Astbill Implementators Manual

DRAFT

Version 3.0

For astbill version 0.9.0.16

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You will find the latest version of this document at
http://users.tpg.com.au/adslgw22/downloads/

Index:

More Topics to be added here:

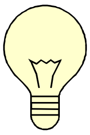
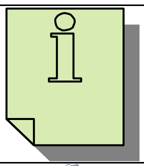
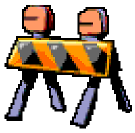
Preface:

This manual is intended for the use of astbill implementators and not for the astbill administrators.

Implementators generally use different tools and methods to setup a customers installation. This manual will provide guidance to them in the internal structure and workings of the astbill system.

A separate section titled **System Administrators Manual** is included in the end of this document, which when finished will be made as an individual download. The notes for this section are incomplete and it is a work in progress.

Legend:

		Tip or Idea
		Information
		Caution: be careful of what your are doing
		Work in progress or under construction
		STOP: Don’t do it

Introduction:

AstBill is not only a web-based, user friendly billing interface for Asterisk and VOIP. It is also a Asterisk configuration and GUI management tool and a standardized implementation of Asterisk using REALTIME and static configuration as you please.

AstBill is open source software licensed under the GPL, and is maintained and developed by a community of users and developers. AstBill is free to download and use. If you like what AstBill can do for you, please work with us to expand and refine AstBill to suit your needs.

Astbill Features:

Here are some of the features of **AstBill**:

User-friendly End User Web interface gives access to a range of functionality:

- Personal Contact Directory with Categories
- View SIP, IAX and Virtual Accounts
- Virtual Accounts You can forward your calls to any extension you want
- Time based forwarding and Billing for Asterisk and VOIP
- Credit Control on outgoing calls
- Show Balance, Expenditure, Payments and number of Calls on each account
- Set warning balance for email when low credit on account
- View Numbers Dialed and add them to the Contact Directory
- View Numbers Dialed by Names from the Contact Directory
- Dynamic International Rate Table (Each customer can have his own price list using Brands)
- Rate Table in Currency of choice
- Call Data Records including cost of each call and time based billing
- Call Data Records in his Currency of choice
- Switchboard (Displays live status of users phones and ongoing calls)
- Allows one click calling from GUI and direct to phone
- Call Parking sends calls to parking and then redirects to phone
- Allows transfers of calls
- Edit your Account setup
- Asterisk Billing and Management
- Edit voicemail setup including email and pin
- Create Time Based Dialing and billing. You to forward your calls based on time and day.
- Each user can have unlimited of SIP, IAX and Virtual Accounts
- Each user can have unlimited Prepaid Card Accounts linked to his userid
- Specify your hardware and change the viewable name of your accounts
- Temporary disable SIP, IAX or Virtual Account
- Manage your Incoming Public Numbers including Time based forwarding

User-friendly Administrator Web interface gives access to a range of functionality:

- Show Balance, Expenditure, Payments and number of Calls on each account
- Call Data Records including cost and Sales on each call
- Branding Module. Allows you to create Brands in any Currency
- For each Brand define Currency, Billing Increment, mark-up and connection charges
- Flexible Dynamic International Rate Table for Each Brand in any Currency
- Server Status (Displays live status of users phones and ongoing calls)
- Show Peers. List of the last clients (SIP and IAX2) that have connected to the Asterisk server.
- Audit Trail. Show IP, Port and UserAgent for each call
- Manage your Incoming Public Numbers including Time based forwarding
- Manage Trunks. You can use unlimited ZAP, IAX and SIP trunks.
- Time Based Trunk Dialing. Each trunk can have his own time based dialplan
- Temporary disable trunks
- Trunks can be rated after cost. Allow for cost based Dialing
- Define maximum concurrent outgoing calls on each trunk
- If lowest cost trunk is fully used (busy) the system will choose the next available trunk.
- Define unlimited outgoing routes and link them to your trunks

- Store cost of your outgoing route for each trunk for efficient cost control
- All outgoing routes are stored independent on the client price list
- Define customer price lists for each Brand and Currency
- Billing Routing and Management software for Asterisk and VOIP
- Advanced customer management and portal management
- Integrated E-commerce module and web shop is available under GPL
- Define list of VOIP hardware commonly used
- Full Hardware Inventory. Store mac address and serial numbers of client hardware
- View and Store Customers payments
- Asterisk Billing and Management
- Manage Pre Paid and Post Paid customers. Full Credit control by User Account
- View important Server logs from web interface
- Define maximum concurrent calls on each Customer Account

AstBill includes the functionality needed by most small- to medium-sized businesses (SMB). It is also very efficient as a platform for small VOIP providers. AstBill is used by several small businesses and is the platform for http://astartelecom.com the new European Asterisk call termination service currently in beta.

Astbill Requirements:

Hardware:

The min requirements are: Any PC or PC compatible with a min of PII 300Mhz processing power with 128MB Ram and 2-4GB Hdd, Floppy & CDROM.

The recommended specifications are: Any PC or PC Compatible with P4 2000Mhz+, 1GB Ram and 10Gb+ HDD & CDROM.

Software:

Linux O/S, Knoppix, Dsl, Debian, Centos etc ..

PHP - The recommended php version is 5.0.4

Asterisk - Version 1.2.x (Most of AstBill is also working on Asterisk 1.0.9 but the 1.2x is needed for the new REALTIME features)

MySQL - Version 5.18 Astbill is using Views and Stored Procedures, which is only in 5 and up)

Apache - Version 2.0.54 (It should run fine with Apache Version 1.x but it's only been tested on 2.x)

Drupal- Version 4.6.3 (Drupal is GPL software that allows an individual or a community of users to easily publish, manage and organize a great variety of content on a website.)

LiveCD:

The LiveCD is a self contained fully installed and configured installation of astbill + php + apache + mysql. All you need to do is boot the server from the Cdrom and you have a working server.

The minimum ram required for the astbill cd is 256 mb ram.

Use it mainly for demo purposes as you will not be able to save any of your changes to the system.

You can also install this livecd onto your hard drive and then work of the hard drive. It will install all programs and dependencies for you. Any changes that you do then are permanent and are saved onto your hard drive.

For more details on the operating system setup and its instructions, please refer to www.damnsmalllinux.org or the knoppix website.

Installation Instructions:

Please refer to the up-to-date install.txt for more information on how to setup your astbill solution.

First of all, make sure you have all the requirements set up properly. If you are having problems please take a look at the Further Advice section.

Windows Specific:

Astbill will only work on a windows operating system if used as a remote astbill solution. You still need to run the asterisk server on other Linux servers.

Apache 2.x or IIS 5.1 or Higher
PHP 5.0.4
Mysql 5.0.18
Drupal

Or you can download the latest version of Xampp from www.apachefriends.org . Xampp has apache, php, mysql & phpmyadmin all integrated into one simple install.

Follow the instructions in the drupal install.txt file for windows. Once you have a working website using drupal. Then create the astbill databases in mysql using the supplied mysql database files.

Then edit the settings.php file in drupal to change to astbill database and add the prefix "pbx_"

Change the location of your astbill.conf module to the root of your webdirectory.

You need to manually edit the astbill modules to change the astbill.conf location variable in these modules to the correct one.

Then you will be able to get the channel and the server status.

Note: When running on windows + iis, you need to enable the gd2.dll in php.ini and set the email server details as well.

Support

Don't forget that this is free software under development! Chances are good there's a crucial step that hasn't made it into the documentation. If you have any problems feel free to contact us on the forum http://astbill.com/forum/3

Please provide enough information to work with, and preferably be aware of what you're doing! And keep track of major changes to the software, including performance improvements and security patches.

The best way to get support is by using the forums. This allows other users to learn from your request.

http://astbill.com/forum/3

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Configuration:

System Setup:

Before you start using astbill, you should first customize it to suit your needs. By default astbill uses UK defaults. To change the defaults to suit your setup, you need to do the following:

You first need a blank database without any of the demo user accounts or any of the other data. Or you can use the existing one supplied by astbill. Then use phpmyadmin to empty the tables.

You need to use phpmyadmin or the astbilladmin program to maintain the data in your tables:

Download & install phpmyadmin from their website and configure it using the username and password of your astbill mysql username & password.

The order of customisation your astbill solution is as follows:

1) Edit the table astsystem and see if the defaults suit you.
2) Add or Edit the astcurrency table for creating new currencies and editing old ones.
3) Edit/Add brands in the astbrands table.
4) Edit/Add prices in the astprices table for your brand
5) Edit astusers table and change the users table Ex: country & currency. or alter table structure to set your defaults.
6) Create trunks (asttrunk table)
7) Edit/create routes in astroutes for the above trunks

Remember all new accounts created automatically by astbill defaults to IAX only, you need to manually change them to SIP if you need a sip connection and belong to the default brand.

System Defaults:

Database Structure:

Astbill version 0.9.0.16 has a total of 109 tables & views. Tables with the prefix pbx_ belong to the drupal cms application.

All astbill Tables start with ASTxxxxx and all astbill Views start with ASVxxxx

The tables contain the actual data, where as the views show the data from the tables.

astsystem:

serverid (*)	name (*)	value (*)	comment (*)	viewstatus (*)
DEF	accountstart	7	This is the first digit of the automatically created account number. Automaticly created account number must be numberic	E
DEF	accountlength	5	This is the length of the automatically created account number	E
DEF	firstaccount	72000	If there is no account defined in the 7 range. This will be the first account created	E
DEF	TopUpURL	/topup/	The URL used to topup AstBill VOIP Software	E
DEF	LOGMessages	2000	This is the numbers of chars visible from the Asterisk Messages Log File	E
DEF	AccountDropDown	1	If 1 We will use a Drop Down box to show Accountcode in Payment screen and other Screens. If 0 a text entry box will be used.	E
DEF	GlobalDialPrefix		It is common to Add 00 as Global Dial Prefix. This allows users to dial international numbers without 00 or 011 prefix	N
DEF	MaxMinute	60	This is the maximum alloved minutes for a call. If your call last longer than this you will be disconnected by the System.	E
DEF	AstBill-DB-Updated	2006-02-14	This is the date of the last Database patch applied to your system	D
DEF	AstBill-Version	AstBill-0.9.0.14	The Official Version number of your AstBill Installation	D
DEF	AstBill-GUI-Updated	2005-12-08 08:20:44	The last date your Web GUI have been updated	D
DEF	Log-Path	/home/astbill/logs/	This is the Path to log files created by agi-bin scripts	E
DEF	def-voicemail-pin-length	4	The Voicemail PIN/Password Length to use when creating new accounts	E
DEF	def-password-length	6	The Password Length to use when creating new accounts	E

Accountstart: This field specifies the FIRST DIGIT of all the user accounts. This impacts on the LOCAL Numbers. By Default it is “7”. You may choose any starting digit. If you change this digit here, be sure to change the value in the field "pattern" in the astroute table for the local trunk.

Accountlength: The default value is 5, which means that your account numbers will consists of a five digit number. By changing the length here, you need to update the field "patternlen" in the astroute table for the local trunk.

Firstaccount: The value in this field is used to create the very first account. This is used in a fresh install with a blank database. If you have any existing accounts starting in this range, then the system will continue on with the last created account.

TopUpUrl: This value specifies the URL Node where TopUp's to an account can be done.

LogMessages: Useful on a very busy system. Shows the number of characters from the asterisk message log file as defined here.

CountryPrefix: Set this value to the country code of your choice. The Default value here will be used when creating any new user.

AccountDropDown: This setting enables how the accountcode is filled in the payments screen. If it set to 1, then astbill shows a dropdown list of the accounts to choose from. If set to 0, then astbill will show a text box to fill the accountcode in manually.

Recommended is to use 1, as this will enable correct accountcodes to be selected.

GlobalDialPrefix: It is common to Add 00 as Global Dial Prefix. This allows users to dial international numbers without 00 or 011 prefix. So if you enter 00 or 011 in this value, your users need not dial this prefix. Astbill will use this prefix.

MaxMinute: This is the maximum allowed minutes for a call. If your call last longer than this you will be disconnected by the System. The default value is 60 minutes.

The recommended setting is 30 minutes.

	Don't change your system setup once you have started using astbill. This will break the astbill solution. So plan ahead and set your defaults accordingly.

Astcurrency:

currency (*)	ratetabledesc (*)	currencysymbol (*)	currencysymbol 2 (*)	centsymbol (*)	currencyrate
GBP	UK Pence	£	£	p	1.0000
USD	US Cents	$	$	c	0.0000
NOK	NOK ore	NOK	NOK	ore	0.0000
SEK	SEK ore	SEK	SEK	ore	0.0000
EUR	Euro Cents	€	€	c	0.0000
AUD	AUD Cents	$	$	c	0.0000

Define your currencies in this table. This will be used in the Brands table to create your astbrands and in the astusers table

Currency: Input the short currency code
CurrencySymbol: The symbol for this currency
Currencysymbol 2: The second currency symbol
Centsymbol: the symbol to show cents
Currencyrate: the currency rate for this currency. This is currently not active.
Ratetabledesc: This field describes the currency

Untill the foreign currency rates are implemented, it is recommended that all your costs and sales prices be set in a single currency systemwide.

Astcountrycode:

This table holds the country codes and the name of the country.

db prefix	pbx_ The db prefix by default is pbx_ This is usefull when running multiple sites from one database installation.
countrycode	The numeric code for the country
country	Country Name
usstate	The state name. Applies to US only
idd	The idd number for the country. Ex: the prefix code 00 before the international Tel number

This data in this table is pretty much stable and you may have no need to change them at all.

Astplans:

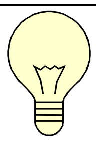
Define your plans in this table.

name (*)	currency (*)	language (*)	publishednum (*)	did (*)	markup	billincrement	connectcharge	status
Star	GBP				50.00	1	1.00	1
Hiper	GBP				50.00	6	1.00	1
default	GBP				0.00	30	1.00	1

name	The name of plan
currency	The currency this plan will use
language	
publishednum	
did	
markup	Amount in decimal amount
billincrement	In seconds (time)
connectcharge	Amount In decimal amount
status	0= inactive, 1=active

You need to have at least one plan. The default plan can be customized for your needs.

From the above example you can see that the default plan is the most expensive on terms of the billincrement flag and it becomes smaller as the markup increases.



Remember to keep your default plan the most expensive one. The other plans can be setup based on cost, quality & importance.

Astpricelist:

You need to define your pricelists. You need at least one pricelist for each route.

id	A numeric identifier, needs to be unique
countrycode	The country code
pattern	The pattern to match a dial string
ccn	
name	Destination name
weight	0
connectcharge	Connect Charge for this call in decimal format
includedseconds	List and included seconds in the above connect change
minimumprice	The minimum price in decimal amounts
price	The price
brand	The plan to which this price line belongs to
txtcomment	

You need to setup a pricelist for every destination, for every plan you setup.

So if you have a plan called "default" then you need to enter the prices for your destinations here for this default plan.

Pricelist are the prices that you charge your customer.

If you do not have a price setup for any dialed destination they astbill will log an error in the astlog table.

Remember: to have at least one price for every country in the world. Do not leave any country out. For example:

Country Code =61 Pattern =61 Price=0.30
Country Code =62 Pattern =62 Price=0.35 & etc …

Asttrunk:

Define your trunks, which will be used by your Routes.

tid	The trunk id, auto generated by the system.
name	Name of the trunk, should be a unique name
tech	The technology used for this trunk. Local, Zap, Iax2 etc.
path	Only used if you want to use trunks created in your static files, sip.conf etc. if so then add "@" to the beginning of the trunk name used in your static configuration.
static	R=realtime
isdefault	1=yes default, 0=NO not default
serverid	Not Used currently
comment	Comments about this trunk
vat	If you want to add VAT to the Vendors prices before we calculate our cost. This field will make astcdr.ourcost include VAT. This is useful if you are not able to get your VAT back from your Vendors:
vat 2	If you don't want astcdr.ourcost to include the VAT but want to store the rate for use in reports. Update the vendors VAT rate. This rate is not used during calculations of astcdr.ourcost:
usagecount	This field is updated by astbill when calls occur, do not change this. Leave at 0
maxusage	Enter the maximum number of calls via this trunk. Default is 9999
trunkcost	The cost of the trunk in relation to the other trunks in your astbill. The lower trunks are used first.
accountcode	The accountcode to which this trunk belongs, leave blank
username	Username
currency	USD
tenantid	The tenanted number, default is 0
addprefix	The prefix you want to send to your service provider. Some providers require this.
removeprefix	The prefix that needs to be removed from your dialed number, useful if your service provider expects you to send a proper E164 formatted number. So you drop off the prefix like 00 & etc
registerstring	This is the register string you enter to register with your service provider
usstyleprefix	0

Trunks are your service providers. You enter the details of your service providers here. The minimum information that you need from your provider is the registration details and the connection type.

You need to setup at least one Trunk so that your customers can diallout using this trunk.

When setting up a single trunk, be sure to tick the default trunk tick box.

If you have multiple trunks setup here, then choose one of the trunk as the default trunk.

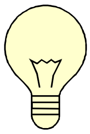
In addition to the trunks that you create above, astbill has some system trunks that are not exposed in the above interface. This trunk relates to your extensions. It is called the Local trunk. This trunk makes it possible for your customers to call each other using their account numbers.

So if you are editing astbill tables using phpmyadmin then please be aware of this requirement.

Bug Note for astbill version 0.9.0.16:
There is a bug in the trunk editing code, where by when you edit a trunk; you will loose the trunk path details. This is a bug.

Astroute:

id	A unique route id generated by astbill
pattern	The pattern you want to use this route for.
patternlen	The length of the pattern. Only useful for fixed length routes. Ex: your local users etc. otherwise leave at 0
costplan	0
connectcharge	The connect charge
includedseconds	The included seconds in the connect charge
billincrement	Billing time in seconds
minimumcost	The minimum cost
cost	The cost for this trunk
trunk	The trunk to use
timestamp	25/10/2005
countrycode	The country code
routename	The route name



Remember: Astroute are your cost prices that are charged to you by your provider.

STOP: Have you remembered to create a route for every possible destination. If you do not have a route setup for a particular dialed number pattern, then astbill will charge your customer incorrectly for this call, using default cost values.

This is an astbill design flaw and should be addressed by the developers.

Astdialplan:

did	1
accountcode	72001,
conditions	Offline
action	Voicemail
mon	Yes
tue	Yes
wed	Yes
thu	Yes
fri	Yes
sat	Yes
sun	Yes
start time hr	0
start time min	0
end time hr	23
end time min	59

This dialplan is used for setting your preferences in accepting calls and or diverting calls to voicemail based on your time preferences.

Astextensions:

id	context (*)	exten (*)	priority	app (*)	appdata (*)
1	mycontext	_574555XXXX	1	Wait	2
2	mycontext	_574555XXXX	2	SayNumber	102
3	mycontext	2815551212	1	Playback	pbx-invalid
5	cytel	8322008630	1	Dial	SIP/3044,30
7	cytel	80	1	Voicemailmain	@cytel
8	cytel	_832.	1	Dial	SIP/${EXTEN}@66.88.74.85|30
9	cytel	_9X.	1	Dial	IAX2/devasterisk:asterisk@asterisk-alpha/${EXTEN}@cytel-internal
10	cytel	3013	1	Dial	SIP/3013|30
11	cytel	_3XXX	1	Dial	IAX2/devasterisk:asterisk@asterisk-alpha/${EXTEN}@cytel-internal

This table is not in use currently and can be ignored

Asttennants:

tid	30000
company	Default Tenant
contactname	
address 1	
address 2	
zip	
city	
state	
country	
phone	
phone 2	
fax	
status	0
date created	00/00/0000
timestamp	04/10/2005
resellerid	40000

Currently being developed so do not use it now.

Astuser:

uid	0
tid	Default Tenant
brand	default
Country Prefix	United Kingdom
currency	GBP, Pound Sterling
creditlimit	0.00
callbackto	72000
lastaccount	72021
comment	default
timestamp	07/11/2005

Do not edit this table directly. The only field you should edit is the brand value (renamed as plan)

This table is a mirror copy of the drupal pbxusers table.

Any users created using the drupal administration section will also add an identical user in this table.

Note: when you edit or delete users in drupal, then the changes are not update in this table.

Astaccounts:

This table where all the action happens in astbill. Contains all user settings for asterisk to allow the customer to register , receive and make calls.

Be careful when changing any of the account codes and account numbers.

db prefix	pbx_
accountcode	72001
uid	1, anil
tech	SIP
accountname	
callerid	
forwardto	
allow	
amaflags	
auth	
callgroup	
cancallforward	yes
canreinvite	no
dbsecret	
defaultip	
deny	
disallow	
dtmfmode	rfc2833
fromdomain	
fromuser	
authuser	
username	
host	dynamic
inkeys	
insecure	
secret	90233
md 5secret	
context	default
language	us
mailbox	
mailboxpin	

mailboxemail	
pager	
stamp	
attach	yes
saycid	yes
hidefromdir	no
nat	yes
qualify	yes
emailvoicemail	1
mask	
musiconhold	
notransfer	
permit	
pickupgroup	
regexten	
restrictcid	
restrictid	
rtpholdtimeout	
rtptimeout	
type	friend
comment	
serverid	
active	1
hardwareid	1
usagecount	0
creditlimit	0.00
lowbalance	5.00
lowbalanceemail	0
date created	07/11/2005
timestamp	07/11/2005
ipaddr	
regseconds	0
port	
reg changed	00/00/0000
cccardvalue	0.0000
ccfirstused	00/00/0000
cclastused	00/00/0000
ccexpiredate	00/00/0000
ccexpiredays	0

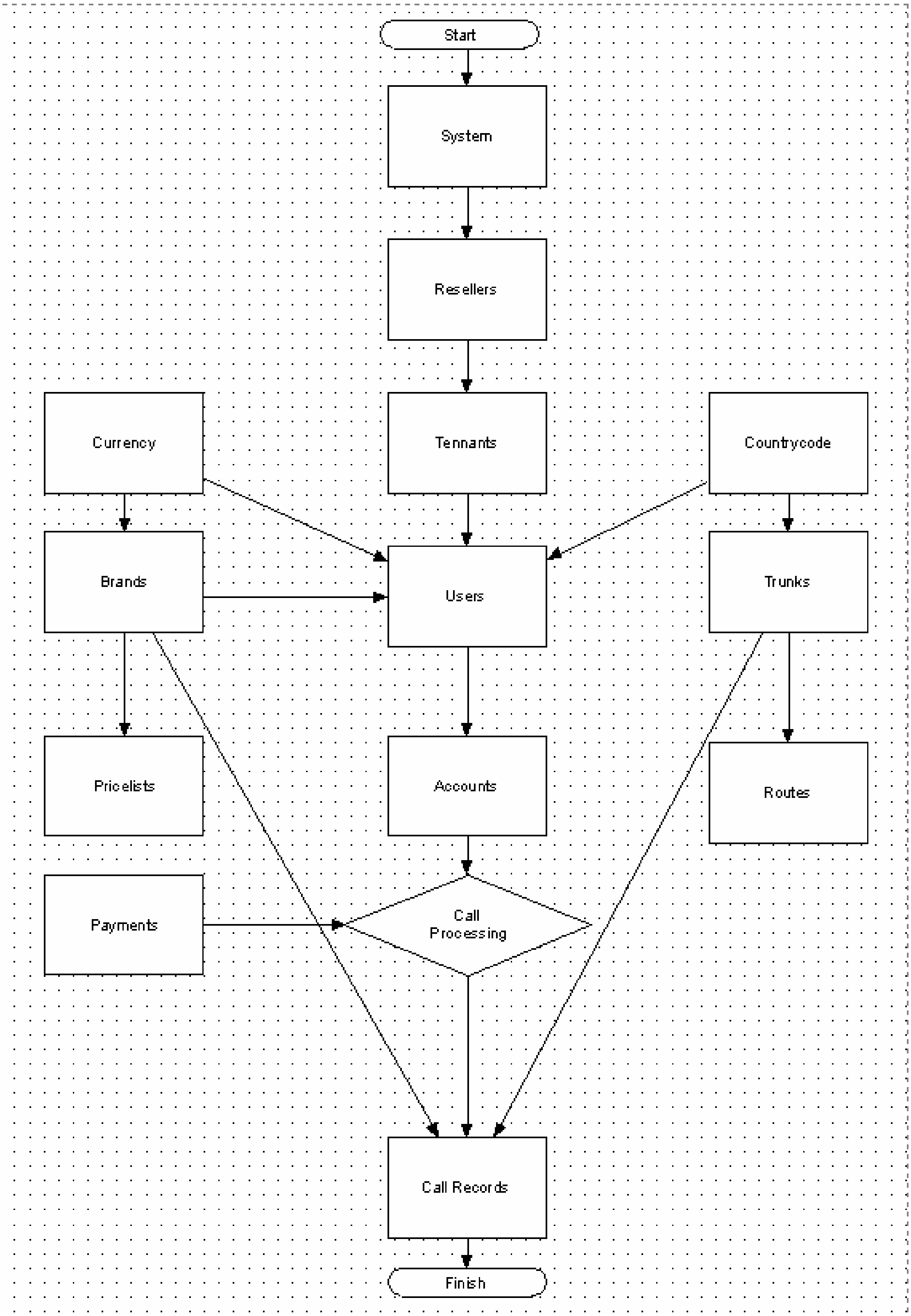
ccbatchno	
ccserialno	
startingcredit	0.0000

Astpayment:

paid	1
accountcode	72001,
paytype	Credit Card
comment	test pmt
paidamount	10.00
paiddate	08/11/2005
date created	08/11/2005
timestamp	08/11/2005

Contains details of payments received for the accountcode.

Astbill Flow Chart:



Astbill Mysql Procedures:

astCreateAcc

This is called by astCreateAccount and allocates the first available unused accountcode to the new user created. This also creates the users in the atbill database if it does not exist as compared against the drupal users table.

astCreateAccount

This procedure is run when a new account is created and checks to see if there a any unused accounts for user 0. if less than 5 then it creates additional accounts for each technology.

RateAddcdr

RateCost

RateGetTrunk

RateStarDead

TrunkDialPlan

astTestBilling

RateSale

RateReserveCredit

FAQ:

What is the default system currency?

None. There is no default system currency used at the moment.

How is the currencyrates used?

This is not used Yet. Will be used to automatically calculate price lists in different currencies.

In what currency is the user payments received in?

Same currency as the currency assigned to the account using the brand code of the user. We have to put that on the payment screen.

In what currency does the system calculate the costs and the user balances?

Same currency as the currency assigned to the account using the brand code of the user

what is the purpose for the virtual account?

What is it used for?

Virtual Account = Forward calls to any Extension

Virtual Account = You can forward your calls to any extension you want.

You can also forward the call to many extensions by entering the numbers you want to forward to separated by &

(Example: 9000&8000)

You don't put SIP or IAX in front of the number as AstBill is keeping track of the protocol used.

Personally I work in different locations during the week and I have my number a Virtual Account. This allows me to put in my Real SIP or IAX number for forwarding at any time.

It is like a number that is not bound to a location. If you put a SIP phone on your desk it is still on your desk when you are gone. When you are not there your phone is still there.

Very soon the virtual accounts will also be able to transfer calls to Mobile phones and other external numbers.

I normally put my Virtual Number in my Caller ID for each SIP phone I am using like this.
Are

If my Virtual Number is 70999. This will make anybody to call me back on 70999 and I can have the same number in any location without having to reprogram any SIP or IAX phone.

how to round dialed seconds

if someone made a call for 42 seconds
how to round it to 50?

seems that astbill rounds call to 30 sec.
how to change this?

AstBill can round anything :-)

It is in the Mysql Table astbrand.billincrement
You can edit it in the web interface using
AstBill Admin / Settings
Brand
Edit

There you have the Billing Increment: field. If you put 10 there AstBill will round up to 50 seconds. This is a Brand Dependent parameter.

Drupal User Control

I have a problem creating drupal users:

When I create a user which is only an "authenticated user" the menus that he sees are:

Contact directory
my account
log out

When I create a user (similar to astbilladmin): "authenticated user, Admin, Adoroar"

the menus that he sees are:

Contact directory
create content
my account
AstBill Admin
Incoming Friends
Call Data Record
Create Account
Hardware
Server Status
Tenants
administer
log out

I am not sure which menus the authenticated user should see, but clearly he should see more menus.

What am I missing?

Access Control

Click on administer / Access Control

I can see now that many of the modules don't have access for the Admin Role. You just need to enable the Modules you want admin users to have access to. I think All the modules are a good start. The AstBill Modules are the ones starting with ast.

By Drupal Design user number 1 (astbilladmin) have access to all installed modules and functions. He is like a Super Admin.

What is the difference between the payments menu and the submenu credit?

When we put an entry in the payments menu shouldn't it in an automatic way update the credits of a specific account?

Payments are for prepaid
Credit is for example: post paid limit.

I like to think of it as the credit limit set for the postpaid user's accountcode.

Changing the name of the AstBill database

I was doing and install today where I had to use a different name on the AstBill Database in MySQL.

It was not a problem but a few things to remember so I created a Wiki Entry.

http://wiki.astbill.com/wiki/Change_database

There are 4 files to edit if you want to change the name of the AstBill Database.

/home/astbill/astbill.conf
/etc/asterisk/res_mysql.conf
/etc/asterisk/extconfig.conf
sites/default/settings.php

We are changing the name from astbill to voipbilling

astbill.conf
dbhost = localhost
dbname = voipbilling
dbuser = astbilluser
dbpass = astbill419

res_mysql.conf
[general]
dbhost = localhost
dbname = voipbilling
dbuser = astbilluser
dbpass = astbill419
dbport = 3306
; For Debian
dbsock = /var/run/mysqld/mysqld.sock
; For Asterisk @ Home
; dbsock = /var/lib/mysql/mysql.sock

extconfig.conf
;example => odbc,asterisk,alttable
iaxusers => mysql,voipbilling,asv_iax
iaxpeers => mysql,voipbilling,asv_iax

sipusers => mysql,voipbilling,asv_sip
sippeers => mysql,voipbilling,asv_sip
voicemail => mysql,voipbilling,asv_voicemail
;extensions => odbc,asterisk
;queues => odbc,asterisk
;queue_members => odbc,asterisk

settings.php
Make sure the $db_url line matches the database defined in the previous steps:

$db_url = "mysql://username:password@localhost/database";
where 'username', 'password', 'localhost' and 'database' are the
username, password, host and database name for your set up.

The correct example for this guide is :
$db_url = 'mysql://astbilluser:astbill419@localhost/voipbilling';

Dial Features:

700 Call Parking (then Dial 701 to 720 to set position)

**1 Blind Transfer

*1 One Touch Record

**2 Attended Transfer

198 VoicemailMain

199 Voicemail for your Extension

500 Date & Time

501 Monkeys

502 Echo Test

503 It is free playback

Asterisk Commands:

! Execute a shell command
abort halt Cancel a running halt
add extension Add new extension into context
add ignorepat Add new ignore pattern
add queue member Add a channel to a specified queue
ael debug contexts Enable AEL contexts debug
ael debug macros Enable AEL macros debug
ael debug read Enable AEL read debug
ael debug tokens Enable AEL tokens debug
ael no debug Disable AEL debug messages
ael reload Reload AEL configuration
agent logoff Sets an agent offline
agi debug Enable AGI debugging
agi no debug Disable AGI debugging
cdr mysql status Show connection status of cdr_mysql
cdr status Display the CDR status
database del Removes database key/value
database deltree Removes database keytree/values
database get Gets database value
database put Adds/updates database value
database show Shows database contents
database showkey Shows database contents
debug channel Enable debugging on a channel
debug level Set global debug level
dnsmgr reload Reloads the DNS manager configuration
dnsmgr status Display the DNS manager status
dont include Remove a specified include from context
dump agihtml Dumps a list of agi command in html format
dundi debug Enable DUNDi debugging
dundi flush Flush DUNDi cache
dundi lookup Lookup a number in DUNDi
dundi no debug Disable DUNDi debugging
dundi no store history Disable DUNDi historic records
dundi precache Precache a number in DUNDi
dundi query Query a DUNDi EID
dundi show entityid Display Global Entity ID
dundi show mappings Show DUNDi mappings
dundi show peers Show defined DUNDi peers
dundi show peer Show info on a specific DUNDi peer
dundi show precache Show DUNDi precache
dundi show requests Show DUNDi requests
dundi show trans Show active DUNDi transactions
dundi store history Enable DUNDi historic records
extensions reload Reload extensions and *only* extensions

feature show channels Show status of feature channels
group show channels Show active channels with group(s)
help Display help list, or specific help on a command
iax2 debug Enable IAX debugging
iax2 jb debug Enable IAX jitterbuffer debugging
iax2 no debug Disable IAX debugging
iax2 no jb debug Disable IAX jitterbuffer debugging
iax2 no trunk debug Disable IAX trunk debugging
iax2 provision Provision an IAX device
iax2 prune realtime Prune a cached realtime lookup
iax2 reload Reload IAX configuration
iax2 set jitter Sets IAX jitter buffer
iax2 show cache Display IAX cached dialplan
iax2 show channels Show active IAX channels
iax2 show firmware Show available IAX firmwares
iax2 show netstats Show active IAX channel netstats
iax2 show peer Show details on specific IAX peer
iax2 show peers Show defined IAX peers
iax2 show provisioning Show iax provisioning
iax2 show registry Show IAX registration status
iax2 show stats Display IAX statistics
iax2 show users Show defined IAX users
iax2 test losspct Set IAX2 incoming frame loss percentage
iax2 trunk debug Enable IAX trunk debugging
include context Include context in other context
indication add Add the given indication to the country
indication remove Remove the given indication from the country
init keys Initialize RSA key passcodes
load Load a dynamic module by name
local show channels Show status of local channels
logger reload Reopens the log files
logger rotate Rotates and reopens the log files
logger show channels List configured log channels
meetme Execute a command on a conference or conferee
mgcp audit endpoint Audit specified MGCP endpoint
mgcp debug Enable MGCP debugging
mgcp no debug Disable MGCP debugging
mgcp reload Reload MGCP configuration
mgcp show endpoints Show defined MGCP endpoints
mixmonitor Execute a MixMonitor command
moh classes show List MOH classes
moh files show List MOH file-based classes
moh reload Music On Hold
no debug channel Disable debugging on a channel
pri debug span Enables PRI debugging on a span
pri intense debug span Enables REALLY INTENSE PRI debugging
pri no debug span Disables PRI debugging on a span

pri set debug file Sends PRI debug output to the specified file
pri show debug Displays current PRI debug settings
pri show span Displays PRI Information
pri unset debug file Ends PRI debug output to file
realtime load Used to print out RealTime variables.
realtime mysql status Shows connection information for the MySQL RealTime driver
realtime update Used to update RealTime variables.
reload Reload configuration
remove extension Remove a specified extension
remove ignorepat Remove ignore pattern from context
remove queue member Removes a channel from a specified queue
restart gracefully Restart Asterisk gracefully
restart now Restart Asterisk immediately
restart when convenient Restart Asterisk at empty call volume
rtp debug Enable RTP debugging
rtp debug ip Enable RTP debugging on IP
rtp no debug Disable RTP debugging
set debug Set level of debug chattiness
set verbose Set level of verboseness
show agents Show status of agents
show agi Show AGI commands or specific help
show applications Shows registered dialplan applications
show application Describe a specific dialplan application
show audio codecs Shows audio codecs
show channel Display information on a specific channel
show channels Display information on channels
show channeltypes Show available channel types
show codecs Shows codecs
show codec Shows a specific codec
show conferences Show status of conferences
show config mappings Show Config mappings (file names to config engines)
show dialplan Show dialplan
show features Lists configured features
show file formats Displays file formats
show functions Shows registered dialplan functions
show function Describe a specific dialplan function
show hints Show dialplan hints
show image codecs Shows image codecs
show image formats Displays image formats
show indications Show a list of all country/indications
show keys Displays RSA key information
show manager command Show a manager interface command
show manager commands List manager interface commands
show manager connected Show connected manager interface users
show modules List modules and info

show modules like List modules and info
show parkedcalls Lists parked calls
show queue Show status of a specified queue
show queues Show status of queues
show switches Show alternative switches
show translation Display translation matrix
show uptime Show uptime information
show version Display version info
show version files Show versions of files used to build Asterisk
show video codecs Shows video codecs
show voicemail users List defined voicemail boxes
show voicemail zones List zone message formats
sip debug Enable SIP debugging
sip debug ip Enable SIP debugging on IP
sip debug peer Enable SIP debugging on Peername
sip history Enable SIP history
sip no debug Disable SIP debugging
sip no history Disable SIP history
sip notify Send a notify packet to a SIP peer
sip prune realtime Prune cached Realtime object(s)
sip prune realtime peer Prune cached Realtime peer(s)
sip prune realtime user Prune cached Realtime user(s)
sip reload Reload SIP configuration
sip show channels Show active SIP channels
sip show channel Show detailed SIP channel info
sip show domains List our local SIP domains.
sip show history Show SIP dialog history
sip show inuse List all inuse/limits
sip show objects Show all SIP object allocations
sip show peer Show details on specific SIP peer
sip show peers Show defined SIP peers
sip show registry Show SIP registration status
sip show settings Show SIP global settings
sip show subscriptions Show active SIP subscriptions
sip show users Show defined SIP users
sip show user Show details on specific SIP user
skinny debug Enable Skinny debugging
skinny no debug Disable Skinny debugging
skinny show devices Show defined Skinny devices
skinny show lines Show defined Skinny lines per device
soft hangup Request a hangup on a given channel
stop gracefully Gracefully shut down Asterisk
stop now Shut down Asterisk immediately
stop when convenient Shut down Asterisk at empty call volume
unload Unload a dynamic module by name
zap destroy channel Destroy a channel
zap show cadences List cadences

zap show channels Show active zapata channels
zap show channel Show information on a channel
zap show status Show all Zaptel cards status

Recommended:

1) Have two trunks at least. A “DEF” trunk with the most expensive rates and another one with your normal sales prices.
2) DO not tick your normal trunk as the “default” one. Unless you have only one trunk.

Error Messages:

1) `ERROR', 'No Route-RateAddcdr.proc-10164'` happens because you do not have a route setup for this particular dialed number pattern.
2) `('ERROR', 'No Route-RateCost.proc-10153'` happens

System Administrators Manual

DRAFT

Version 3.0

For astbill version 0.9.0.16

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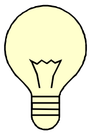
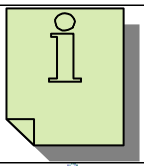
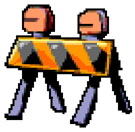
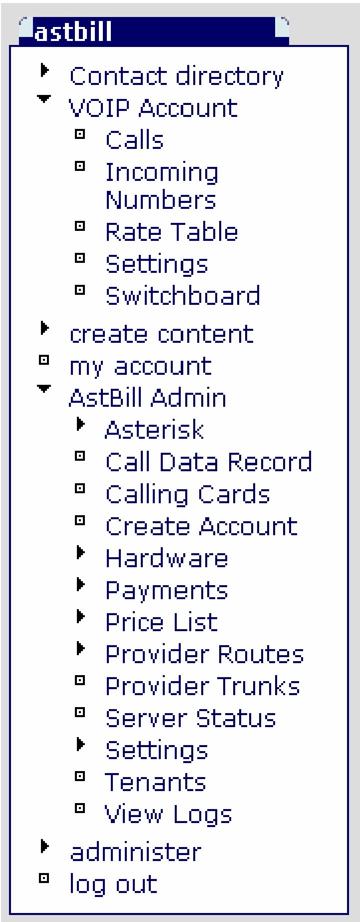
You will find the latest version of this document at
http://users.tpg.com.au/adslgw22/downloads/

Legend:

		Tip or Idea
		Information
		Caution: be careful of what your are doing
		Work in progress or under construction
		STOP: Don’t do it


User login
Username:
astbill
Password:
Log in
Request new password


astbill
Contact directory
VOIP Account
Calls
Incoming Numbers
Rate Table
Settings
Switchboard
create content
my account
AstBill Admin
Asterisk
Call Data Record
Calling Cards
Create Account
Hardware
Payments
Price List
Provider Routes
Provider Trunks
Server Status
Settings
Tenants
View Logs
administer
log out

- administer
  - access control
  - blocks
  - categories
  - content
  - input formats
  - logs
  - menus
  - modules
  - settings
  - themes
  - url aliases
  - users
  - help
- log out

Home » administer

access control

permissions | roles | account rules

In this area you will define the permissions for each user role (role names are defined on the user roles page). Each permission describes a fine-grained logical operation, such as being able to access the administration pages, or adding/modifying a user account. You could say a permission represents access granted to a user to perform a set of operations.

Permission	Admin	Adoroar	anonymous user	authenticated user
astadmin module				
astadmin accesslog	☐	☐	☐	☐
astadmin asterisk	☐	☐	☐	☐
astadmin payment	☐	☐	☐	☐
astadmin use Assign DID	☐	☐	☐	☐
can use AstBill Admin	☐	☐	☐	☐
astcallcard module				
can use callcard	☐	☐	☐	☐
astcontact module				
can use astcontact	☑	☑	☐	☑

Home » administer

menus

list | add menu | add menu item | reset menus

Select an operation from the list to move, change, or delete a menu item.

Navigation

Menu item	Expanded	Operations
(disabled)		enable
Contact directory	No	edit disable
- Add contact		edit disable
- Categories	No	edit disable
VOIP Account	Yes	edit disable reset
- Calls	No	edit disable
- Incoming Numbers		edit disable
- Rate Table		edit disable
- Settings		edit disable
- Switchboard		edit disable
compose tips (disabled)		enable
content (disabled)	No	enable
- create content (disabled)	No	enable

Home » administer

modules

Modules are plugins for Drupal that extend its core functionality. Here you can select which modules are enabled. Click on the name of the module in the navigation menu for their individual configuration pages. Once a module is enabled, new permissions might be made available. Modules can automatically be temporarily disabled to reduce server load when your site becomes extremely busy by enabling the throttle.module and checking throttle. The auto-throttle functionality must be enabled on the throttle configuration page after having enabled the throttle module.

Name	Description	Enabled
aggregator	Aggregates syndicated content (RSS and RDF feeds).	☐
archive	Displays a calendar for navigating older content.	☐
astadmin	AstBill Admin	☑
astcallcard	AstBill Callcard	☑
astcontact	AstBill Contact Directory	☑
astentry	AstBill Billing and Configuration	☑
astpbx	AstBill User Module	☑
astpricelist	AstBill Pricelist	☑
astroute	AstBill Routing	☑
astsettings	AstBill Settings	☑
asttenant	AstBill Tenants	☑
block	Controls the boxes that are displayed around the main content.	required
blog	Enables keeping an easily and regularly updated web page or a blog.	☐

Home » administer

settings

General configuration options for your site. Set up the name of the site, e-mail address used in mail-outs, clean URL options, caching, etc.

General settings

Name:

AstBill

The name of this web site.

E-mail address:

noreply@astbill.com

A valid e-mail address for this website, used by the auto-mailer during registration, new password requests, notifications, etc.

Slogan:

The slogan of this website. Some themes display a slogan when available.

Home » administer

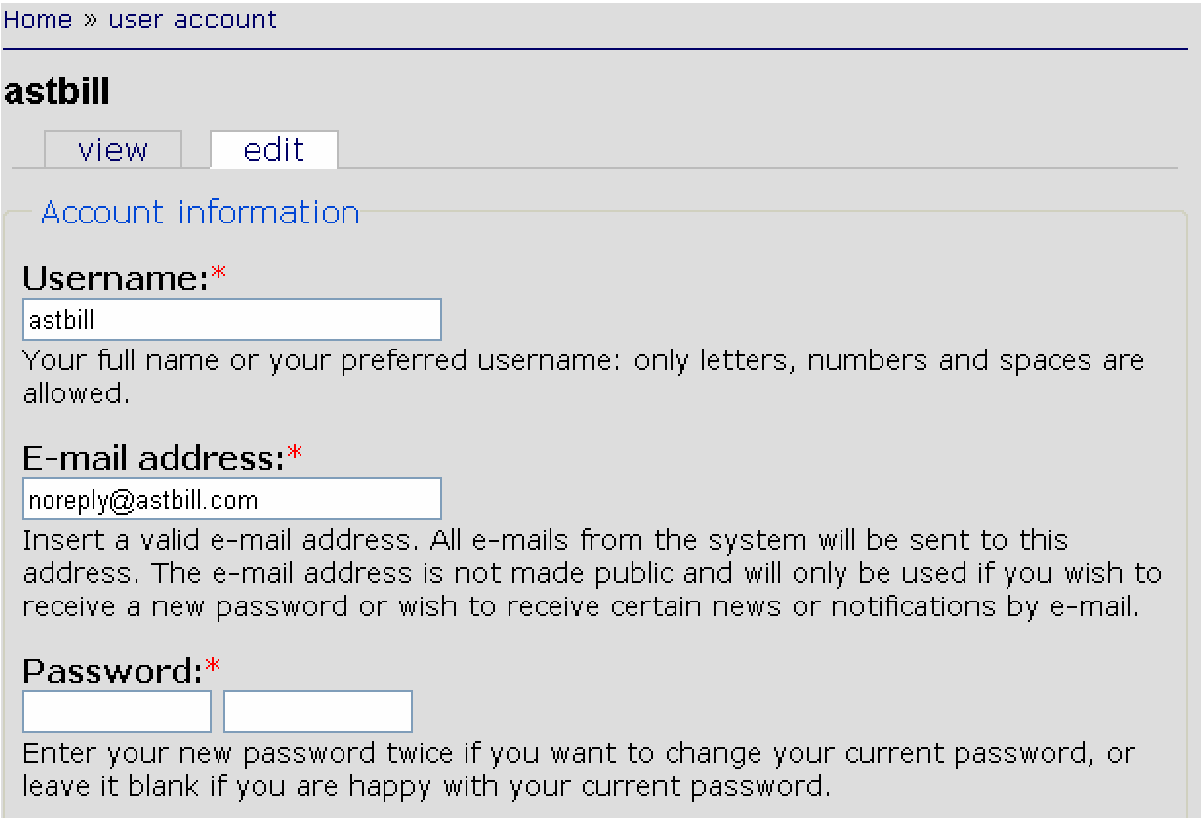
users

list | add user | configure

Drupal allows users to register, login, logout, maintain user profiles, etc. No participant can use his own name to post content until he signs up for a user account.

[more help...]

ID	Username	Status	Roles	Last access	Operations
1	astbill	active	authenticated user	2006-04-20 13:26	edit
2	demo	active	authenticated user	2005-11-06 15:52	edit
3	demoadmin	active	authenticated user, Admin	2005-09-24 09:17	edit
4	astbilltest	active	authenticated user, Admin, Adoroar	2005-09-23 17:31	edit



Home » user account

astbill

view | edit

Account information

Username: *

astbill

Your full name or your preferred username: only letters, numbers and spaces are allowed.

E-mail address: *

noreply@astbill.com

Insert a valid e-mail address. All e-mails from the system will be sent to this address. The e-mail address is not made public and will only be used if you wish to receive a new password or wish to receive certain news or notifications by e-mail.

Password: *

Enter your new password twice if you want to change your current password, or leave it blank if you are happy with your current password.

- AstBill Admin
  - Asterisk
  - Call Data Record
  - Calling Cards
  - Create Account
  - Hardware
  - Payments
  - Price List
  - Provider Routes
  - Provider Trunks
  - Server Status
  - Settings
  - Tenants
  - View Logs

- AstBill Admin
  - Asterisk
    - Iax Users
    - Incoming Friends
    - Manager
    - Show Peers
    - Sip Users

Home » AstBill Admin » Asterisk

Iax Users

View

Hover over Account to view Accounts Password

Accountcode	Callerid	Context	Mailbox	Type
02079999998		incomming-numbers		user
70104	Are <70200>	default	70100	friend
70107	Are <70200>	default	70108	friend
71303	Are	default	71303	friend
71415		default	71415	friend
71417	demo	default	71417	friend
71421	Are IPKall	incomming-numbers	71421	friend
71449	demoadmin	default	71449	friend

Home » AstBill Admin » Asterisk

Incoming Friends

View Incoming Numbers, Peers, Users and Friends

User / UID	Accountcode	Type	Publicnumber	Username	Operations		
0	voip_sip	peer		11111111	Edit	Assign	Remove
0	voiptalk-iax	peer			Edit	Assign	Remove
0	sipgate	friend	02079999999	8888888	Edit	Assign	Remove
0	IP24_Out	peer	4799999999	99999999	Edit	Assign	Remove
0	iaxfwd	user			Edit	Assign	Remove

Home » AstBill Admin » Asterisk

Manager

Manager

Click here to write configuration for accounts marked with STATIC:

This will overwrite the following files:

/etc/asterisk/sip_additional.conf
/etc/asterisk/iax_additional.conf
/etc/asterisk/voicemail_additional.conf

Write

Click here to Reload Asterisk:

Reload

Home » AstBill Admin » Asterisk

Show Peers

View

This is a list of the last clients(SIP and IAX2) that have connected to the Asterisk server.
It allows you to keep track of when each client last authenticated with Asterisk.
You also get the IP and configuration information.

Accountcode	Tech	Changed	Expires	Name	Callerid	IP
70103	SIP	2005-09-23 17:39:03	2005-09-10 21:50:02	astbill	Smith <70200>	72.229.231.211:5060
70104	IAX	2005-09-23 17:39:25	2005-09-10 21:39:21	demo	Are <70200>	202.214.169.142:4569
70105	SIP	2005-09-23 17:38:25	2005-09-06 19:56:40	demoadmin	Dulce <70105>	89.174.234.20:62719
70107	IAX	2005-09-23 17:38:27	2005-09-01 08:59:20	astbill	Are <70200>	89.174.234.20:61320

Home » AstBill Admin » Asterisk

Sip Users

View

Hover over Account to view Accounts Password

Accountcode	Callerid	Context	Mailbox	Nat	Type
voip_sip	0	ip24	0		peer
IP24_Out	0	ip24	0		peer
71459	71459	default	71459		friend
71434	71434	default	71434		friend
71433	71433	default	71433		friend
71425	71425	default	71425		friend
71423	71423	default	71423		friend
71422	71422	default	71422		friend
71382	london3	default	71382		friend
70111	Are <70200>	astar-debug	70100		friend
70108	Are <70200>	default	70100		friend

Home » AstBill Admin

Call Data Record

View

Search for Accountcode Go

Forward

User	ID	Accountcode	Callednum	Type	Trunk	Date Created	Dial Status	Billtime	Operations
demo	5666	70104	4790123	IAX2	IP24	2005-09-10 12:08:01	ANSWER	30	View
demo	5665	70108	4790123	SIP	IP24	2005-09-10 12:05:35	CANCEL	0	View
demo	5664	70104	70100	IAX2	Local	2005-09-09 21:35:36	ANSWER	22	View
astbill	5663	70103	447704606	SIP	DEFVOIP	2005-09-07 22:36:52	CANCEL	0	View
astbill	5662	70103	4790678	SIP	IP24	2005-09-07 22:32:00	ANSWER	270	View

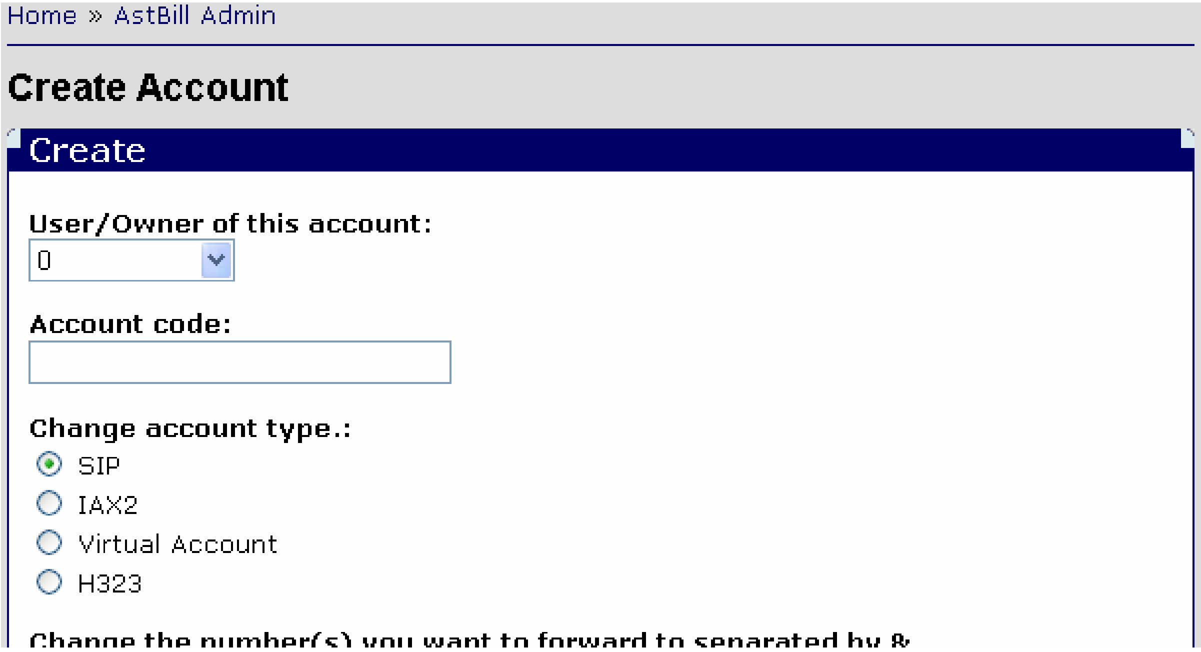
Home » AstBill Admin

Calling Cards

View

Search for Card Number Go

Card Number	Serialno	Firstused	Lastused	CardValue	CardBalance	A	Batchno	Operations
70100		0000-00-00	0000-00-00	0.00	0.00	1		Edit
70103		0000-00-00	0000-00-00	0.00	-5.46	1		Edit
70104		0000-00-00	0000-00-00	0.00	-3.21	1		Edit
70105		0000-00-00	0000-00-00	0.00	-19.02	1	dvcx	Edit
70106		0000-00-00	0000-00-00	0.00	-0.40	1		Edit
70107		0000-00-00	0000-00-00	0.00	-0.25	1		Edit



Home » AstBill Admin

Create Account

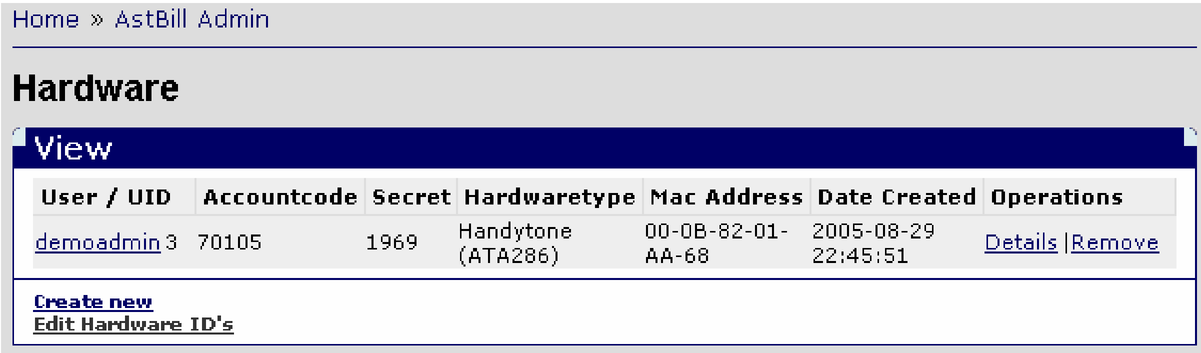
Create

User/Owner of this account:
0

Account code:

Change account type.:
- (•) SIP
- ( ) IAX2
- ( ) Virtual Account
- ( ) H323

Change the number(s) you want to forward to separated by &



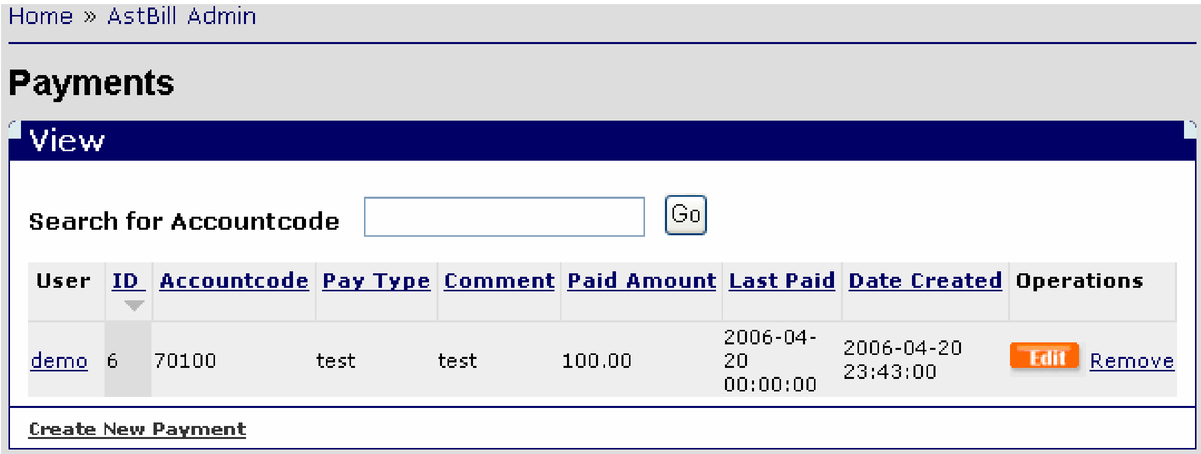
Home » AstBill Admin

Hardware

View

User / UID	Accountcode	Secret	Hardwaretype	Mac Address	Date Created	Operations
demoadmin 3	70105	1969	Handytone (ATA286)	00-0B-82-01-AA-68	2005-08-29 22:45:51	Details | Remove

Create new
Edit Hardware ID's



Home » AstBill Admin

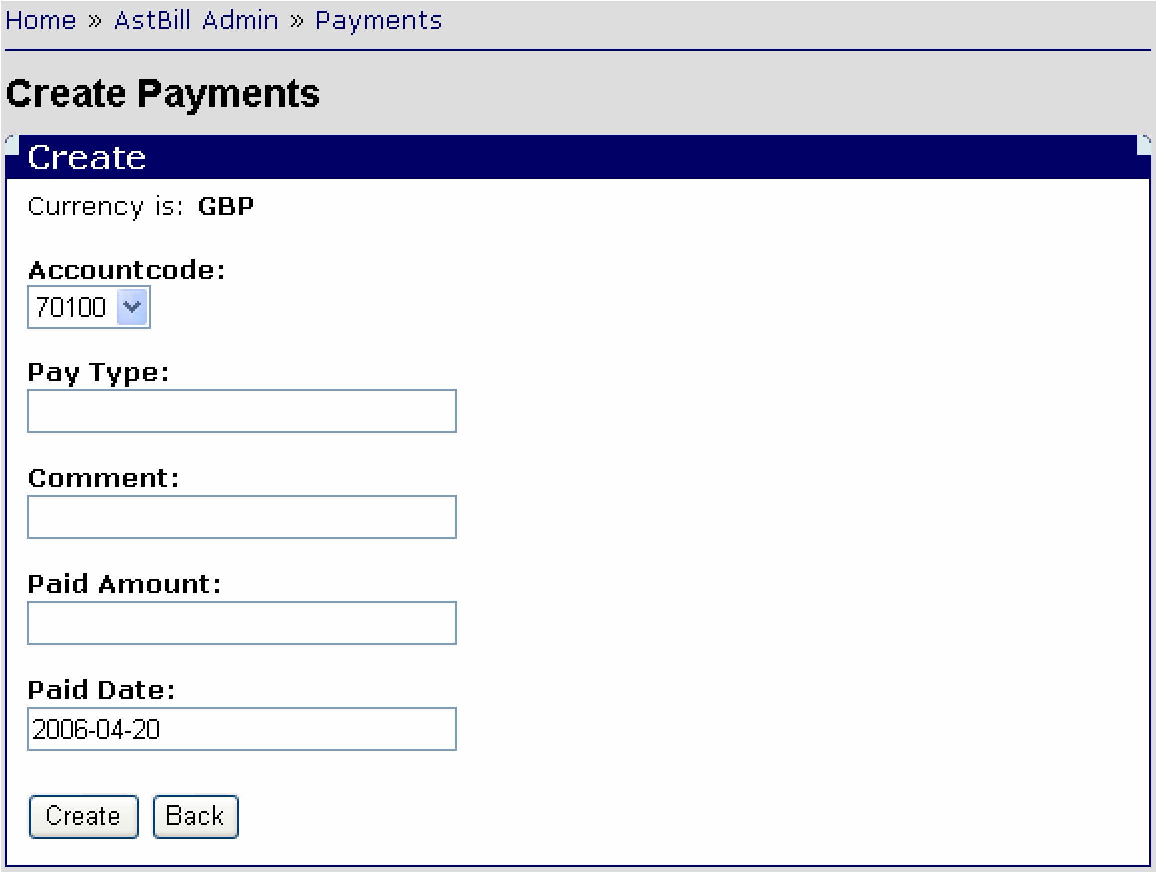
Payments

View

Search for Accountcode [ ] Go

User	ID	Accountcode	Pay Type	Comment	Paid Amount	Last Paid	Date Created	Operations
demo	6	70100	test	test	100.00	2006-04-20 00:00:00	2006-04-20 23:43:00	Edit Remove

Create New Payment



Home » AstBill Admin » Payments

Create Payments

Create

Currency is: **GBP**

Accountcode:
70100

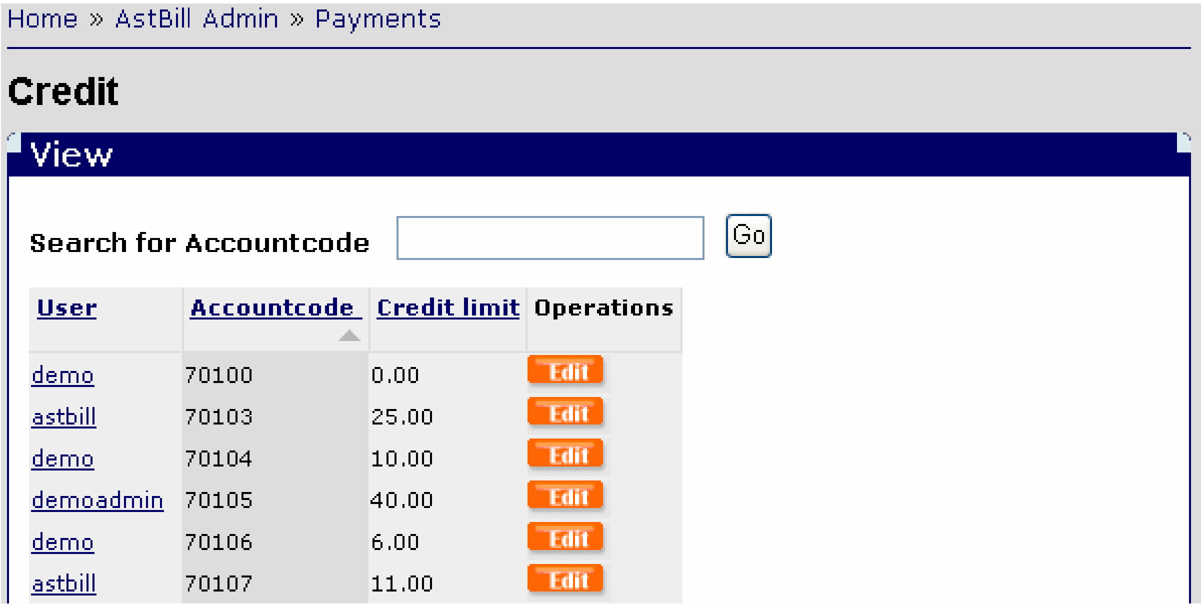
Pay Type:

Comment:

Paid Amount:

Paid Date:
2006-04-20

Create Back



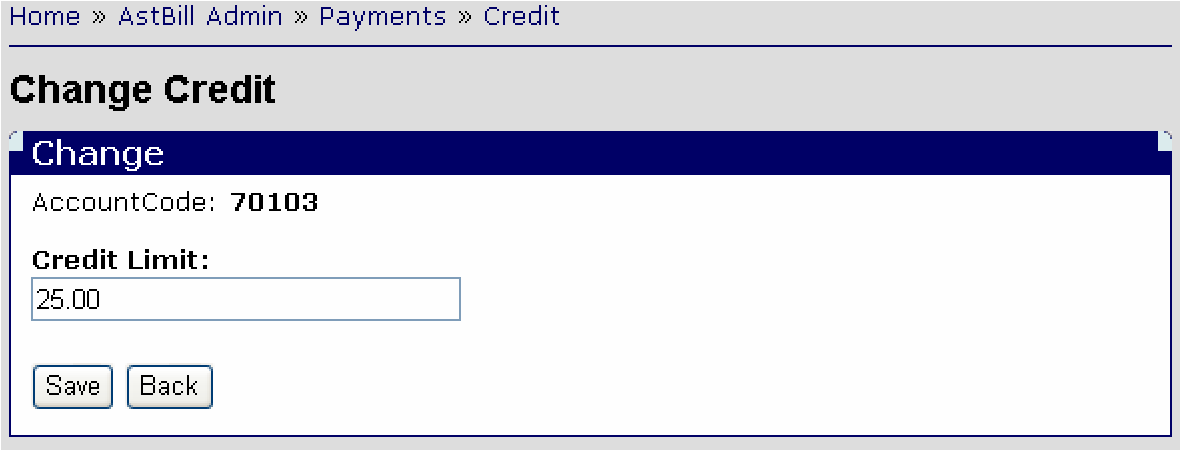
Home » AstBill Admin » Payments

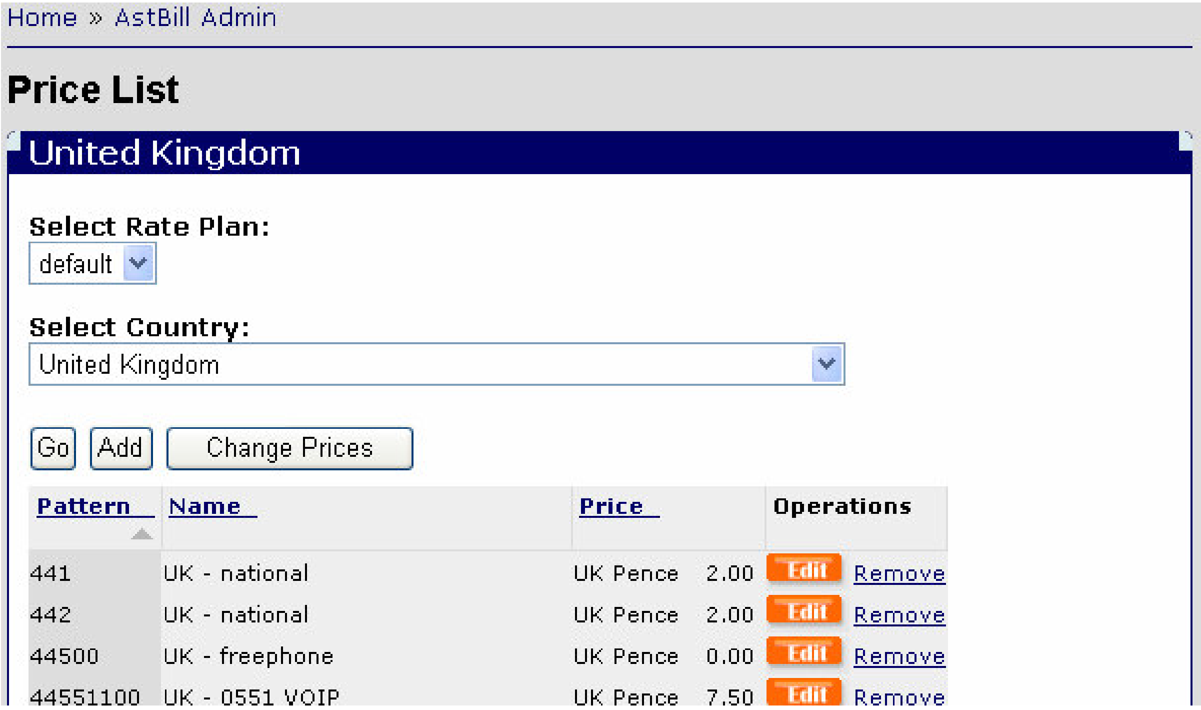
Credit

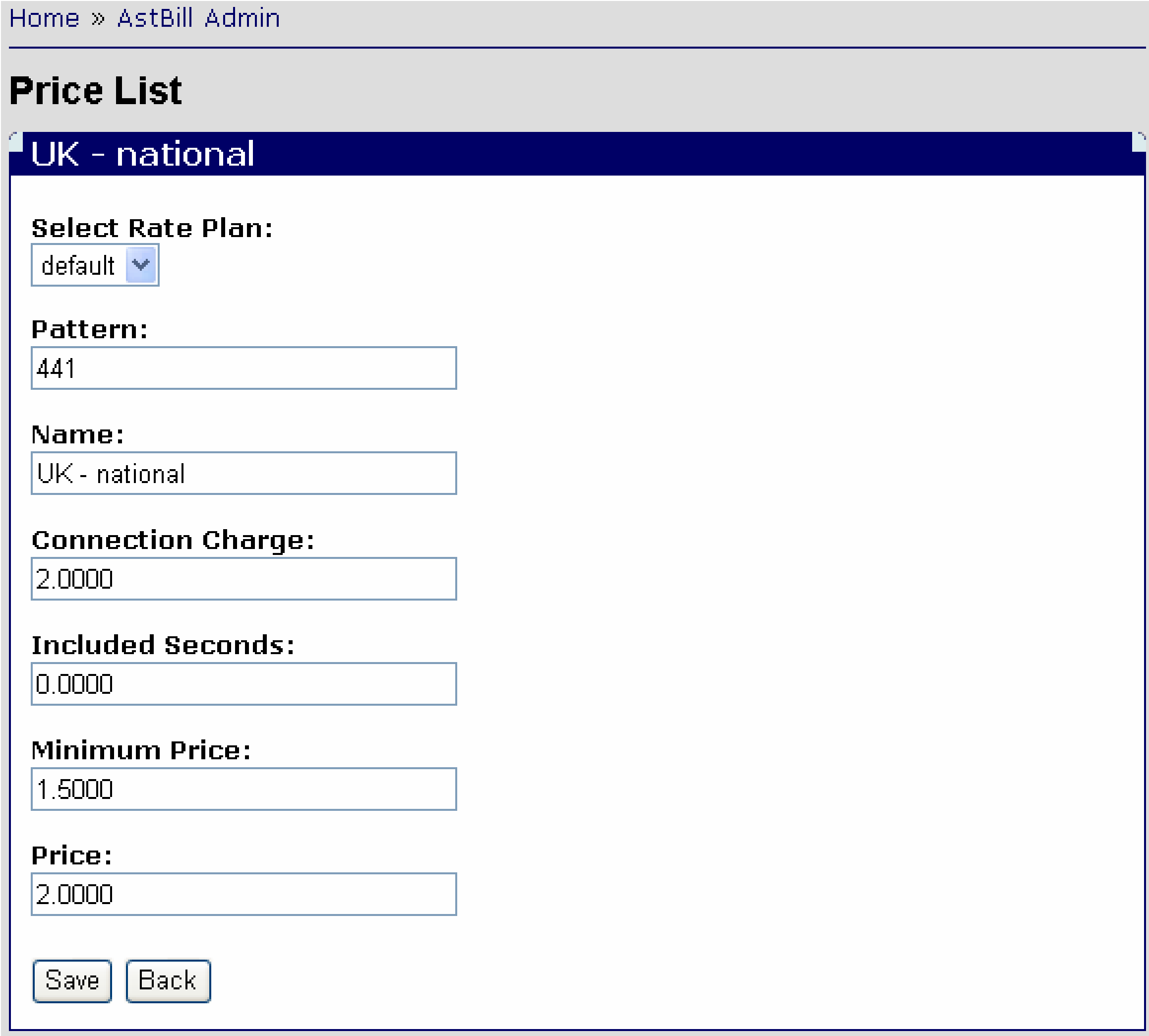
View

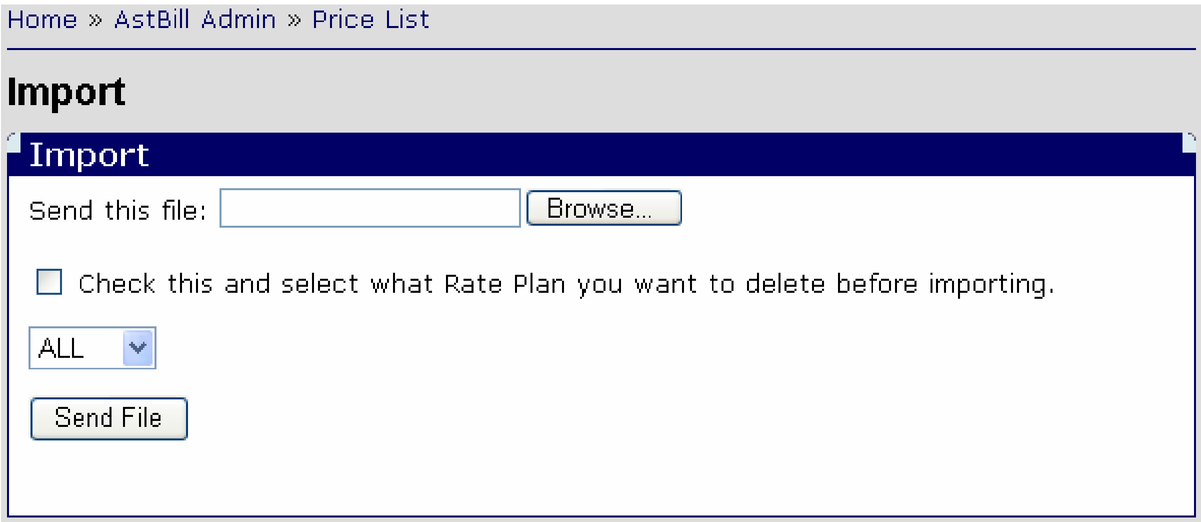
Search for Accountcode Go

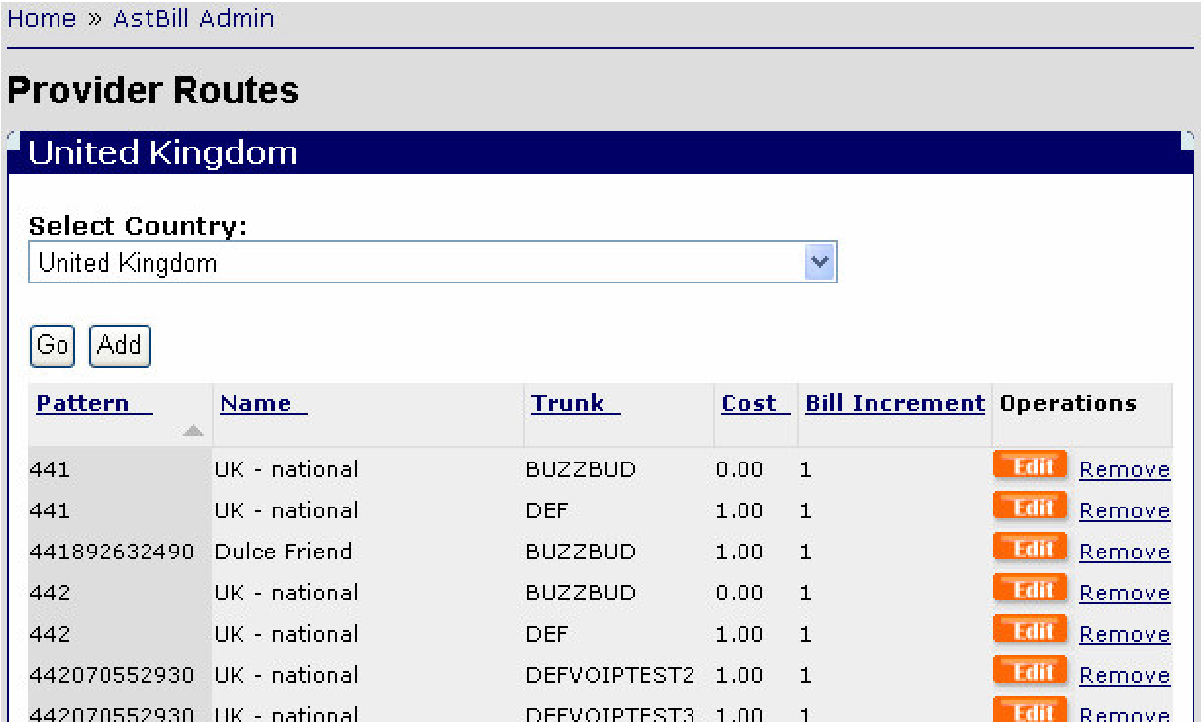
User	Accountcode	Credit limit	Operations
demo	70100	0.00	Edit
astbill	70103	25.00	Edit
demo	70104	10.00	Edit
demoadmin	70105	40.00	Edit
demo	70106	6.00	Edit
astbill	70107	11.00	Edit


Home » AstBill Admin » Payments » Credit
Change Credit
Change
AccountCode: 70103
Credit Limit:
25.00
Save
Back


Home » AstBill Admin
Price List
United Kingdom
Select Rate Plan:
default
Select Country:
United Kingdom
Go
Add
Change Prices
Pattern
Name
Price
Operations
441
UK - national
UK Pence 2.00
Edit
Remove
442
UK - national
UK Pence 2.00
Edit
Remove
44500
UK - freephone
UK Pence 0.00
Edit
Remove
44551100
UK - 0551 VOIP
UK Pence 7.50
Edit
Remove


Home » AstBill Admin
Price List
UK - national
Select Rate Plan:
default
Pattern:
441
Name:
UK - national
Connection Charge:
2.0000
Included Seconds:
0.0000
Minimum Price:
1.5000
Price:
2.0000
Save
Back


Home » AstBill Admin » Price List
Import
Import
Send this file:
Browse...
Check this and select what Rate Plan you want to delete before importing.
ALL
Send File



Home » AstBill Admin

Provider Routes

United Kingdom

Select Country:

United Kingdom

Go | Add

Pattern	Name	Trunk	Cost	Bill Increment	Operations
441	UK - national	BUZZBUD	0.00	1	Edit Remove
441	UK - national	DEF	1.00	1	Edit Remove
441892632490	Dulce Friend	BUZZBUD	1.00	1	Edit Remove
442	UK - national	BUZZBUD	0.00	1	Edit Remove
442	UK - national	DEF	1.00	1	Edit Remove
442070552930	UK - national	DEFVOIPTEST2	1.00	1	Edit Remove
442070552930	UK - national	DEFVOIPTEST3	1.00	1	Edit Remove

Home » AstBill Admin

Provider Routes

UK - national

Pattern: 441

Route:
UK - national

Select Trunk:
astartelecom

Cost Plan:
0

Connection Charge:
0.0000

Included Seconds:
0.0000

Billing Increment:
1

Minimum Cost:
0.0000

Cost:
0.0000

Save Back

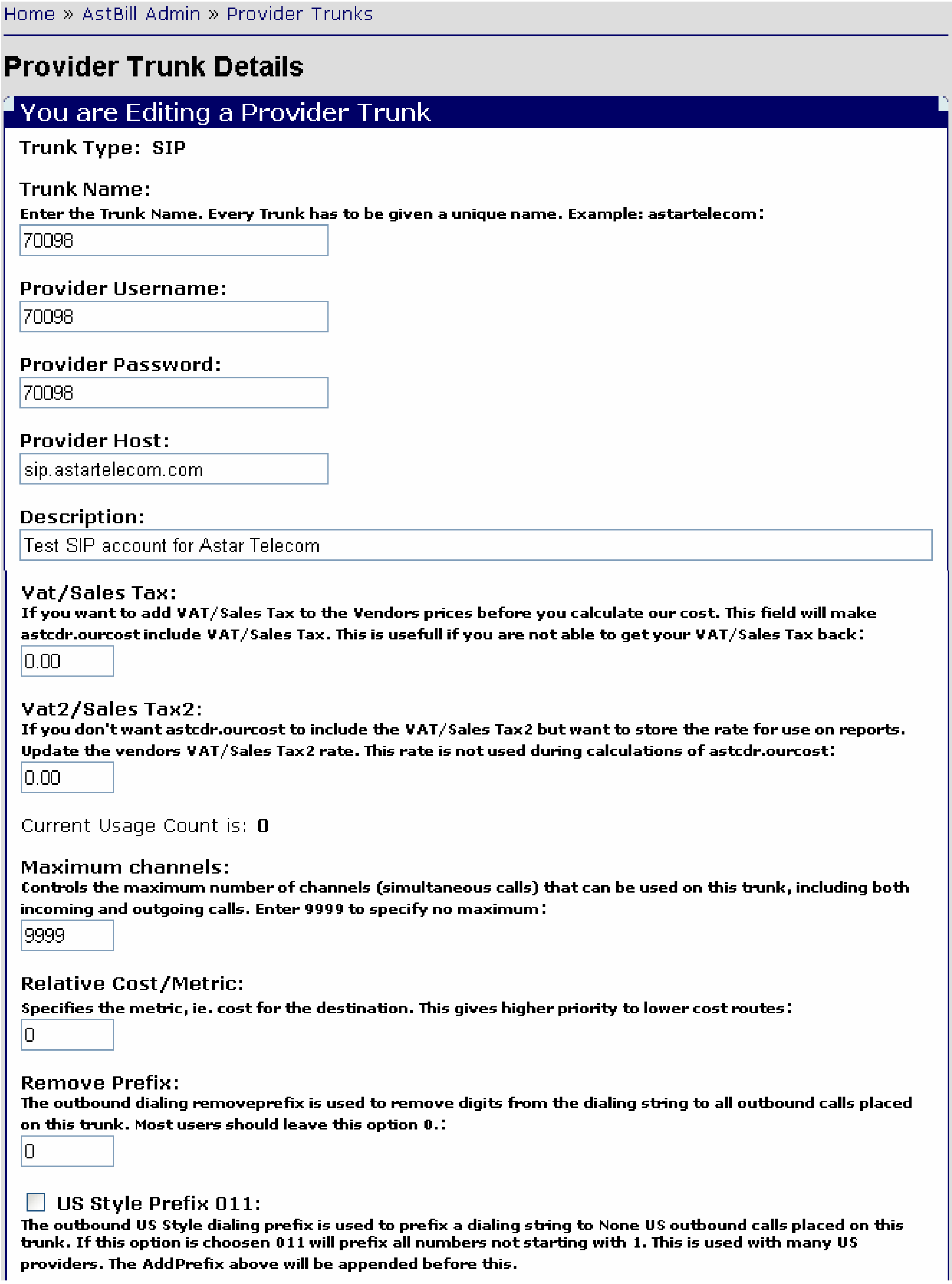
Home » AstBill Admin

Provider Trunks

View

Name	Path	Tech	Cost	Use	DB	Default	Dialplan	Operations
70098	@70098	SIP	0	0	R		Disabled	Edit Dialplan |Remove
astartelecom	@astartelecom	IAX2	0	0	R	*	Disabled	Edit Dialplan |Remove

Create New Provider Trunk
Create New STATIC Provider Trunk


Home » AstBill Admin » Provider Trunks
Provider Trunk Details
You are Editing a Provider Trunk
Trunk Type: SIP
Trunk Name:
Enter the Trunk Name. Every Trunk has to be given a unique name. Example: astartelecom:
70098
Provider Username:
70098
Provider Password:
70098
Provider Host:
sip.astartelecom.com
Description:
Test SIP account for Astar Telecom
Vat/Sales Tax:
If you want to add VAT/Sales Tax to the Vendors prices before you calculate our cost. This field will make astcdr.ourcost include VAT/Sales Tax. This is usefull if you are not able to get your VAT/Sales Tax back:
0.00
Vat2/Sales Tax2:
If you don't want astcdr.ourcost to include the VAT/Sales Tax2 but want to store the rate for use on reports. Update the vendors VAT/Sales Tax2 rate. This rate is not used during calculations of astcdr.ourcost:
0.00
Current Usage Count is: 0
Maximum channels:
Controls the maximum number of channels (simultaneous calls) that can be used on this trunk, including both incoming and outgoing calls. Enter 9999 to specify no maximum:
9999
Relative Cost/Metric:
Specifies the metric, ie. cost for the destination. This gives higher priority to lower cost routes:
0
Remove Prefix:
The outbound dialing removeprefix is used to remove digits from the dialing string to all outbound calls placed on this trunk. Most users should leave this option 0.:
0
US Style Prefix 011:
The outbound US Style dialing prefix is used to prefix a dialing string to None US outbound calls placed on this trunk. If this option is choosen 011 will prefix all numbers not starting with 1. This is used with many US providers. The AddPrefix above will be appended before this.

AddPrefix
The outbound dialing prefix is used to prefix a dialing string to all outbound calls placed on this trunk. For example, if this trunk is behind another PBX or is a Centrex line, then you would put 9 here to access an outbound line. Most users should leave this option blank.:

Register String:
Register String Many VoIP providers require your system to REGISTER with theirs. Enter the registration line here. example: username:password@sip.astartelecom.com:

☐ **Click to make default:**
Only one Provider Trunk can be used as default

Change Back

Advanced Provider Trunk Settings

- Settings
  - ANI/CLI
  - Batch Activation
  - Country List
  - Generate Account PIN's
  - Rate Plans
  - System Config
- Tenants
- View Logs

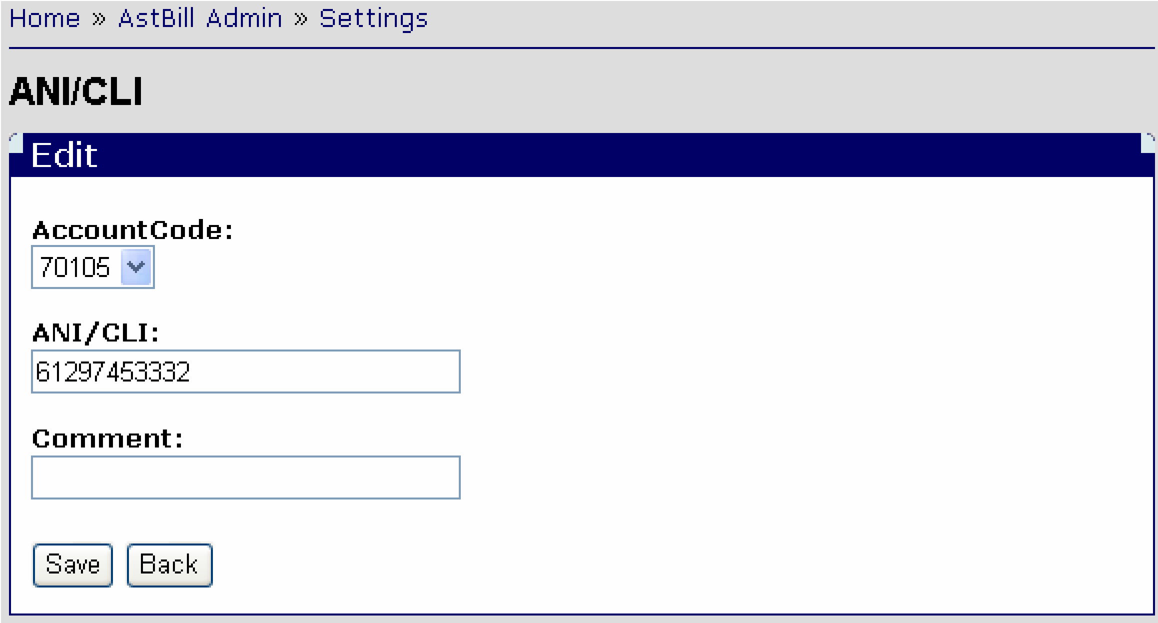
Home » AstBill Admin » Settings

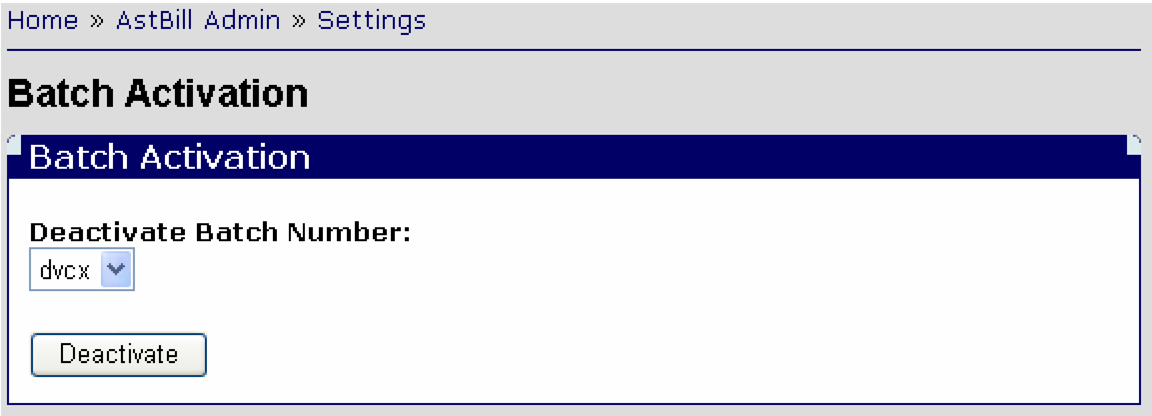
ANI/CLI

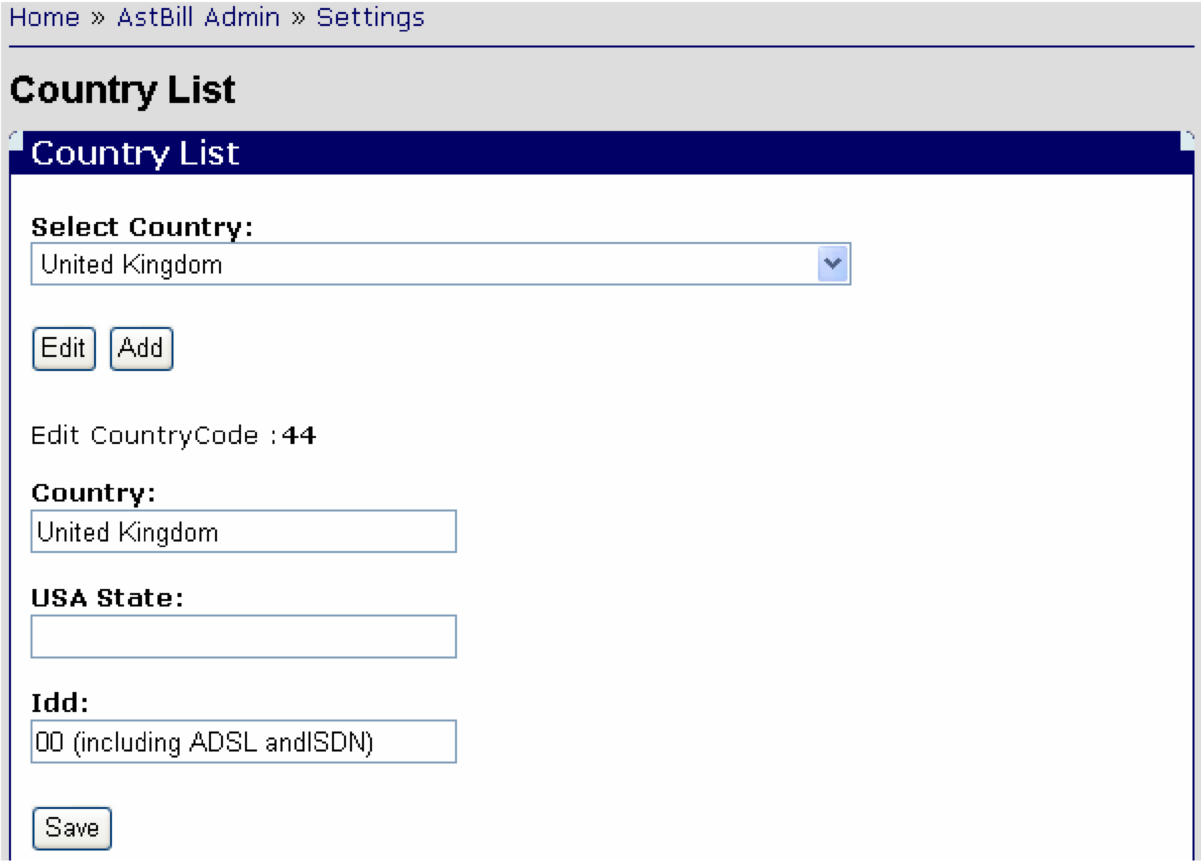
View

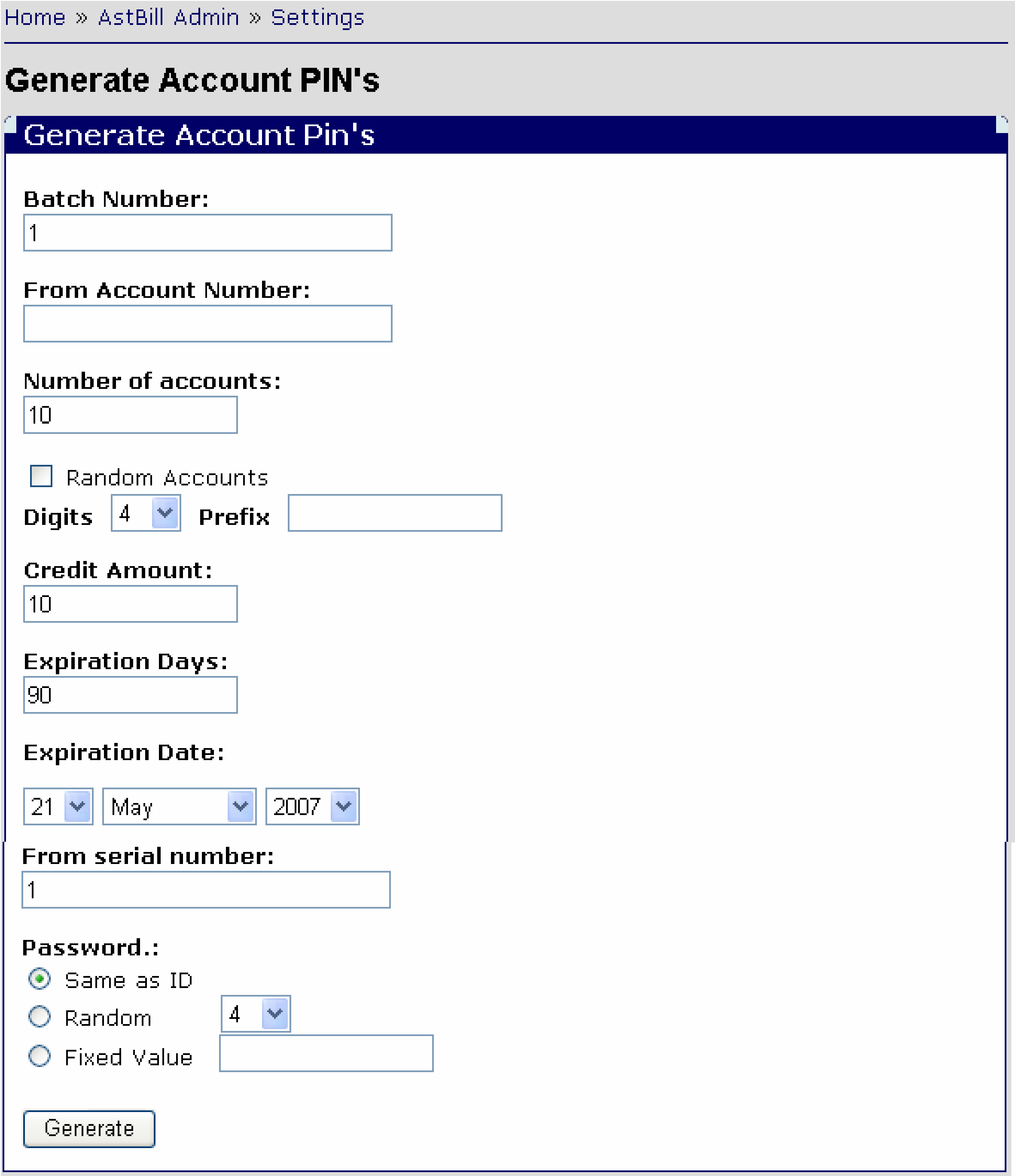
Accountcode	ANI/CLI	Comment	TimeStamp	Operations
70105	61297453332		2006-04-20 23:53:21	Edit Remove

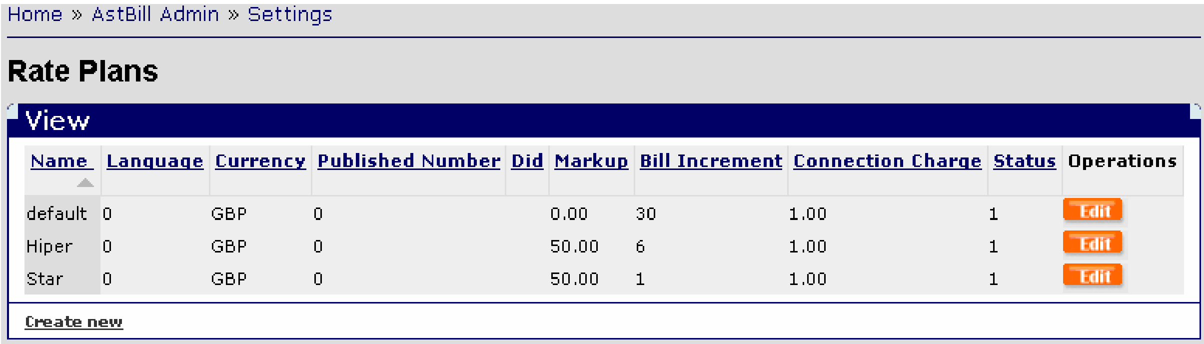
Create new

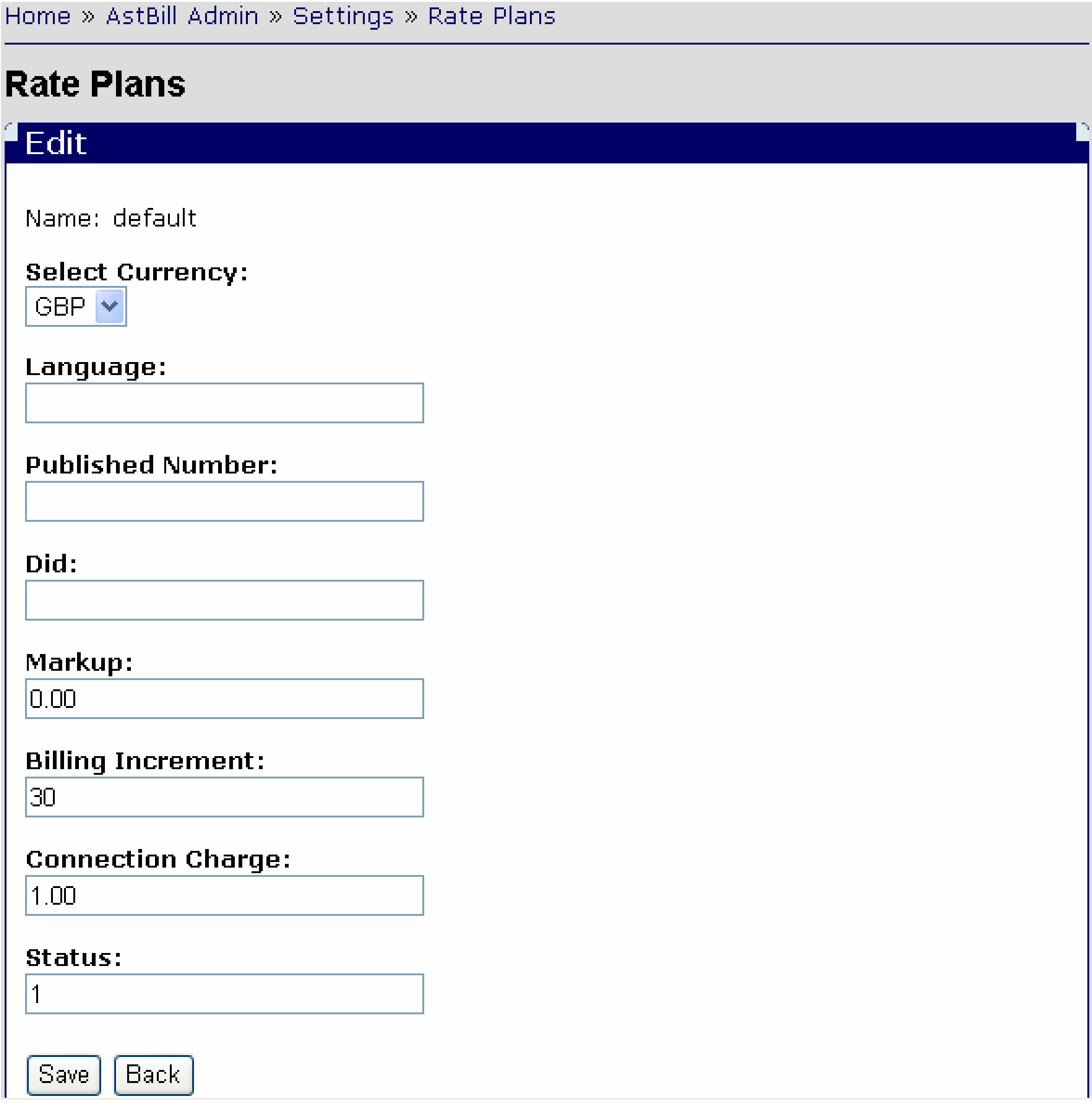

Home » AstBill Admin » Settings
ANI/CLI
Edit
AccountCode:
70105
ANI/CLI:
61297453332
Comment:
Save
Back


Home » AstBill Admin » Settings
Batch Activation
Batch Activation
Deactivate Batch Number:
dvcx
Deactivate


Home » AstBill Admin » Settings
Country List
Country List
Select Country:
United Kingdom
Edit
Add
Edit CountryCode : 44
Country:
United Kingdom
USA State:
Idd:
00 (including ADSL andISDN)
Save


Home » AstBill Admin » Settings
Generate Account PIN's
Generate Account Pin's
Batch Number:
1
From Account Number:
Number of accounts:
10
Random Accounts
Digits
4
Prefix
Credit Amount:
10
Expiration Days:
90
Expiration Date:
21
May
2007
From serial number:
1
Password.:
Same as ID
Random
4
Fixed Value
Generate


Home » AstBill Admin » Settings
Rate Plans
View
Name Language Currency Published Number Did Markup Bill Increment Connection Charge Status Operations
default 0 GBP 0 0.00 30 1.00 1 Edit
Hiper 0 GBP 0 50.00 6 1.00 1 Edit
Star 0 GBP 0 50.00 1 1.00 1 Edit
Create new


Home » AstBill Admin » Settings » Rate Plans
Rate Plans
Edit
Name: default
Select Currency:
GBP
Language:
Published Number:
Did:
Markup:
0.00
Billing Increment:
30
Connection Charge:
1.00
Status:
1
Save
Back

Home » AstBill Admin » Settings

System Config

Value	Comment
1	If 1 We will use a Drop Down box to show Accountcode in Payment screen and other Screens. If 0 a text entry box will be used.
5	This is the length of the automatically created account number
7	This is the first digit of the automatically created account number. Automaticly created account number must be numberic
6	The Password Length to use when creating new accounts
4	The Voicemail PIN/Password Length to use when creating new accounts
72000	If there is no account defined in the 7 range. This will be the first account created
/home/astbill/logs/	This is the Path to log files created by agi-bin scripts
2000	This is the numbers of chars visible from the Asterisk Messages Log File
60	This is the maximum alloved minutes for a call. If your call last longer than this you will be disconnected by the System.
/topup/	The URL used to topup AstBill VOIP Software

The default CountryPrefix to use when creating new accounts:

44

The default Rate Plan to use when creating new accounts:

default

The default Channel type to use when creating new accounts (SIP or IAX):

SIP

Save Settings

Home » AstBill Admin

Tenants

View

Users	TID	Company	Contact Name	Operations
5	30000	Default Tenant		Edit Remove

Create New Tenant