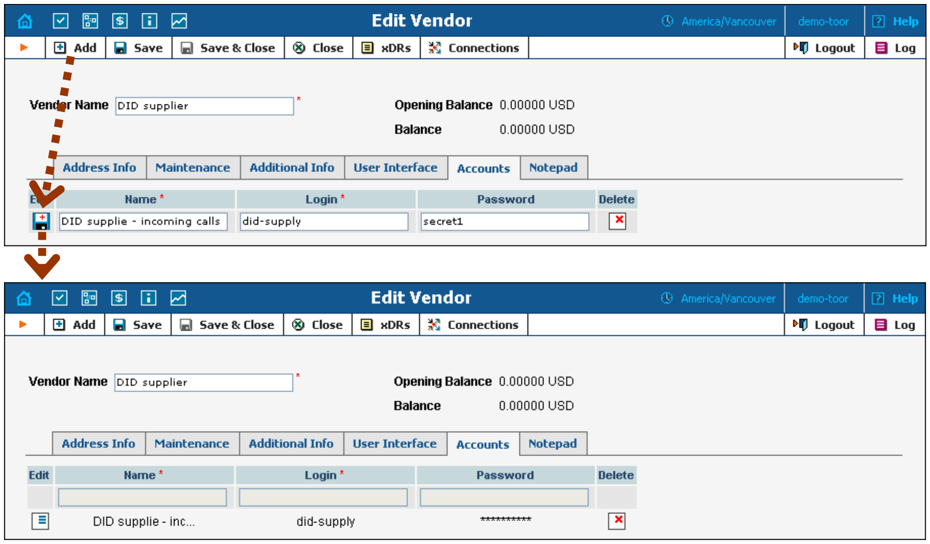


Add Vendor

Basic SIP service

Save

Add



- **Name**
- **Login**

cisco

- **Password**

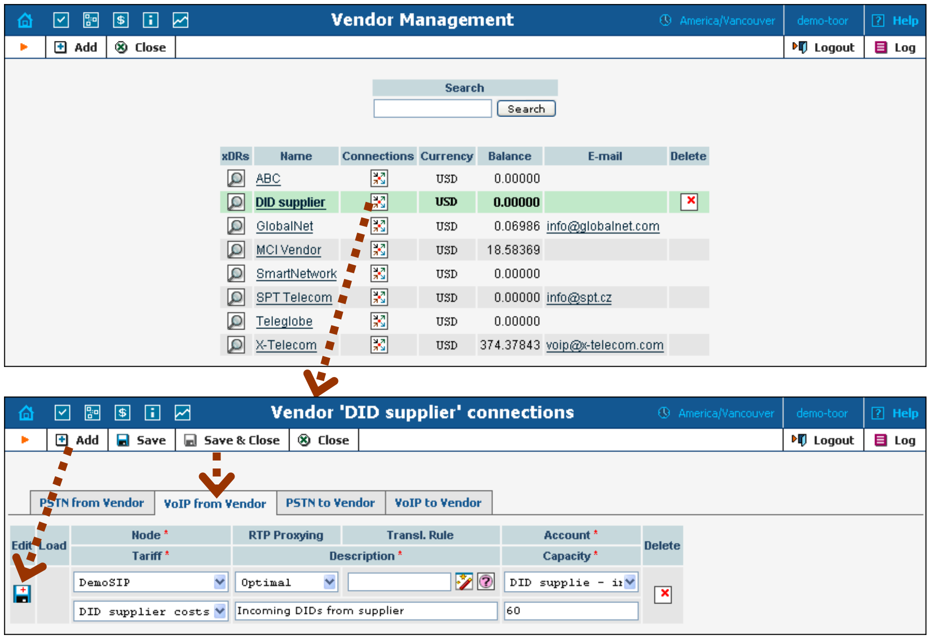
Save

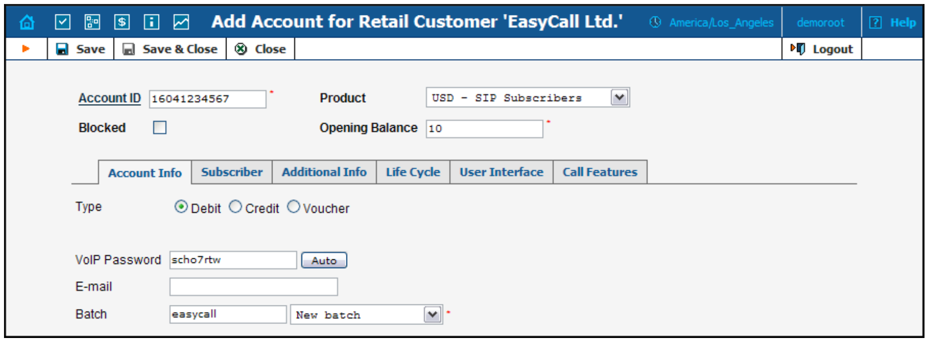
Close **Vendors**

Define Connections

Vendors

Connections


Vendor Management
America/Vancouver
demo-toor
Help
Add
Close
Logout
Log
Search
Search
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
America/Vancouver
demo-toor
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 *
Tariff *
Description *
Capacity *
Delete
DemoSIP
Optimal
DID supplie - in
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**

  -
  -
  -

  - **Follow-me**
    Forward

  -

- **Timeout, sec**

Save&Close

3. Setting up Auto-provisioning of IP Phones

PortaBilling Web Reference Guide

Setting up Auto-provisioning of IP Phones

Checklist

Operation	Done
General configuration	
IP phone inventory	
Assigning a phone number to the IP phone	
IP phone settings	
Testing	

Create an IP Phone Profile

IP Phone Profiles.

Add

- **Name**
- **Managed By**

Administrator Only

- **Type**
- **As Copy Of**

None

- **Effective From** **immediately**

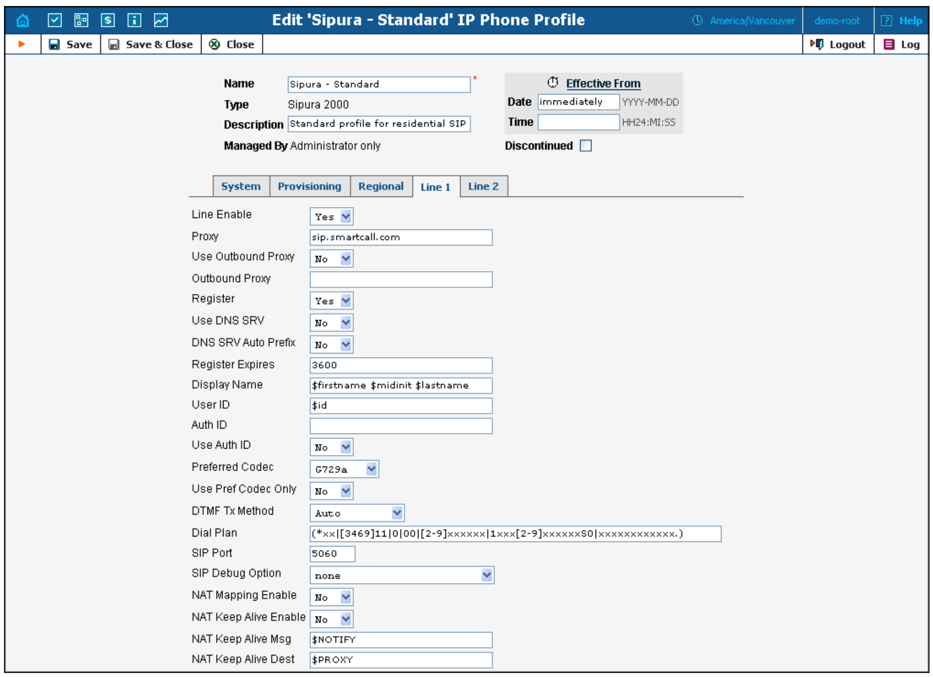
Save.

IP Phone Profile Settings

Save

GPP A, GPP B, GPP C	Profile Rule

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

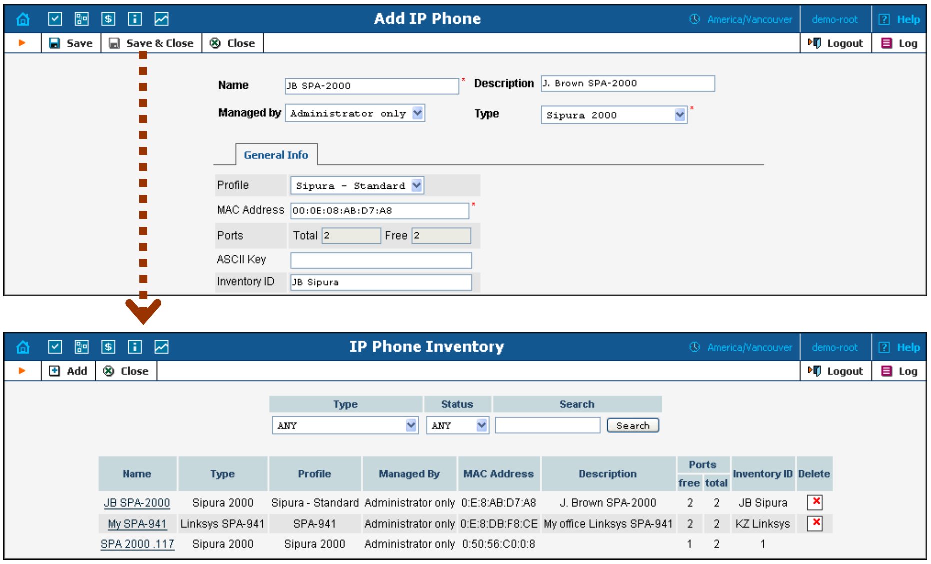


Line 1 Line 2

Field	Description
Proxy	
Outbound Proxy	
Use Outbound Proxy	
Register	

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

Create an IP Phone Entry



IP

Phone Inventory.

Add

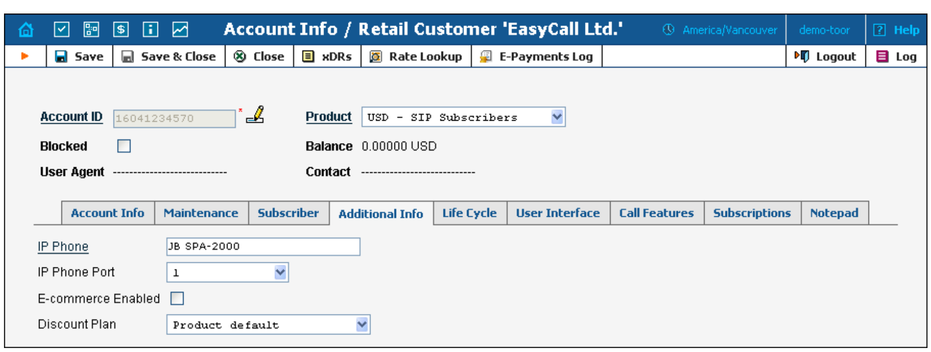
- Name
- Type
- Profile
- MAC Address

- Ports

Save & Close

Provisioning an Account on an IP Phone

Additional Info



IP Phone

IP Phone Port

Save & Close

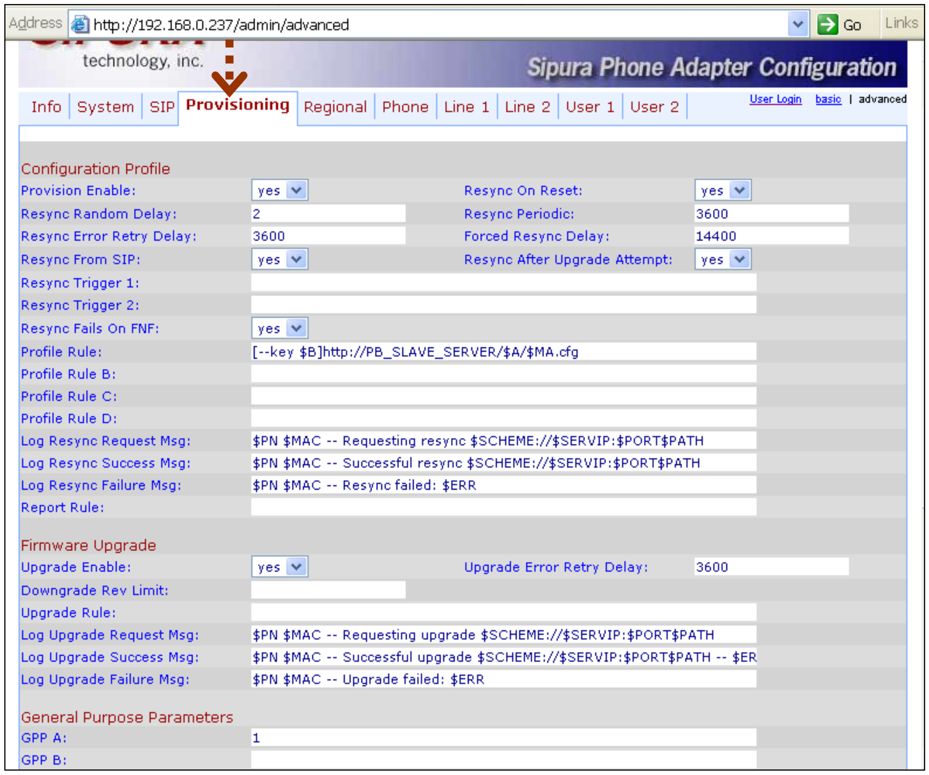
Note: The **IP Phone** select field shows a list of phones that have not been used before in other accounts, or phones with available (unused) ports.

/home/porta-admin/apache/htdocs/

IP Phone Device Configuration

Admin login **Advanced**

Provisioning



- **Profile Rule**

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

- **GPP A**

- **GPP B**
  Submit All Changes

Advanced Provisioning Tips

/usr/home/porta-admin/profile

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

home/porta-admin/apache/htdocs

DocumentRoot

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

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

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

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

/etc/inetd.conf

4. Appendices

APPENDIX A. Clients' Cisco ATA 186 Configuration for PortaSIP

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

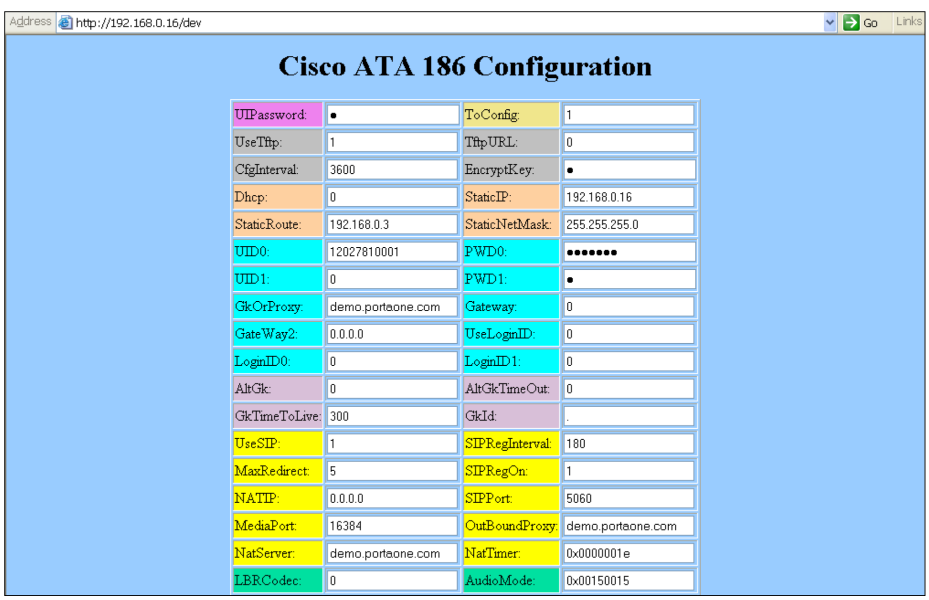
AltNTPIP	
DNS1IP	
DNS2IP	
UDPTOS	
SigTimer	
OpFlags	
VLANSettings	
NPrintf	
TraceFlags	

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

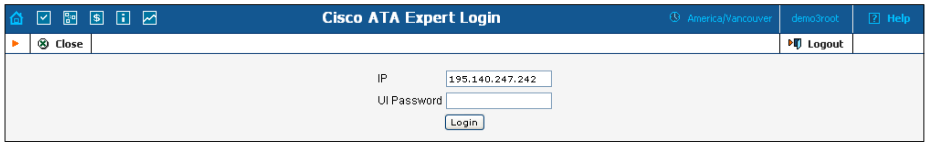


Cisco ATA

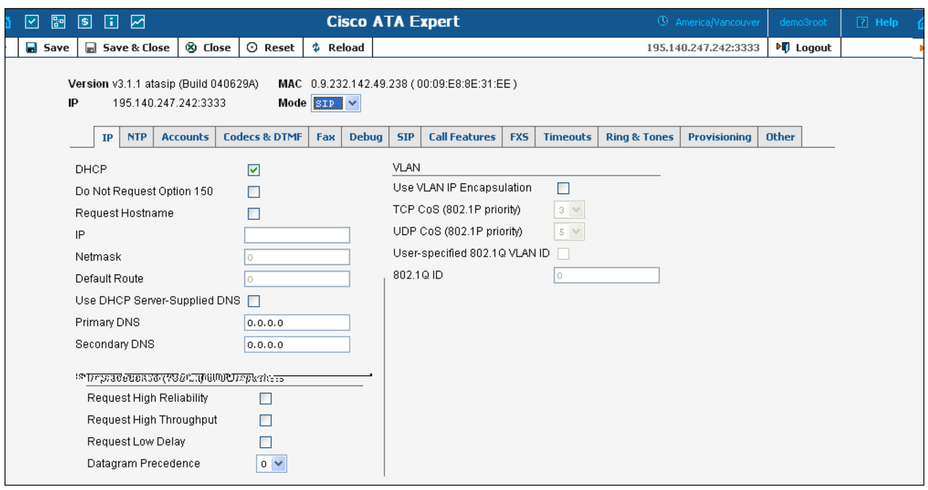
Expert

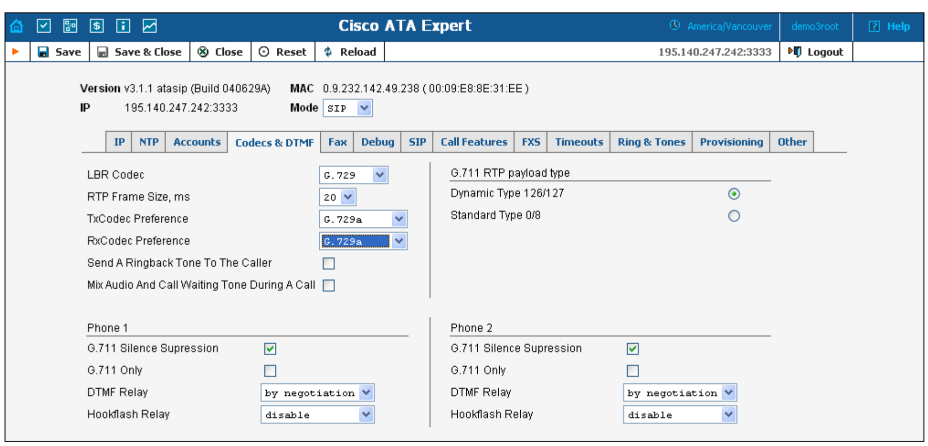
Cisco

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

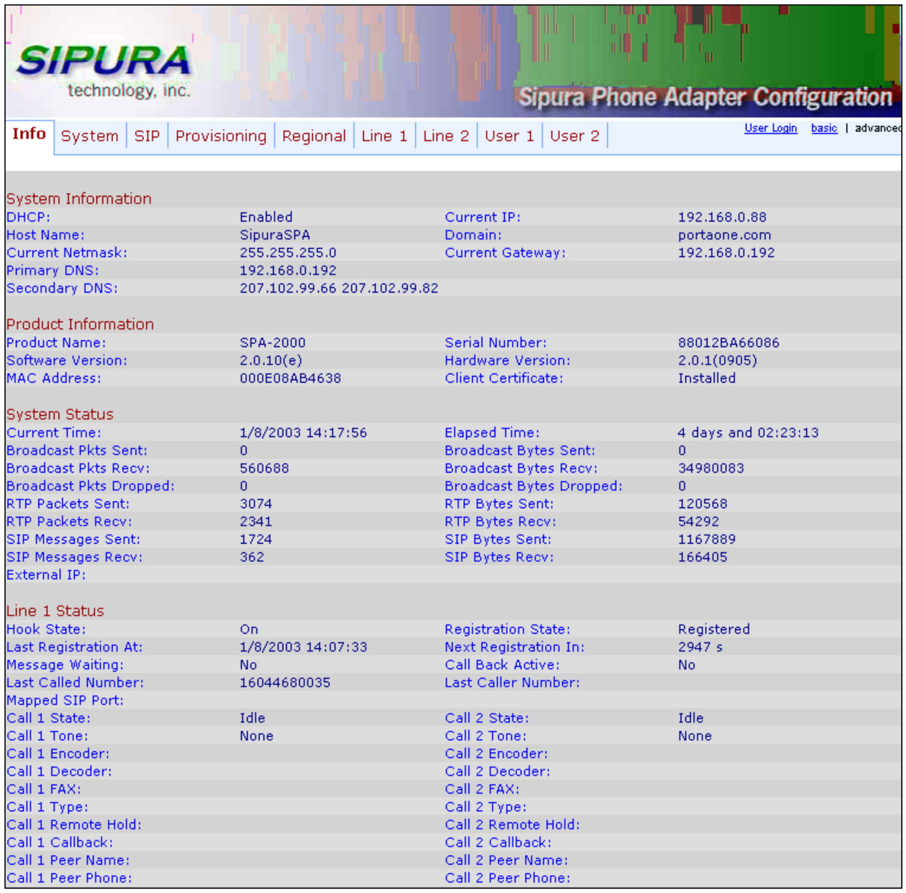
Subscriber

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

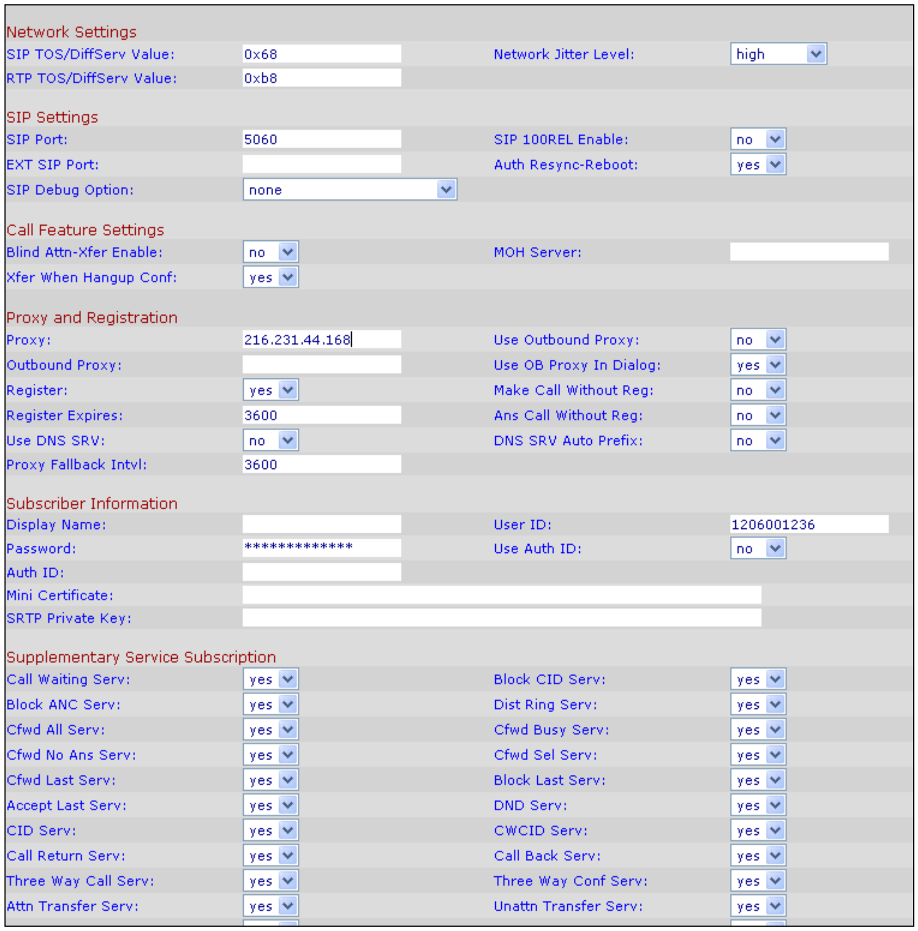
Product Name:	SPA-2000	Serial Number:	88012BA66086
Software Version:	2.0.10(e)	Hardware Version:	2.0.1(0905)
MAC Address:	000E08AB4638	Client Certificate:	Installed

System Status

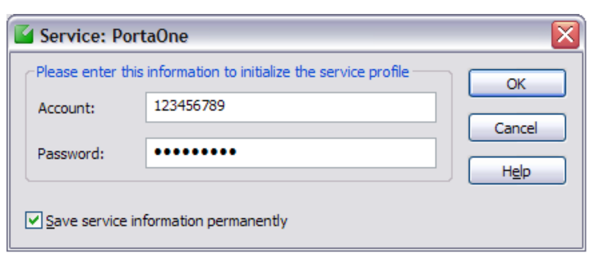
Current Time:	1/8/2003 14:17:56	Elapsed Time:	4 days and 02:23:13
Broadcast Pkts Sent:	0	Broadcast Bytes Sent:	0
Broadcast Pkts Recv:	560688	Broadcast Bytes Recv:	34980083
Broadcast Pkts Dropped:	0	Broadcast Bytes Dropped:	0
RTP Packets Sent:	3074	RTP Bytes Sent:	120568
RTP Packets Recv:	2341	RTP Bytes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

Line 1 Status

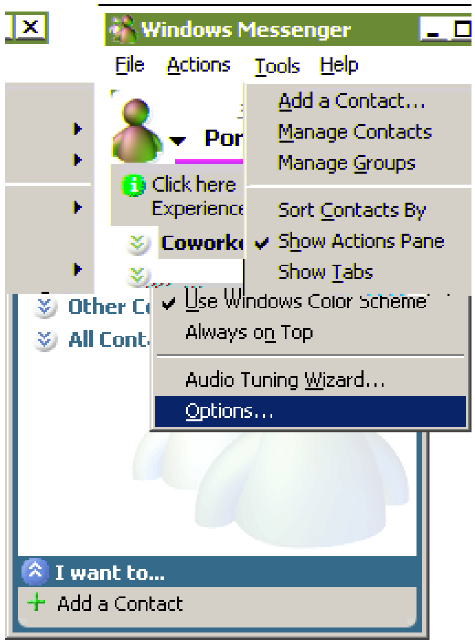
Hook State:	On	Registration State:	Registered
Last Registration At:	1/8/2003 14:07:33	Next Registration In:	2947 s
Message Waiting:	No	Call Back Active:	No
Last Called Number:	16044680035	Last Caller Number:	
Mapped SIP Port:			
Call 1 State:	Idle	Call 2 State:	Idle
Call 1 Tone:	None	Call 2 Tone:	None
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:	



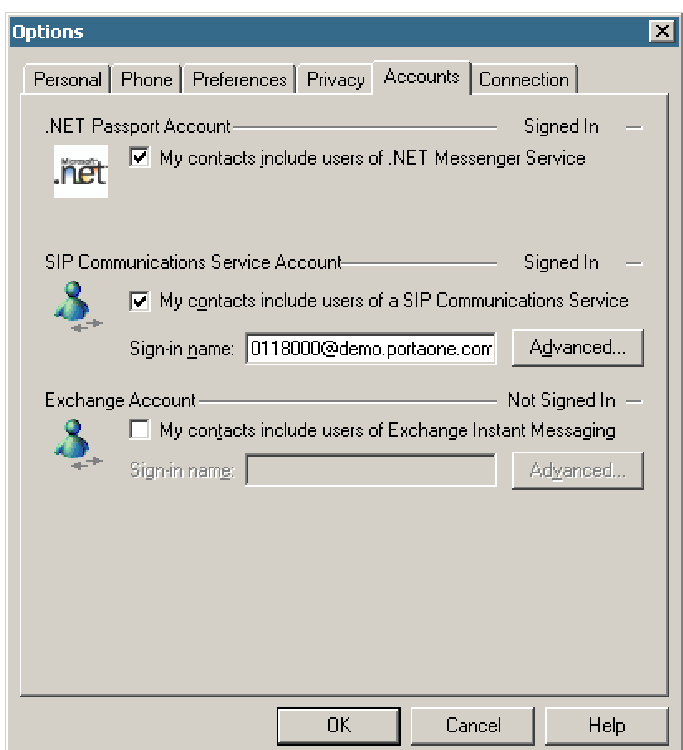
APPENDIX D. SJLabs Softphone Configuration for PortaSIP

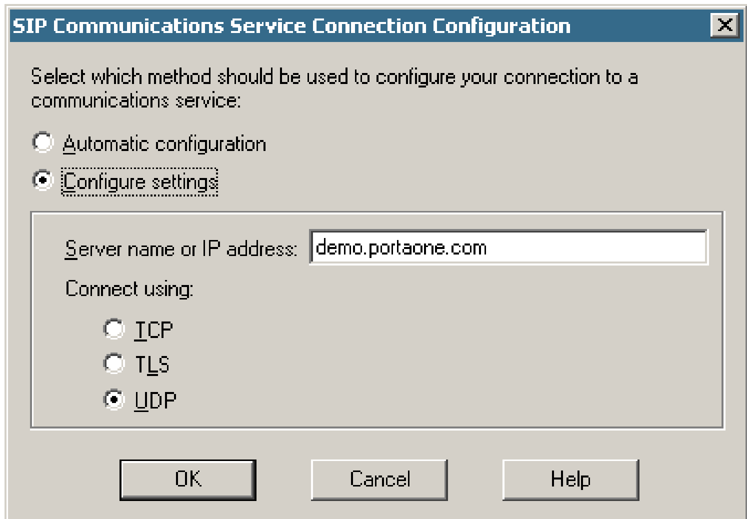


APPENDIX E. Configuring Windows Messenger for Use as a SIP User Agent

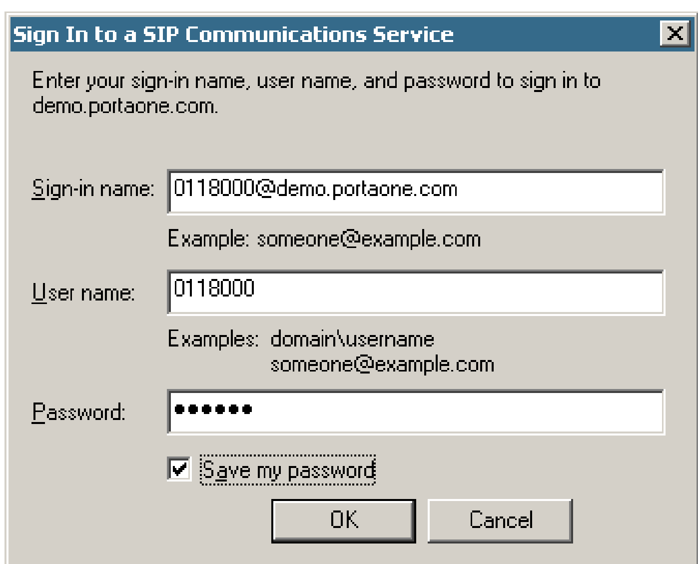


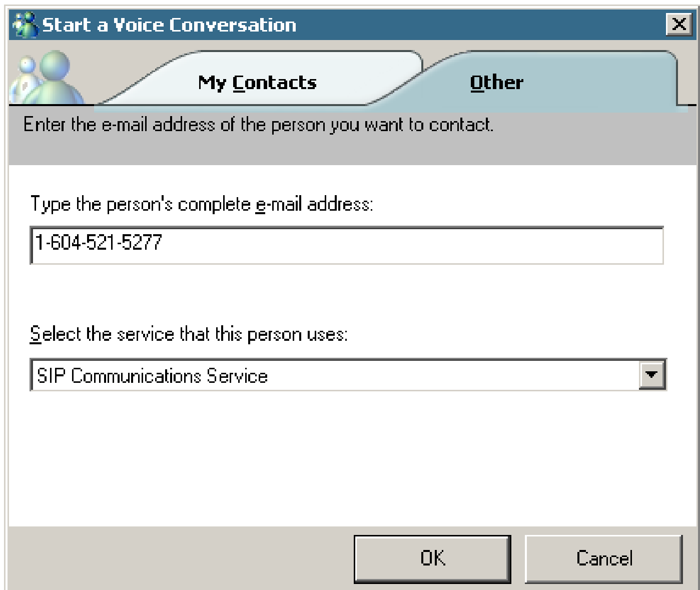
username@address *username* *address*





username@address *username*
address





APPENDIX F. Auto-provisioned IP Phones and Adapters