Voipswitch Manual

for version 340 and higher

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OVERVIEW **3**
SOFTSWITCH **4**
REQUIREMENTS. **10**
PROGRAM INSTALLATION. **10**
LAUNCHING THE MAIN APPLICATION VOIPSWITCH **12**
GATEWAYS **18**
GK/REGISTRAR **19**
PC2PHONE CLIENTS **28**
GK/REGISTRAR CLIENTS **28**
CALLBACK CLIENTS **29**
IVR CLIENTS **30**
COMMON CLIENTS OR SHARED CLIENTS **30**
DIALINGPLAN **31**
TARIFF TO DNIS **34**
CALLS AND FAILED CALLS **35**
STATISTICS **37**
EXEMPLARY SCENARIOS **39**
WHOLESALE TERMINATION 39
ACCEPTING CONNECTIONS FROM SOHO CLIENTS. 40
ASSIGNING TELEPHONE NUMBERS TO THE VOIPSWITCH'S CLIENTS 43

Overview

VoipSwitch is a platform that allows to implement various types of Voice Over the Internet Protocol (VOIP) services, with retaining shared, uniform management interface. The feature that distinguishes this platform is the implementation of an integrated, embedded billing system that cooperates with SQL – MS SQL or MySQL databases' servers.
This solution results in the simplicity of preparing the system, by the operator, to be fully functional for providing services and also for administrating it in the future.
VoipSwitch's software consists of the following modules:

- **VoipSwitch manager** – the main part of the application. It allows to monitor the whole incoming traffic. Apart from the current connection status and the type of logged in clients, it also presents a number of additional information on the processes that are taking place.

- **VoipSwitch Config** – management interface. It is a tool for configuring the whole system. It has a number of features that allow to manage clients' billing and analyze traffic information, based on the statistical data.

- **Web CDR –** the module that allows clients to check their current account status and the history of the calls made. There is a possibility of exporting data to the file from this level.

- **PC to Phone Client –** softphone based on the g723.1 codec.

- **Webphone –** softphone that can be initiated directly from the website.

- **Callback Client –** client's software that allows to initiate calls between two telephones.

- **Webcallback -** client's callback version that is initiated directly from the website.

- **SMS callback –** module that cooperates with the SMS operators. It also allows to create access number for SMS callback service using the mobile phone.

- **IVR module –** module responsible for playing back voice messages utilized by various services e.g. 2 stage dialing, account balance or IP PBX.

- **Online Shop –** module that allows to make payments using credit card or Paypal service.

- **Reseller's module –** web based system that allows resellers to create users accounts, manage them, manage the tariffs, and analyze the information on the traffic that is taking place etc.

Each of the module is mutually integrated what allows, for instance, Pc2Phone user to access, from this application, information on the history of calls made, price list or current account status. Also, from the same level, it is possible to add funds to the account or to utilize SMS Callback service.

On the other hand, division into the separates modules allows adjusting the implementation of given services to the desired time frame. For instance, Callback service, IVR, online payments or Reseller Module can be added at any time without disturbing the functionality of the system. Compatibility and simplicity of usage also refers to all the modules created by VoipSwitch company, what gives the provider the possibility of extending the range of services offered, or building on the services already utilized according to the existing clients' expectations and emerging of the new technologies.

Softswitch

Softswitch is the main element of the platform, which merges the functionality of the following VOIP architecture's elements.

H323 switch

H323 gatekeeper

SIP Proxy

SIP registrar

Each of the described elements can operate simultaneously with the others. Moreover, the clients, regardless of the protocol, or the way they transfer connections, can connect between one another. This option allows connecting the networks, which because of the differences in implemented protocols or dialects inside the particular protocol, cannot directly transfer connection between one another. Implementing VoipSwitch as a central traffic controller also introduces a number of additional management, supervision and network security facilitations.

The main characteristics of the softswitch include:

- Simultaneous and transparent support of SIP and H323 protocols (sip→h323 and h323→sip translator
- Possibility of implementing various types of proxy (e.g. RTP-proxy or signaling proxy), possibility of choosing proxy for each prefix defined in dialing plan.
- Advanced routing and rating system
- Full internetworking with most commercially available switches, softswitches, session border controllers and VOIP gateways.
- VOIP equipment support
- NAT support both for SIP and h323 equipment
- Calling to sip devices behind NAT (without the necessity of configuring NAT)
- Calling among users registered to VoipSwitch, support for dynamic IP addresses
- Authentication of VOIP equipment
    - Through IP address
    - Through ANI
    - Through h323id

    - Through the pair of login/password (according to the SIP standard)
- Flexible routing
- Individual, integrated billing system
- Managing pre-paid and post-paid accounts
- Setting up users in the VSConfig program
- Managing users, blocking, setting limits
- Generating the groups of users and managing lots
- Creating and managing tariffs, the possibility of attributing a tariff to an individual user
- Data stored in the MSSQL or MySQL database
- Graphic management interface (presentation of the statistical data, billing information, managing clients' accounts, generating PIN, managing the tariffs, dialing plan and others)
- Graphic interface presenting the current traffic in the real time, number of the logged in clients, with the division into different types of services, presentation of logs and others
- Web interface for clients – presentation of the connections history, possibility of exporting to the file, presentation of the current account status, possibility of making payments online and others
- Easy to set up architecture
- Automatic software re-start facilities in case of system failure
- Scalability for new telecommunication services by enabling additional modules

Advantages of managing the system

- Simplify the management processes and network configuration changes of VoIP equipment
- Unify equipment supporting different protocols (or dialects of one protocol)
- Manage concentration and routing processes of VoIP traffic
- Centralize authorization and billing tasks of VoIP calls in one point

- Hide the network structure from third parties, if necessary
- Utilize possibility of implementing value-added services such as: calling card system, IPPBX, callback system using additional software packages from VoipSwitch LLC

Standard applications

Central point of your VOIP network

Main benefits:

- Management of authorization rules of VoIP-gateways
- Setting up call routing rules
- Provisioning of compatibility for H323 and SIP- equipment of various vendors
- Security and load planning of VoIP-traffic by using optional RTP-proxying
- Access to the statistical data (ASR, PDD and others)
- Transparent interface of the billing system

Network security

When using RTP-proxying VoipSwitch provides a single entry point for VoIP traffic. Both for clients and carriers there is only one IP address available.

Integration of equipment with support of different protocols

One of the most important features of RSF1000 is its ability to support widely accepted signaling IP-protocols - SIP and H323. The system provides transparent converging of one

protocol into another, thus allowing performing calls from one type of equipment to another.

Scalability

Through launching subsequent modules, it is very convenient for a provider to extend the range of services offered. Available modules:

- IVR for calling cards
- Web/SMS/ANI callback (with IVR)
- Reseller's module
- Online shop

Specifications

Supported protocols	
1	H.323 v.2 (H.245 v7, H225 v4) with/without FAST START
2	SIP (RFC 3261)
3	proxying of RTP/RTCP streams
4	Signalling proxy
5	Support of T38 (SIP, H323)
6	Transparent conversion of SIP to H323 and vice versa

Support of the Devices Behind the NAT	
1	SIP-devices
2	H323-devices
Authentication	
1	by IP address – SIP and H323
2	by H323ID – h323 terminals/gateways
3	by ANI (calling party number) – SIP and H323
4	by login and password - SIP equipment
5	by login and password – HearLink pc to phone/web to phone dialer (included in the package)
6	gatekeeper registration based on aliases
Intelligent routing	
1	based on prefixes (the possibility of defining prefixes differentiating individual users)
2	based on accessibility of the VOIP gateway
3	based on priorities when choosing a gateway
4	depending on available voice codecs
5	depending on prefixes specified in the tariff of an individual client
Phone Numbers Translation	
1	Deletion of the set number of digits from the called party number
2	Addition of the set number of digits to the called party number
3	Deletion of the set number of digits from the caller number
4	Addition of the set number of digits to the caller number
5	Virtual prefixes (for differentiation of the dialing plans)
Information for the Billing System	
1	Real-time, built in billing system
2	Storage in SQL database (MSSQL or MYSQL)
3	pre-paid and post-paid accounts
4	Payments history

5	CDR – examining the logs of the calls carried out from the VSCConfig level, possibility of filtering data according to the set parameters, possibility of exporting data to the file (html, excel, txt, or csv type), presenting the CDR on the WWW pages available for clients
System Management and Control Features	
1	Graphic User Interface for managing the overall functionality of the system
2	Visual presentation of current connections along with the information on their status
3	The number of statistical data presenting the information on the traffic intensity with its various parameters e.g. ASR, PDD. Possibility of limiting the number of data presented by using available filters e.g. only incoming traffic from the particular client, traffic directed to the particular gateway, or prefix etc.
4	Visual presentation of logged in clients and their current status, with the division into types of services e.g. gatekeeper users, SIP users, pc2phone, callback.
Operating Systems	
1	Windows 2000, 2003, XP

Requirements.

Voipswitch system is compatible with Windows 2000, Windows XP and Windows 2003. SQL database is also required, either MS SQL or MYSQL version 4. MYODBC driver is recommended to connect MYSQL database with VoipSwitch application. It can be downloaded from the www.mysql.com website.
In addition it is recommended that MDAC version 2.8 or higher be downloaded (more information available on www.microsoft.com)

Program installation.

After extraction files from VoipSwitch.zip three catalogs will be created: “VoipSwitch”, “hearlink” and “callback”. In order to install VoipSwitch, it has to be saved to the folder that has the same name and is located on the server; and installation program has to be launched. Installation program will create VoipSwitch group in the “programs” menu. The next step is to create a connection with the database. In order to accomplish that VPSConfig program has to be activated. After running VPSConfig, “database settings” window will open and “create connection string” button has to be chosen. After clicking on “create connection string”

button, “data link properties” window will appear with the list of database drivers. Depending on the type of database that is used the following steps have to be completed:

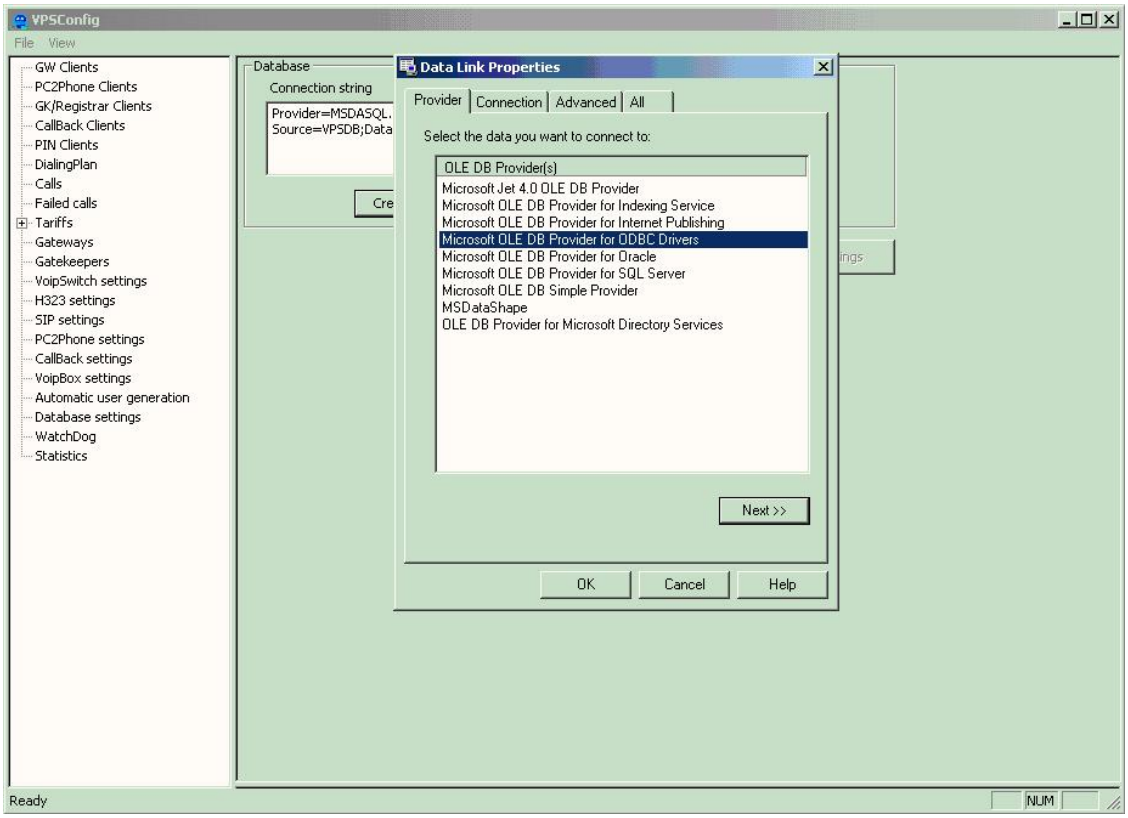
MSSQL:

1. From the list of drivers “Microsoft OLEDB provider for SQL servers” has to be chosen.
2. “Next” button has to be clicked on.
3. Name of the server or IP address has to be entered or chosen. Depending on the way user logs in to the SQL, button “Use Windows NT integrated security” or “Use specific user name and password” has to be chosen. In case of the subsequent all the necessary data have to be entered and “allow saving password” button chosen. “Test connection” button can be used to check if the connection succeeded.
4. If the connection succeeded, “OK” button has to be clicked on which will result in closing the “data link properties” window.
5. “Create database structure” button has to be chosen. From the unfolding options with the database types MSSQL has to be chosen and confirmed. After completing all the above steps the information confirming successful creation of VoipSwitch database should appear.

MySQL:

1. From the list of drivers “Microsoft OLEDB provider for ODBC drivers” has to be chosen.
2. “Next” button has to be clicked on.
3. “Use data source name” option has to be marked and the name representing the connection to MySQL has to be chosen. During the installation of MyODBC the connection named “myodbc3-test” is created by default. After choosing it database’s username and password have to be entered, unless they have been specified before. “Test connection” button can be used to check if the connection succeeded
4. If the connection succeeded, “OK” button has to be clicked on which will result in closing the “data link properties” window.
5. “Create database structure” button has to be chosen. From the unfolding options with the database types MySQL has to be chosen and confirmed. After completing all the above steps the information confirming successful creating of Voipswitch database should appear.

After the proper creation of the database structure and connecting with it, menu on the left side of VPSConfig window will become active.



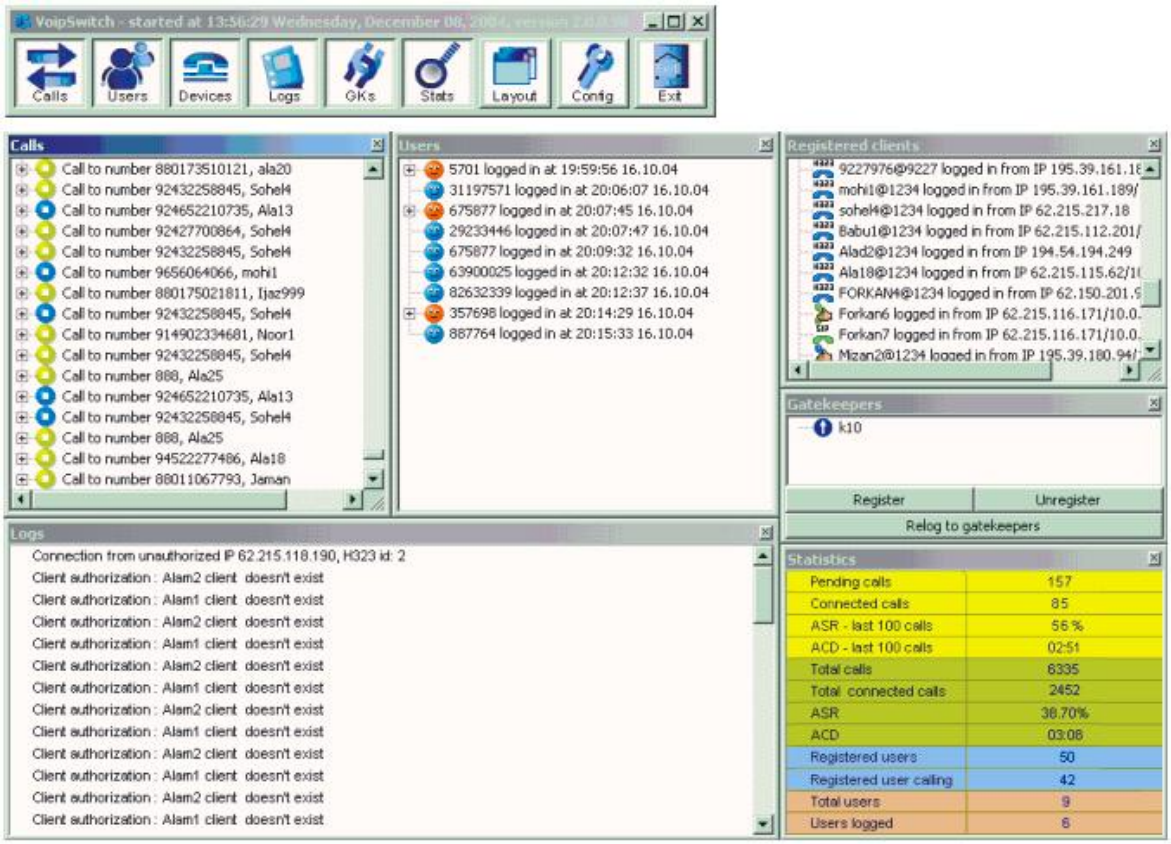
Picture 1 Connecting to database

Launching the main application VoipSwitch

After the initial set up of VoipSwitch parameters the main application "VoipSwitch manager" can be launched. It is installed in the menu programs/Voipswitch/Voipswitch.exe
After starting the application the information on the active services, previously defined in the VPSConfig application, should be displayed in any of the windows showing the applications' logs.

The main Voipswitch software has a window composed from a button bar and 6 sub-windows: Calls, Users, Registered Clients, Gatekeepers, Logs, and Statistics.

Each subwindow can be closed or opened from the buttons in the toolbar. The ***Layout*** button will rearrange all windows instantly. The ***Config*** button will open the VPSConfiguration manager.



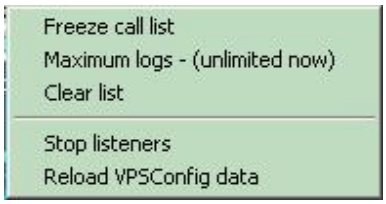
Picture 1 main program

The subwindow ***Calls*** shows the latest calls passed through the VoipSwitch. The calls connected will be shown in yellow, the calls not connected in blue and the calls that presented errors (like no money in account or no route to destination) in red.

Also you will find many details regarding call connection expanding each line. Like this you can easily debug call connections. You will find instantly connection details like the dialed number or the messages received from the remote gateway.

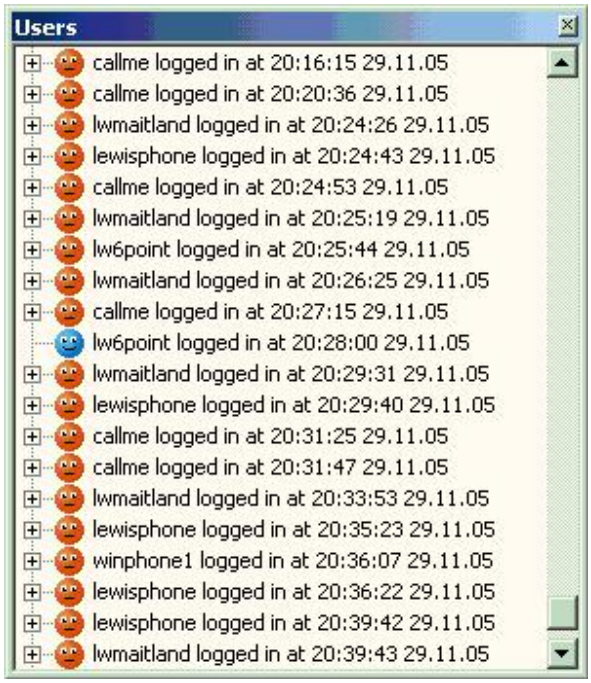
Picture 2 calls

If you right click on ***Calls window*** you will get a submenu. It is good to remember that here you can easily reload the configuration parameters without restarting VoipSwitch.



Picture 3 calls submenu

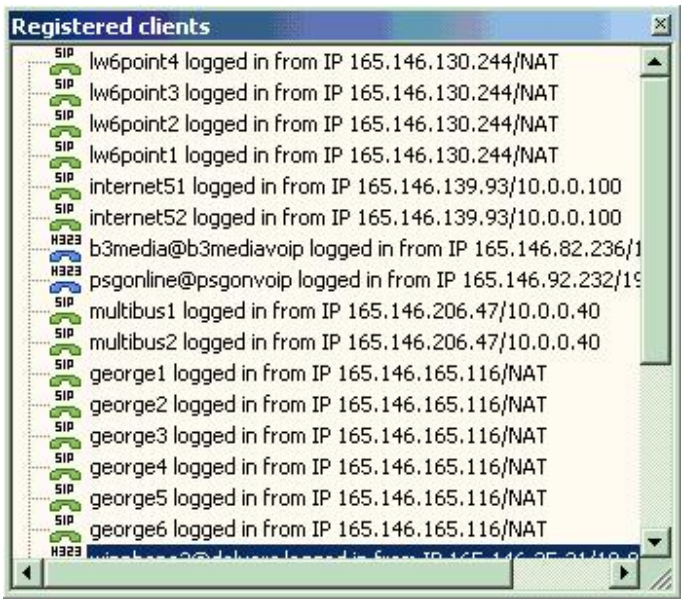
The subwindow ***Users*** shows the pc2phone (and web2phone) users and also the Callshop users. The current connected users will appear in blue and the disconnected users in red. Expanding each line you will see the calls made by each user.



Picture 4 Users pc2phone

The subwindow ***Registered Clients*** shows the users registered to the Voipswitch gatekeeper or registrar. Like h323 and SIP clients registered with user and password.

You can right click each registered user that appear in this window and see the current settings for each. Also you can reload the data from the database to show the current configuration in case something changed meantime.



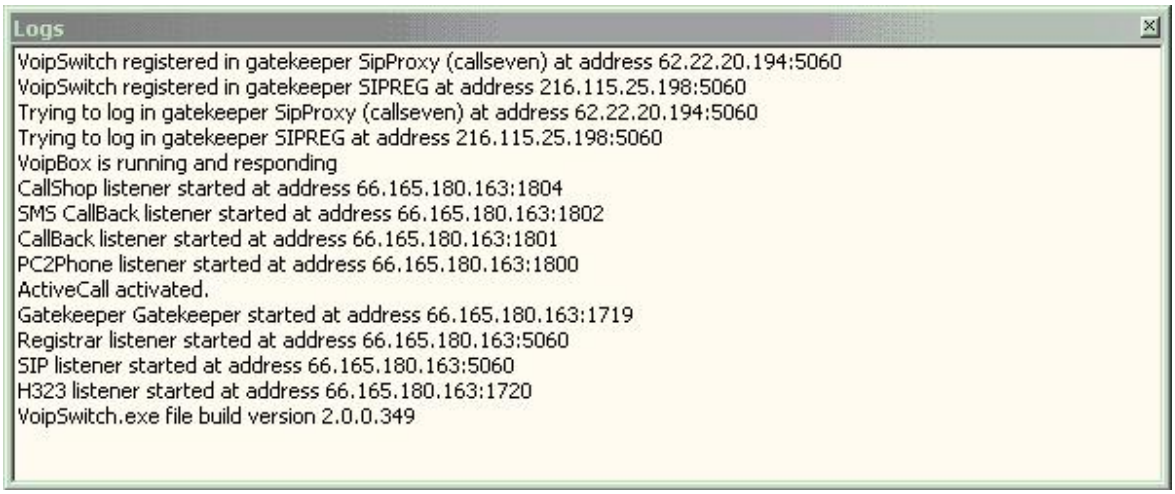
Picture 5 Registered clients

The subwindow ***Gatekeepers*** shows the gatekeepers to which Voipswitch is registered. After you create new gatekeeper settings or change existent settings in VPSConfig -> GK/Registrar you will have to click here the "Relog to gatekeepers" button to make Voipswitch to attempt (or reattempt) to connect to all gatekeepers.

Picture 6 Gatekeepers

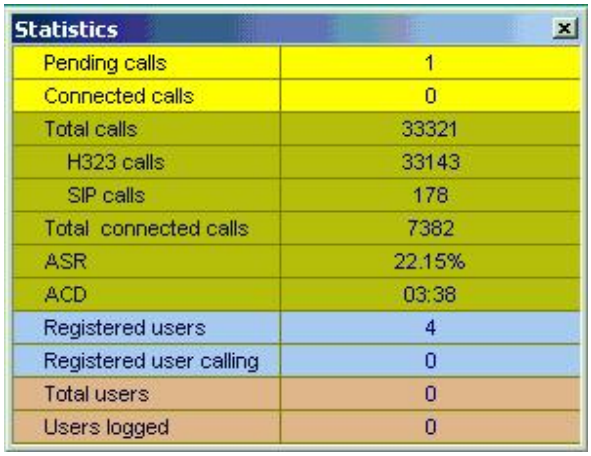
The subwindow ***Logs*** shows all the major events and alarms in Voipswitch.

This window is a good help for debugging the users that try to connect to Voipswitch. For example if Voipswitch rejects a user you will see in the ***Logs*** window what was received from the user and therefore you can deduct why the user was rejected (wrong user name or any other reason).



Picture 7 Logs

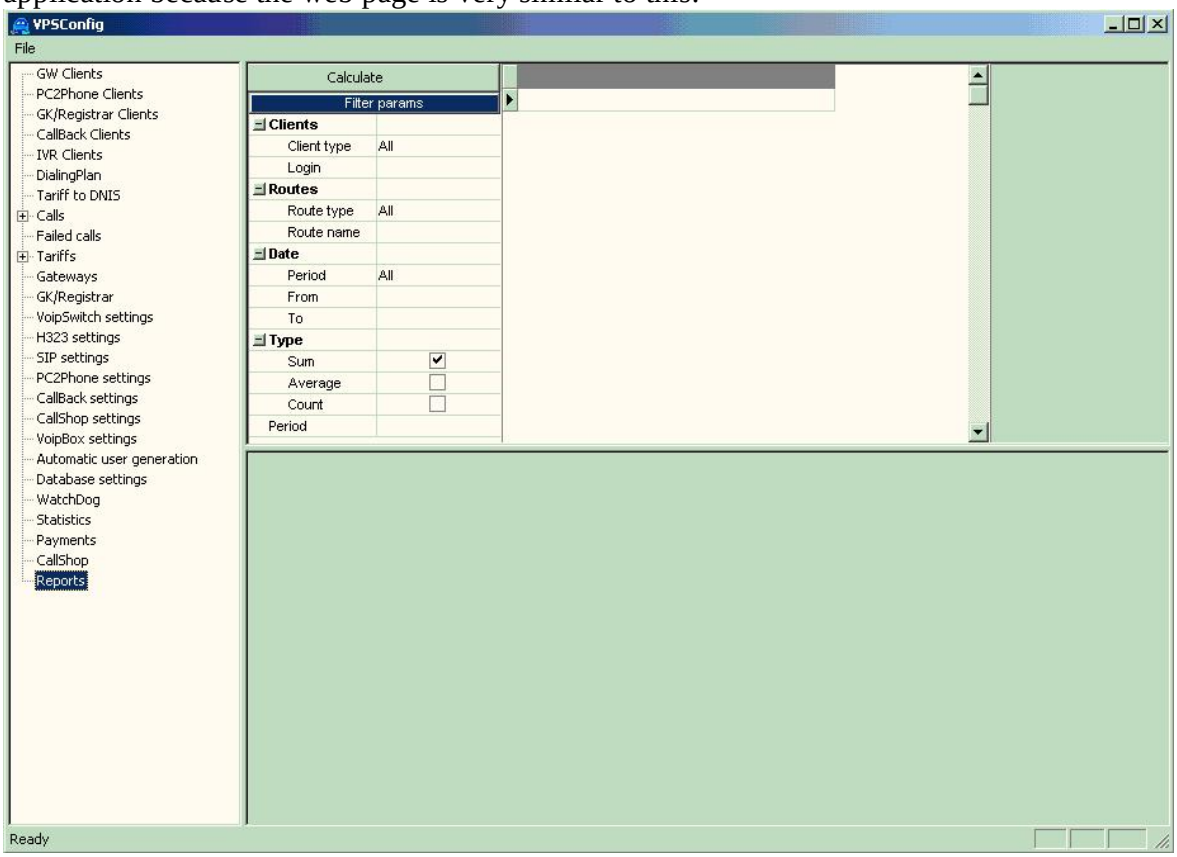
The ***Statistics*** subwindow shows a brief info about the general functioning of the system. Will show statistics about all calls passed through the system. You can get more detailed information about calls statistics from VPSConfig. We will discuss this later in this manual.



Picture 8 Statistics

The Voipswitch Configuration Manager

All the settings for Voipswitch to function are made in a separate window named ***VPSConfig*** or on the admin web page. We will discuss here only about the VPSConfig application because the web page is very similar to this.



Picture 9 VPSconfig

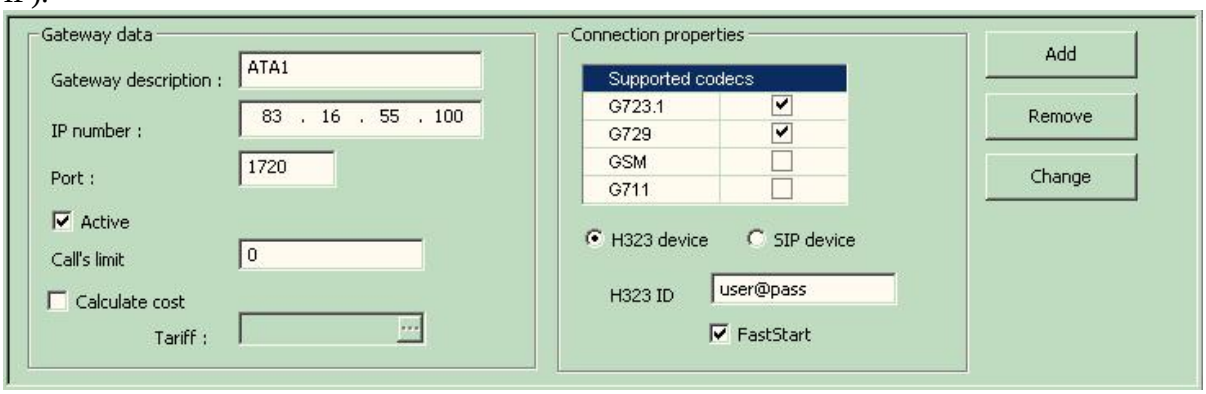
This window is organized with left side as an index for all settings while the right side will show the details.

First rule you have to remember is the main call flow in Voipswitch is from "Clients" towards "Gateways" or "GK/Registrar". So "Clients" will send calls to Voipswitch while "Gateways" and "GK/Registrars" will terminate them.

First thing to do in VPSConfig is to create the termination accounts. If you will have to send calls to termination gw in direct mode then create an account in ***Gateways***. If the remote device is a gatekeeper and you will need to register first to it then create an account in ***GK/Registrar***.

Gateways

In this section you have to define the termination gateways where you will send the calls. To these gateways defined here the Voipswitch will send the calls in direct mode (IP to IP).



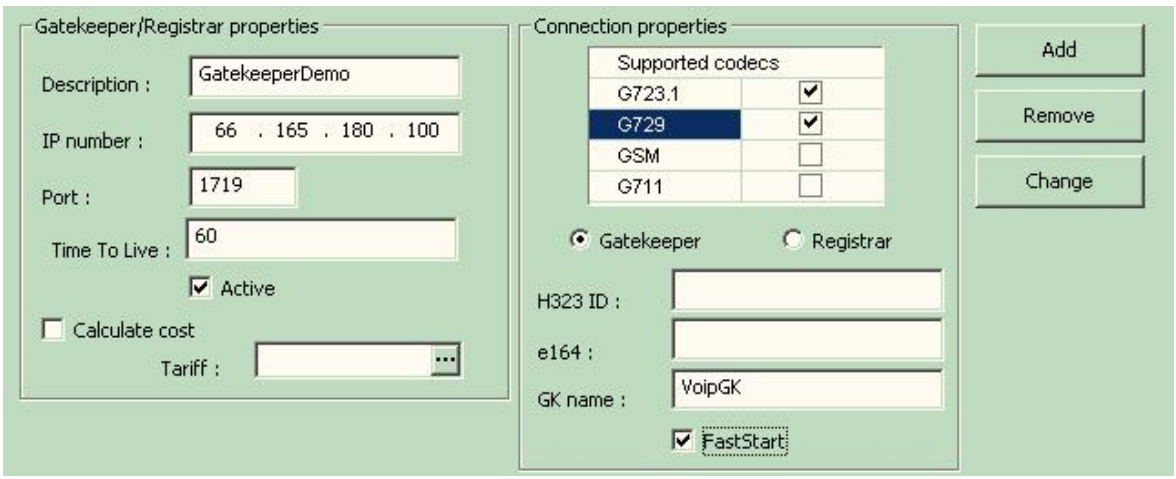
Gateway description is a label for the terminating gateway.
IP number is the IP address of remote terminating gw.
Port on remote gateway where to send the calls. Standard port for h323 protocol is 1720 and for SIP 5060. You have to change the port manually when you change the protocol.
Active sets the gateway active or inactive.
Call's limit sets a limit of maximum simultaneous calls that Voipswitch is allowed to send to this terminating gateway. Zero means unlimited calls.
Calculate cost and ***Tariff*** fields are not used yet. This function is under construction.
Supported codecs – select here only the codecs accepted by the remote gateway.
H323 device – SIP device to select the protocol that Voipswitch will use when sending calls to this gateway.
H323ID and ***FastStart*** are options that can be set when you select H323 protocol. ***H323ID*** can be required by your termination carrier to be sent for authentication. If not required is safe to be left blank. ***FastStart*** is a specific h323 protocol feature that enables faster call connection and advanced in-call options like call on hold and forwarding. You have to ask your carrier if his terminating gateway accepts this feature.

When you select SIP protocol you will be presented with ***Username*** and ***Password*** fields. Set them according to the terminating carrier requests or left them bank.

GK/Registrar

In this section you will define the terminating Gatekeepers or Registrars.

Voipswitch will use these details to register first to the remote Gatekeepers or Registrars. After you create the GK/Registrar account you can go to the main Voipswitch window and click the button ***Relog to gatekeepers*** from Gatekeepers sub-window to force Voipswitch to try to register immediately.



Description field is a label for the termination account.
IP number sets the remote GK or Registrar IP address.
Port where to send the registration request (usually 1719 for h323 Gatekeepers and 5060 for SIP Registrars).
Time To Live in seconds. It sets the amount of time until Voipswitch will check again if the remote GK or Registrar still accepts calls. Is good to se this value smaller or equal than the value set on remote side.
Supported codecs accepted by remote side.
Gatekeeper (h323):

H323 ID, e164, GK name, FastStart – consult your carrier about these settings. If not required leave them blank. But you should set at least GK Name and FastStart.

Registrar (SIP):

User name, Password, Domain user, Domain – consult your carrier about the values in these fields.

Now let's talk about the "Clients". They are divided in few types depending of the authentication method and purpose.

So you will find:

- GWclients for users authorized by IP or H323ID
- GK/RegistrarClients for users that will register with a user/pass to Voipswitch Gatekeeper or Registrar
- Callback clients for callback purposes
- IVRclients for users who will be asked for PIN to authorize

All client accounts have similar settings, so most of the features will be discussed only at the first client type.

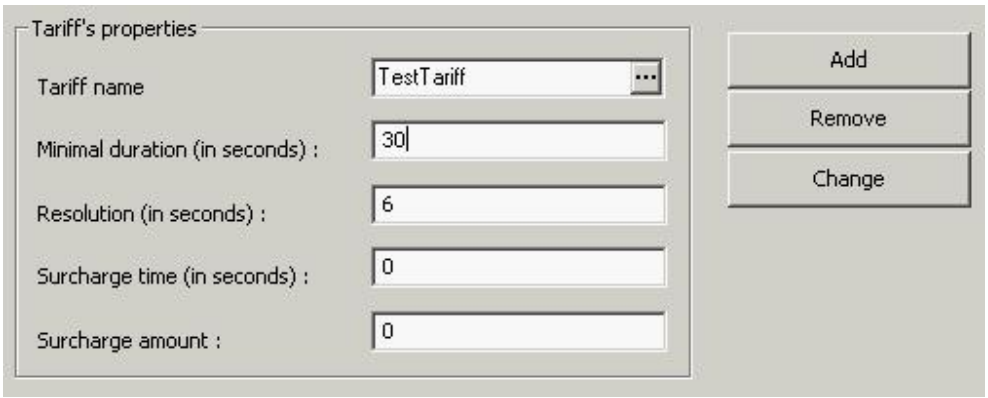
Each client has to have one and only one Tariff. So before you create a new Client account it is good to have the tariff prepared.

Therefore I think it will be good if you learn how to set the Tariffs from beginning.

But before we start you should note that anywhere in VPSConfig if you want to create a new record you should start by selecting an existent one. Then change all settings as you wish for the new record and click ***Add*** button from right side.

Tariffs

Clicking on ***Tariffs*** in VPSConfig will display the following properties window:



Here you can modify the general settings for an existent tariff, remove it or create a new one.

Minimal duration field denotes the minimum time that each call will be charged. The value of this field is in seconds.

Resolution field sets the billing interval. The value is also in seconds.

For example if you set Minimal duration 30 and Resolution 6 each call will be charged for a minimum of 30 seconds even if it lasted 2 seconds. All calls longer than 30

seconds will be charged in 6 seconds intervals according to the rates set for each destination.

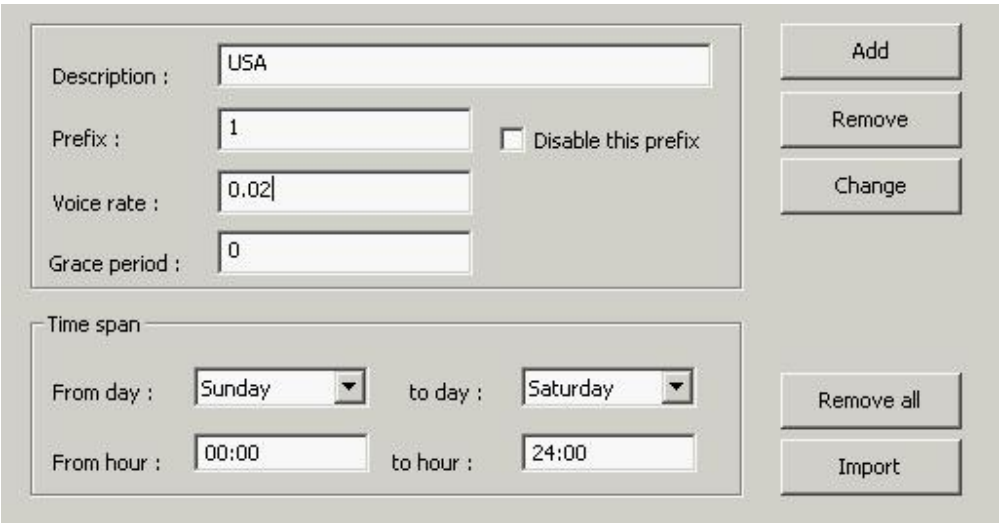
Surcharge time and ***Surcharge amount*** will set extra billing for the beginning of each call.

During the time you set in ***Surcharge time*** Voipswitch does not bill the call according to normal billing. It charges only the amount set in ***Surcharge amount***. After the ***Surcharge time*** expires the billing will start as if there was the beginning of the call.

For example if the ***Surcharge time*** field is 10 seconds and ***Surcharge amount*** is 0.1 then first 10 seconds of each call will be charged with 0.1 and only then the normal billing will start.

Also if you want to charge all calls with same value here you can set ***Surcharge time*** zero and ***Surcharge amount*** the value you want. In this way each call will be charged with the fixed value (surcharge amount) regardless of its duration and the values for Minimal duration and Resolution.

After you added a tariff you can expand the Tariff tree and the name of the new tariff will appear as a leaf. If you click on it, a table with rates will be presented on the right side. Here you can enter the rates one by one or import them from a file.



To add a new rate you will need to fill the fields and click **Add**.

Description field is where you can enter a brief description of the destination, usually the country name.

Prefix is usually the country code.

Voice rate is the price per minute. You can enter it in any currency but you have to keep the consistency having all rates in a tariff on same currency.

Grace period is a period of time, in seconds, while the billing does not begin. Within this grace period the caller has the possibility to hang up without being charged. After this period expires the call will be billed entirely since it was started.

This is useful, for example, when you have a destination where the quality is not always good and you want to give a grace period while the customer can hang up without being charged if is not satisfied with the quality.

Time span can define the intervals of week days and hours while the current rate is active. So for example you can charge differently same destination during the weekday or weekend or during peak hours and off-peak hours.

To cover all possible hours leave the default settings: From Sunday 00:00 to Saturday 24:00.

Disable this prefix option is used when you don't want to allow calls to a particular destination. For example you can limit the access to premium numbers like 1900.

Tariff example: If we will consider two rates:

Description: Poland	Prefix: 48	Voice rate: 0.05
Description: Poland cell	Prefix: 4860	Voice rate: 0.20

this will result in charging the calls starting with 4860 with the rate for "Poland cell" in our case 0.20.
All other calls starting with "48", for example 4822, will be charged with the rate for "Poland" 0.05 in our case.

Important: A call will be connected only if the prefix of the dialed number exists in the tariff. All the dialed numbers without matching prefixes in tariff table will be rejected.

Importing tariffs

It is possible to import the tariff rates from a csv or txt file. For this you will have to prepare the file in the following specific format (order of columns):

Prefix, Description, Rate per minute, From day, To day, From hour, To hour, Grace period

Please note that "From day" and "To day" values should be from 0 to 6 where 0 means Sunday and 6 Saturday.

Be sure you do not have column names in the text file, and the separation of the fields is either comma or semicolon. The file should contain only pure data no other comments or column headers and data should start from first row. Also Description field should not exceed 25 characters.

Usually you can work this rate file in Microsoft Excel and save it as csv.

The resulted file should look like this:

355,Albania,0.203,0,6,0,2400,0
213,Algeria,0.194,0,6,0,2400,0
2131,Algeria cellular,0.189,0,6,0,2400,0
684,American samoa,0.119,0,6,0,2400,0

376,Andorra,0.069,0,6,0,2400,0

When you have the file ready, upload it on the Voipswitch server, and from your tariff settings in VPSConfig press ***Import*** button. You will be asked to select the text file. If there will appear an error, VPSConfig will stop the importing process and announce the error.

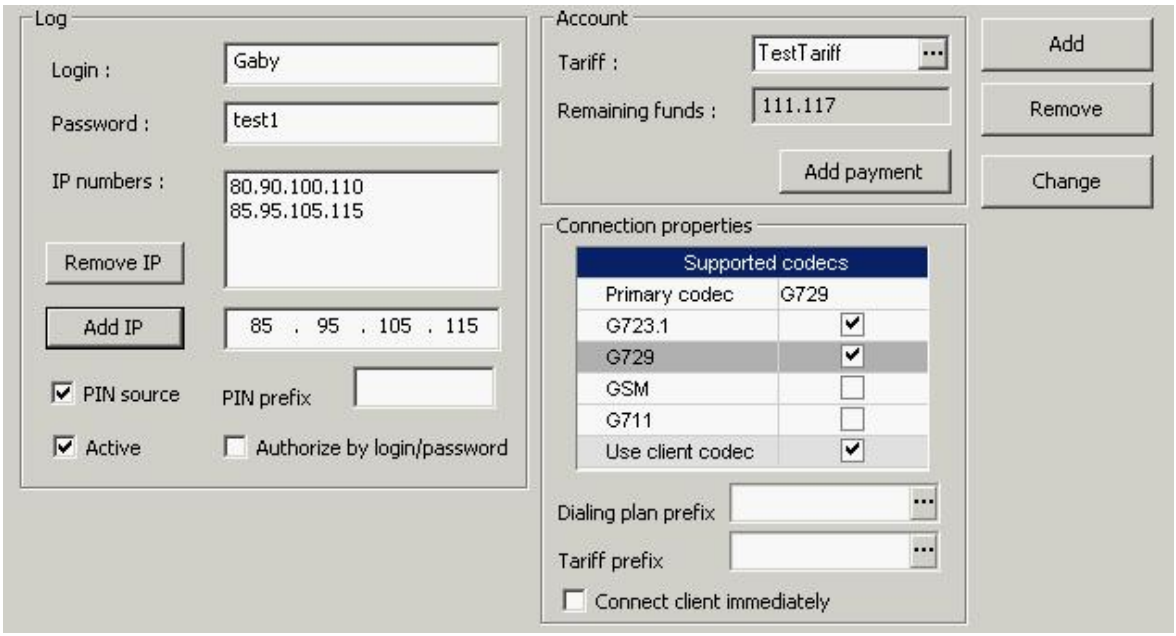
The records from the file will be added to the existing records in the tariff.

If you will want to replace the existing rates, then remove them before importing. There is a ***Remove all*** button there that will delete all rates in that tariff for your convenience.

Now that you have the tariffs ready you can proceed to create Clients.

GWclients

The GWClients are used for voip devices that will send calls to VoipSwitch in direct mode (or IP to IP). They will be authenticated either by the IP address or by the H323ID. GWclients can be either h323 or SIP. The only condition is to send the calls in direct mode.



Log section:
Login field is the username for this account.
Password is the allocated password.

These 2 fields will be used to access the web page to see the CDR's.

Also the *Login@Password* combination is used to match against the H323ID sent by the client in case that *"Authorise by login/password"* feature is enabled.

IP numbers is the list with allowed IP addresses. You can set here an unlimited number of addresses but an IP can be enabled only for one GWClient at a time.

Under the ***IP numbers*** list there is a field where to write the new addresses to be added in the list. Use the ***Add IP*** button after you fill it.

To remove an IP from the list select it first and then click ***Remove IP***.

PIN Source

This feature, if enabled, will grant access to IVR prompts for the GWClient. By default all users can access IVR prompts only to hear the balance. But for other prompts, like ask for PIN or ask for destination number, ***PIN Source*** has to be enabled.

PIN prefix

It is used to define an internal prefix so different clients can use same PIN. For example if 2 clients will want to use same 1234 PIN then we can differentiate them by adding this PIN prefix in front. If PIN prefix is set to 1# for one client and to 2# for another client then users will both dial 1234 but each will be matched against 1#1234 or 2#1234 depending on origination GWclient account. The PIN number entered will be matched against the password field of an IVRclient.

Active

This let's you enable or disable this account.

Authorize by login/password

If this feature is enabled, calls will be allowed also from gateways that send an h323id like Login@Password where Login and Passwords are the values of the fields set for this client. The separator "@" can be changed in H323settings section. Also "Users can log by H323 ID" in that section has to be enabled for this to work.

Account section:

Tariff

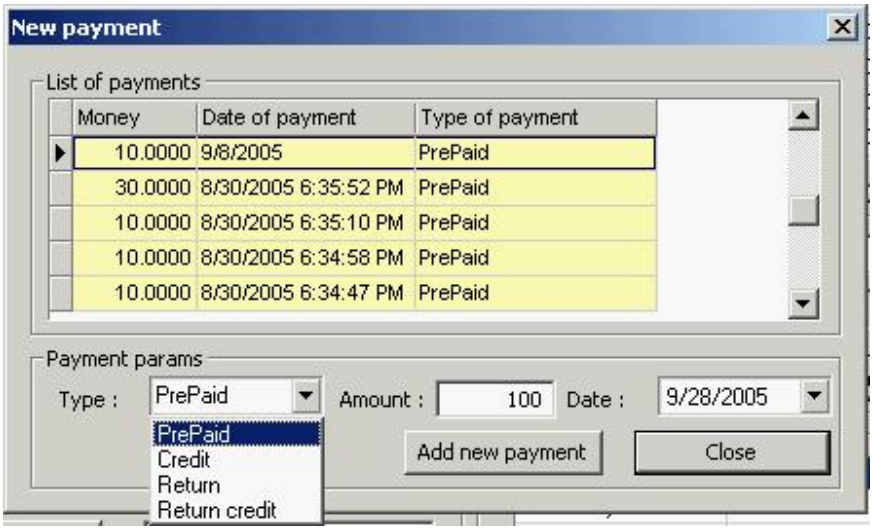
In this field you will have to select the tariff for this client calls. You can do this by clicking on the 3 dots button and select a previous created tariff from the pop-up window that will appear.

Remaining funds

Here is showed the balance of the current account. At the beginning the balance will be zero. And you will be able to ***add payments*** only to accounts that are saved in database (already created).

Add payment

So for already created clients you can press this button and a new window will appear where you can adjust the balance of this client. You also will be able to see a history of all payments there in the ***List of payments***.



There are 4 types of payments available: PrePaid, Return, Credit and Return Credit.

With Prepaid and Credit you can add money to the account and with Return and Return Credit you can subtract.

Prepaid will add money to the balance and will cut the calls of the user when the balance reaches zero. You can subtract money from the amounts added with Prepaid using ***Return***.

Credit is adding credit to the account and the calls will be stopped only when the balance will reach negative value of Credit added. You can subtract from Credit amounts using ***Return Credit***.

Example:

Let's say we want to add 100$ to a client as prepayment. We will select type Prepaid Amount 100 and leave Date the current. Then we click Add new payment button.

The new amount will appear listed in the List of Payments.

Later let's say we find that we should add only 80$ and not 100$. So to correct this we will have to subtract 20$ for the balance. For this we select Type Return, Amount 20 and click Add new payment button.

Now the List of payments will show this amount also as Return and the total balance of this account will became 80$.

When the user makes calls the balance will decrease according to Tariff rates until it reaches zero. Then all the calls will be stopped and no new calls will be allowed.

If you have another client that you want to give Credit, for example 50$, you will select ***Type*** Credit, put ***Amount*** 50 and click ***Add new payment***. 50$ will appear in the list as credit.

If you later want to make his credit only 20$ then you select Return credit as ***Type***, you put ***Amount*** 30$ (to be subtracted) and click ***Add new payment***. The resulting balance will be 20$ credit.

The calls of this user will be allowed until the balance became -20$.

Connection properties section:

Supported codecs

Allows the selection of 4 codec groups depending on what client device can support.

One codec has to be set as primary and it will be the default codec.

Voipswitch supports group of codecs, meaning that if you select g723.1, all kind of g723.1 codecs will be allowed, including g723r63 and g723r53. Same thing for other codec groups.

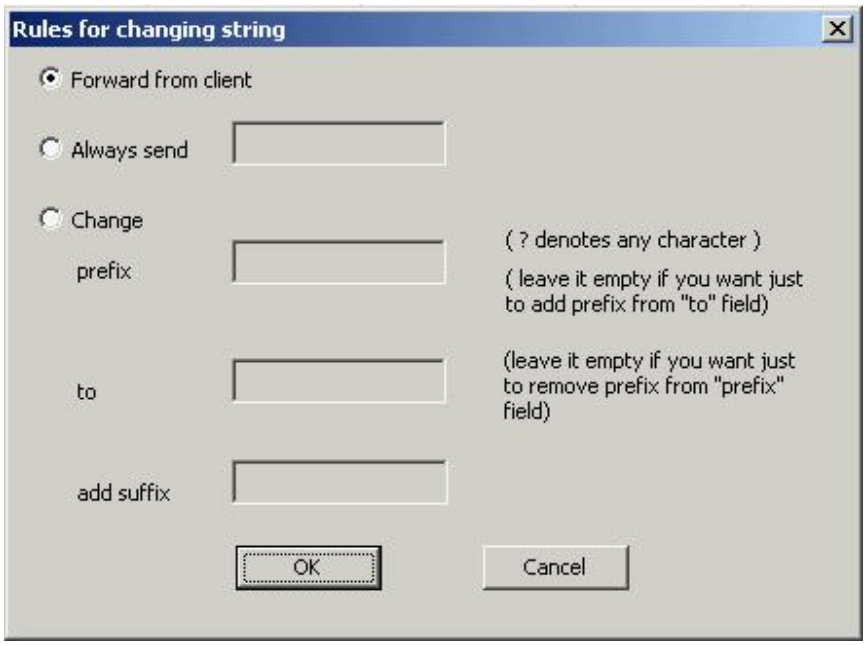
After selection of the codecs you can enable ***Use client codec*** to let VoipSwitch negotiate the right codec from the list with client device. Of course client's device has to be able to autonegotiate codecs.

It is important here to mention that VoipSwitch acts differently in "proxy all" mode and in "proxy only signaling" mode. In "proxy all" VoipSwitch does not allow codec negotiation directly between endpoints and instead will negotiate itself with each endpoint in part. While in "proxy only signaling" the endpoints can negotiate directly the codecs, being possible to choose any codec that both endpoints support even those that are not listed in VoipSwitch settings.

Dialing plan prefix and ***Tariff prefix***

These fields are useful in VoipSwitch internal routing and billing plan. They can accept simple prefixes values that will be added in front of dialed number or complex replacement rules with the rule "A->B": if dialed number starts with A, replace A with B, where B can be nothing.

There is also a helping pop-up window if you press the 3 dots button that will help you to create the rules:



You can write down the value of Dialing plan prefix or Tariff prefix directly (if you know the format) or you can use the helping window.

The options from helping window are:

Forward from client is equivalent with an empty prefix field and it will forward to tariffs or dialingplan exactly what was received from client.
Always send will fill the prefix field in the format "!123" where 123 is the number desired to be sent. That means that the entire dialed number received from client will be always substituted with this number you define here and then sent to tariff or dialing plan depending on prefix field name.
Change will fill the prefix field in the format "X->Y|Z" where X is the prefix field from the helping window, Y is the change to field and Z is the suffix. What means this? If the dialed number received from client starts with X then replace it with Y and add Z at the end of entire number.

If you want to write directly the value in prefix field you can omit "|Z" if you don't want to add a suffix. Also if you want to add only a suffix then the format will be only "|Z".

For better understanding of this mechanism consider that the client (in VoipSwitch, Clients are originators) send a call to VoipSwitch. First, VoipSwitch will want to know how to bill this call depending on the destination so will have to search in the tariff allocated for this client a matching rate. Here comes the help of ***Tariff prefix*** when you want to do special things. For usual cases when the client is dialing exactly with the prefixes you have in tariff you will leave the ***Tariff prefix*** empty. But when the client dials with 00 and in your tariff you have only prefixes without 00 then you enter a replacement rule in ***Tariff prefix*** field like "00->". This will cut the 00 if exists before the number is sent to Tariff to match a rate.

Also you can consider the case that your client always dial without 00 while in your tariffs you have all the prefixes starting with 00. In this case you fill the Tariff prefix with the value "00". So 00 will be added in front of all dialed numbers received from client before they are sent to Tariffs to match a rate.

You can imagine how useful is this because you will not be forced to create one tariff with 00 and another without 00 with same rates. And then from time to time to be forced to update both.

Now that VoipSwitch found the rate and knows how to charge this call will try to send it to Dialingplan to find a matching route for dialed number prefix. And here again we can have a lot of help from the ***Dialingplan prefix*** field. Before the number is sent to the **Dialingplan** for routing we can modify the dialed number by adding a prefix (internal tech prefix) or changing some digits.

Let's take same example when the client dials with 00 and let's consider that in **Dialplan** we have only routes for country codes prefixes. In this case we will fill the ***Dialingplan prefix*** field with the value "00->". That means we will cut 00 (replace 00 with nothing) from numbers dialed by client before we sent them to the **Dialplan** routing. And this is even better because the client can dial either with 00 or without 00 while this replacement rule will cut only if 00 exist at the beginning of number.

Of course you can leave ***Dialingplan prefix*** empty if you have routes in dialplan exactly for what the client is dialing.

But you can create separate rules for each client by providing a different ***Dialplan prefix***. For example we want to route all calls from client A through destination gateway G and all calls from client B, to same country, through gateway H. So we will give to client A a

Dialingplan prefix like for example #11# and to client B a ***Dialingplan prefix*** like **#22#**. Then in **Dialplan** all we have to do is to create routing rules for telephone numbers starting with **#11#** and **#22#**. We will know that calls with **#11#** are coming from client A and those with **#22#** are coming from client B. So we can route same country to different termination gateways without clients to know using the Dialplan prefix as an internal tech prefix.

Connect client immediately

Enable this only when all calls of a client do not connect to any destination. This will open the media channel immediately after routing but in most of the cases will generate also false billing cause the calls will be declared answered immediately.

So this feature is for extreme cases only. Do not use it for normal users.

Pc2phone Clients

This type of client is for pc2phone dialer and web2phone page access only.

Pc2phone is a proprietary application that allows clients who has a valid ***pc2phone client*** account to connect to the Voipswitch and initiate and also receive calls. This dialer uses particular communication ports and is not compatible with other systems.

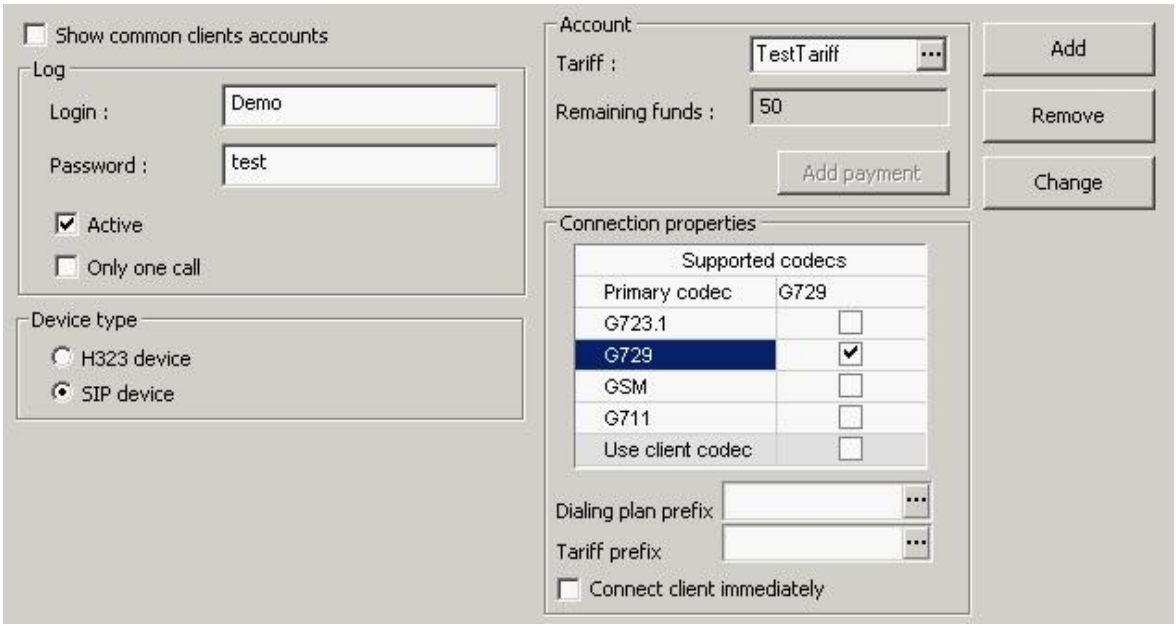
The settings for ***pc2phone clients*** are very simple and the fields have same meanings as for GWclients. Pc2phone application is always using g723.1 codec group so there is no need for codec settings also.

The only new feature is ***Show common clients accounts*** and you will find this in all other type of clients from now on because the ***Common clients*** are a special type of clients that can be used to gain access as any of these types of clients. Common clients have a separate setting section that will be discussed later and that section can be accessed from any type of clients by clicking ***Show common clients accounts.***

GK/Registrar Clients

This client type is used for those devices behind NAT, or those that change the IP often or simply want to register with a user and pass only.

The client will have to configure his device to register to Voipswitch's Gatekeeper (when using h323 protocol) or Registrar (for SIP protocol) using the user and password he received. Also he will need to enter in his device configuration the IP of Voipswitch and the Gatekeeper name that is by default "Gatekeeper" (in case he uses h323).



The majority of the fields here are the same as for GWclients.

The new options are:

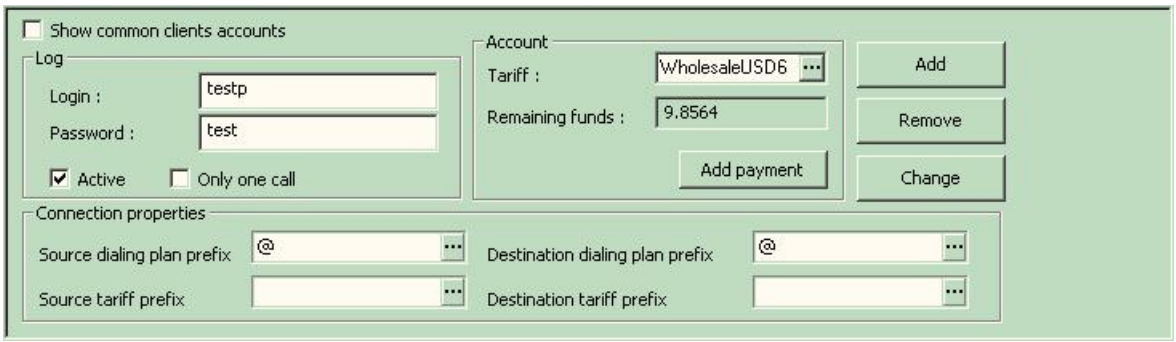
Only one call that force the client to send only one call at a time. If client has already an active call, all other attempts to connect will be rejected. This is very useful to fight against delays in signaling and overlapped billing records for same client.

Device type with its two options H323 device and SIP device allows predefining which protocol will use this client. So if SIP device is selected, calls coming as h323 will not be allowed.

Show common clients accounts option has the same meaning as in ***pc2phone clients.***

Callback clients

Callback client accounts are specially designed for callback service. A callback has 2 telephone legs, one to the origination number (source) and the other one to the destination number.



That's why the callback client settings have separate settings for each call leg, source and destination. Otherwise all the fields have same meanings as before.

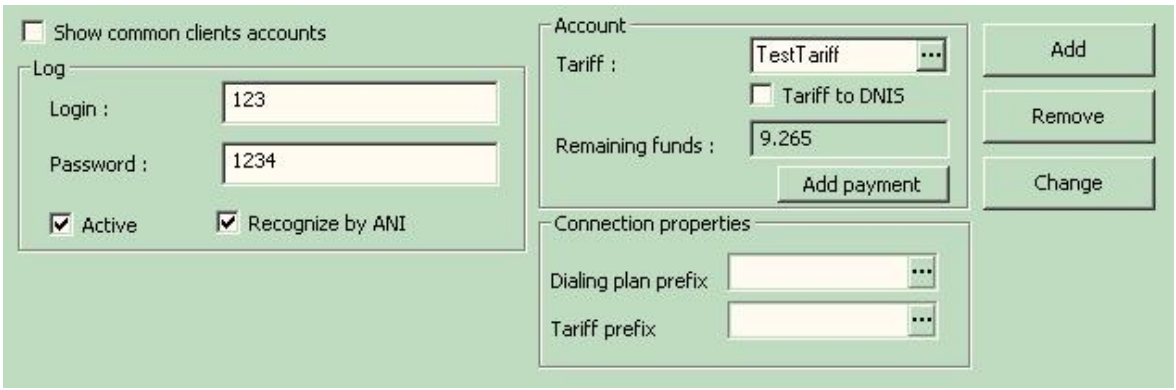
IVR clients

IVRclient accounts are designed to meet the needs of those calls that will be answered with IVR prompts and will be authorized either with a PIN code or with the CLID information.

Therefore when the system will ask for PIN it will compare the received number with the password of an IVRclient account.

Or if the ***Recognize by ANI*** option is enabled the system will compare the ANI (or CLI) number with the *Login* field of an IVRclient account.

If a match will be found the system will allow further calls at the tariff set in this matching IVRclient.



I have to remember you that, to obtain IVR prompts for authorization, a Gwclient needs to be used and the option ***PIN source*** must be enabled for that account.

Common clients or *Shared clients*

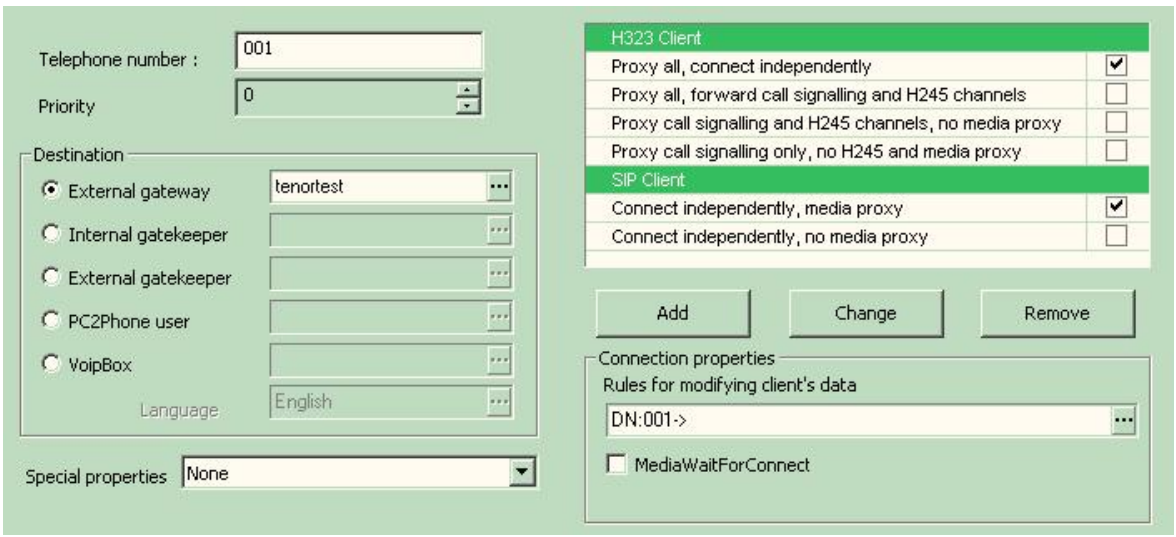
This type of client was made for those cases when you need to give access to your clients to all kind of services (GK registration, Pc2phone, Callback, IVR) using only one account. This is useful in order to maintain one billing account per customer.

This type of clients can be created from a separate section that is showed only when you click ***Show common clients accounts*** option. You will find this option in all the sections that common clients cover. That means in PC2Phone clients, GK/Registrar clients, Callback clients and IVRclients.

By clicking this option the table with common clients will open and you will be able to add new common clients or admin the existent ones. For each common client account you can set all options that you could set for all other type of clients that are covered.

Dialingplan

This is the section where the routing settings are made. Here you can decide where to send a call depending on the prefix of the dialed number.



Telephone number field is the prefix of the telephone number to route. It can be also the entire destination number.

In Dialingplan you can define multiple routes for same destination in case you want to have backup for example. The routes are chosen in the following order:

- the longest ***Telephone number*** match is selected first
- and then the rule with lowest ***Priority*** number (zero is the first, 1 is the second…).

Priority can take integer positive values starting with 0. The rule with priority 0 will be selected first if the ***Telephone number*** is matching the destination number (prefix).

If there are multiple rules for same ***Telephone number*** with same priority they will be chosen in the order they were introduced in database.

Destination defines where to route the call if the ***Telephone number*** was matched.

A call can be sent to:

- an ***External gateway*** defined previously in ***Gateways*** section,
- an ***Internal gatekeeper*** client defined in ***GK/Registrar*** section and registered in the system,
- an ***External gatekeeper*** defined previously in ***GK/Registrars*** section,
- an existent ***PC2Phone user***,
- to the ***VoipBox***, that is an external application that can generate IVR prompts.

Special properties

- ***none*** : special properties are disabled
- ***do not jump*** : when you have multiple rules for same prefix you can enable this option to stop the hunting when the call will fail through the current rule.
- ***Prefix not allowed*** will disable all calls to the prefix defined in ***Telephone number*** field.

- ***Route disabled*** : with this option you can temporary disable a route without loosing the settings.
- ***Map DNIS to GK/Registrar accounts***
- ***Map DNIS to PC2Phone accounts***
- ***Map DNIS to common clients accounts***

  All these ***Map DNIS to…*** features will automatically route the calls to the ***GK/Registrar*** or ***PC2Phone*** or ***Common client*** account that will have the ***Login*** as the dialed number.

  So, for example, to route calls internally between all your pc2phone clients all you need to do is to create the pc2phone accounts with distinct numbers as ***Login*** name and then add a ***Dialingplan*** rule having ***Map DNIS to PC2Phone accounts*** enabled.

  ! When creating this ***Dialingplan*** rule you can select anything as ***Telephone number*** or ***Destination*** because these fields will not count in the routing process, but they need to be set!

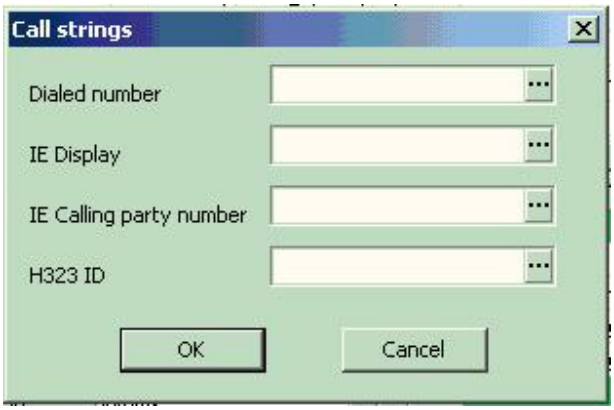
Proxy settings

- ***Proxy all, connect independently***: origination and termination endpoints do not see each other, the VPS connects independently with each endpoint then conference them together.
- ***Proxy all, forward call signaling and H245 signaling channels***: the signaling and media will still be passed through Voipswitch as in first rule. The difference between this option and the previous is that the call setup received from the client is sent to the target gateway. So the two endpoints can use more codecs if they both support them (even if Voipswitch do not support it). Also, information coming from a client through h245 channel is forwarded directly to the termination gateway. So H245 tunneling can be used (if both endpoints support it).
- ***Proxy call signaling and H245 channels, no media proxy***: only signaling information and H245 channel are passed through the switch, media packages are sent directly between endpoints.
- ***Proxy call signaling only, no H245 and media proxy***: in this mode only signaling information is passed through switch. All the rest are flowing directly between the two endpoints.

Rules for modifying client's data

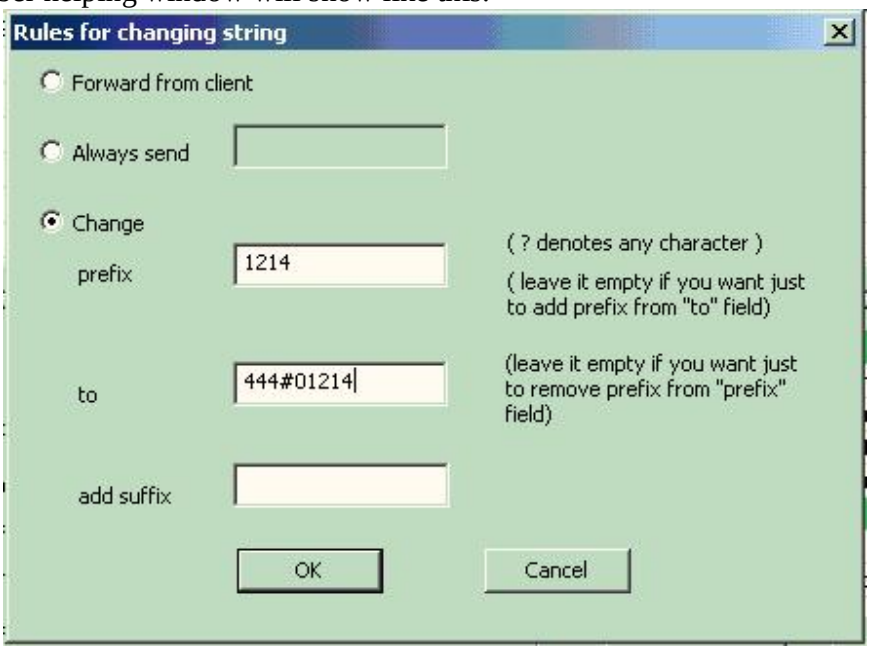
This field is complex and allows modifying different call settings last time before the call is sent to the termination gateway.

At the end of this field is a button with 3 dots. This will open a helping window that will guide you through the possible settings for this rule.



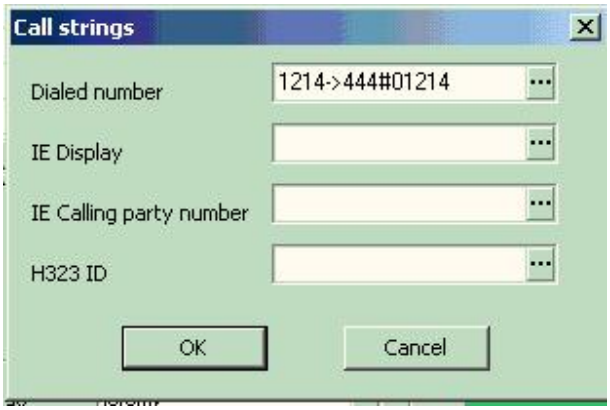
As seen above, with this rule is possible to change the Dialed number, Caller ID or H323ID. Clicking the 3 dots buttons at the end of each field additional windows will help you to set correctly the rules.

For example termination gateways require often receiving the dialed numbers with a tech prefix for authentication. If your dialplan rule was for a telephone number 1214 for example and the termination gateway needs to receive a tech prefix like 444# plus 0 then the Dialed number helping window will show like this:



As you see the original telephone number 1214 will be changed into 444#01214 and then will be sent to the termination gw.

Also you can write directly the replacement rule in the previous window. Like:



IE Display and ***IE Calling party number*** are 2 fields from H323 protocol that refers to Caller ID information. If you want to modify the callerID sent to termination you should modify either one or both fields depending on termination provider (some accept first field others work with the second field).

H323ID sent to the termination gw can be modified or defined here also.

Note: All these h323 field changes will be taken in consideration only when routing h323 calls! If you want to modify the caller ID for a SIP call then first route it as h323 to own and then forward the information to termination gw as SIP!

MediaWaitForConnect if enabled will make Voipswitch to instruct the origination device to generate fake ring tone while waiting for connection. Not all devices can generate fake ring. But Cisco ATA and others like that can, so this helps sometimes when remote gateway do not send proper alerting.

Tariff to DNIS

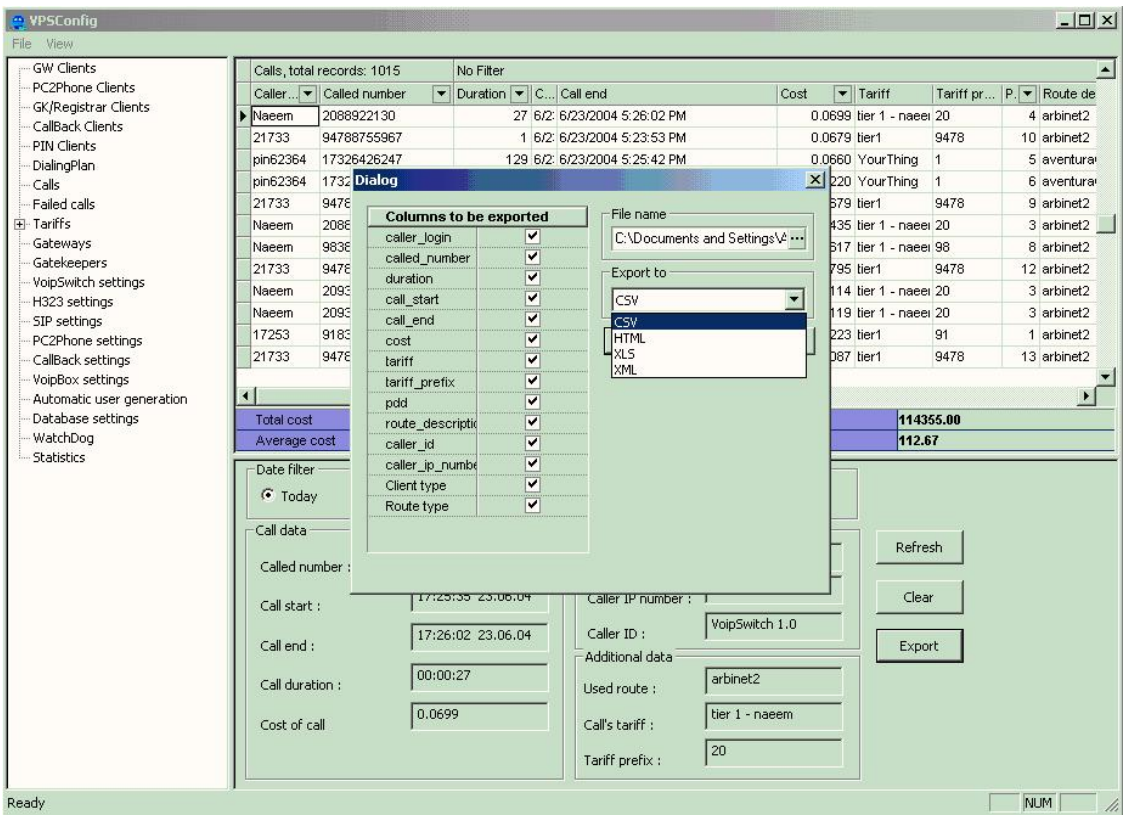
With this feature you can bill differently the IVRclients depending by the access number. To activate this feature for an IVRclient you will need to enable ***Tariff to DNIS*** option in that IVRclient account settings (see IVRclients picture).

You can define here in ***Dialed number*** field different access numbers and associate with them tariffs in ***Tariff*** field.

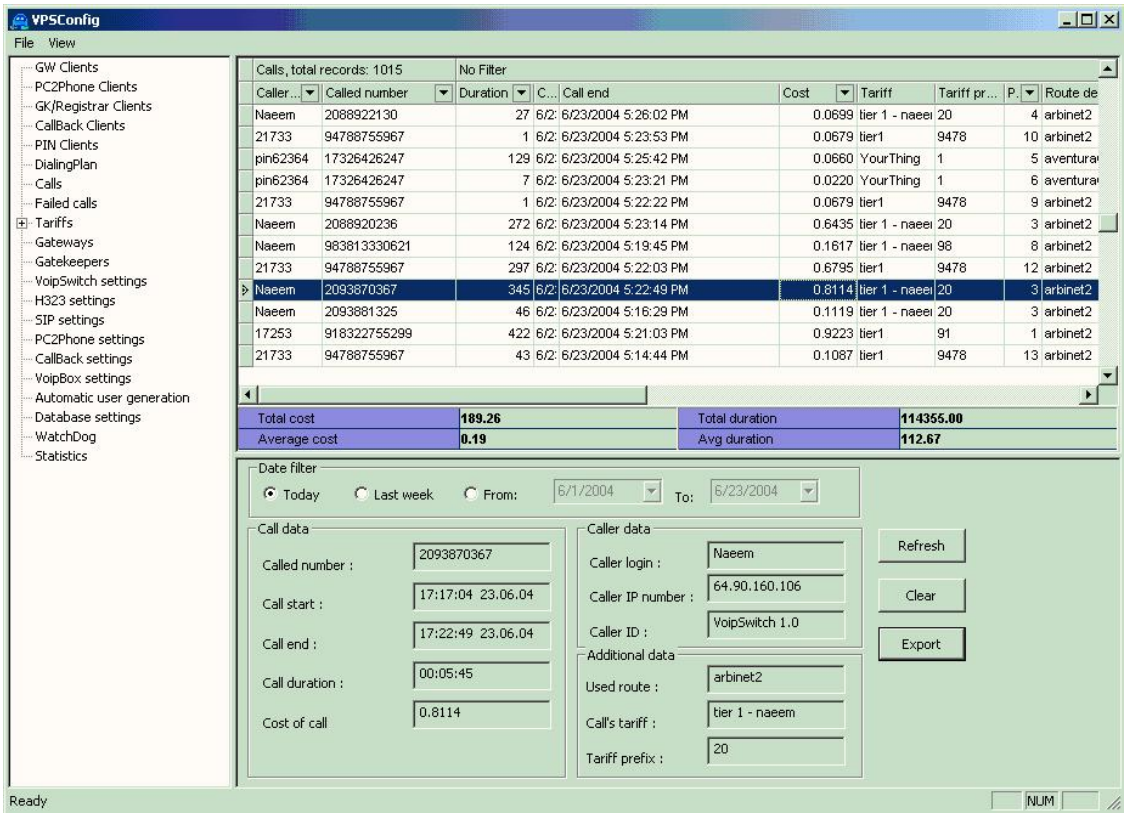
All the authorized IVRclients coming on one of the access numbers defined here will be billed according with the associated tariff defined here (and not to the tariff defined in IVRclients).

Calls and Failed Calls

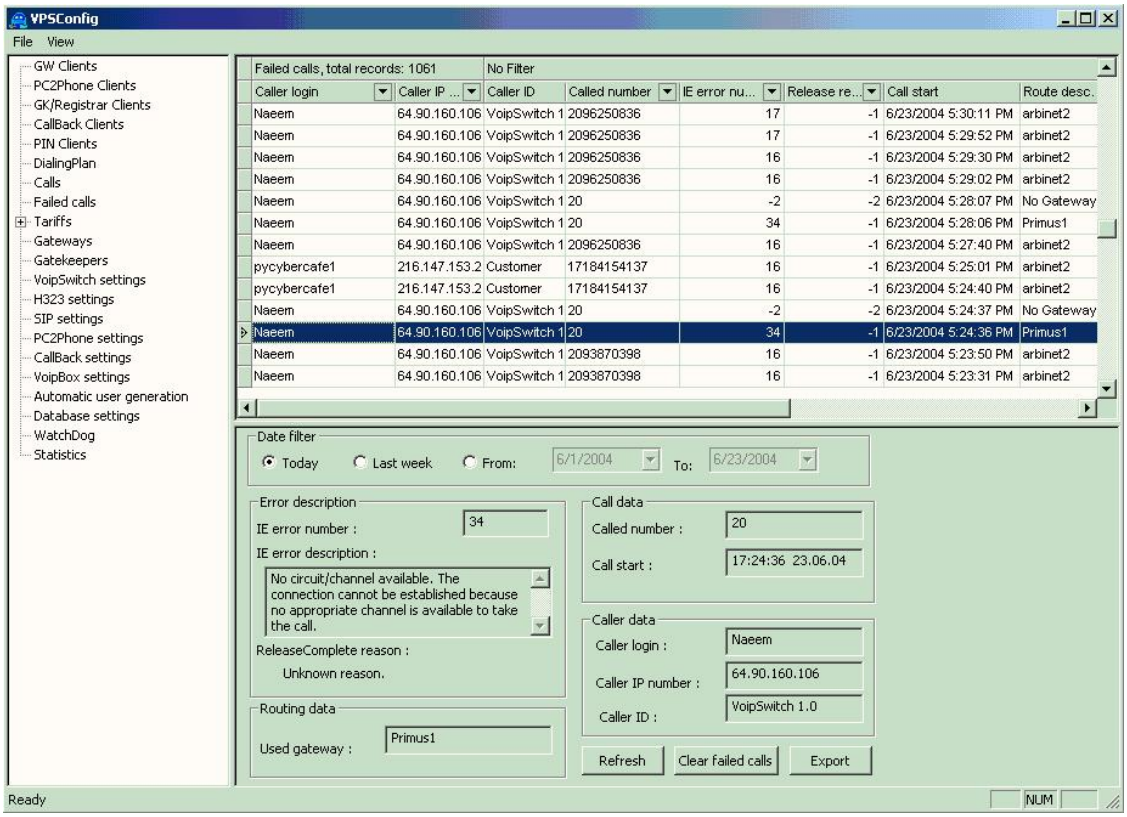
Calls section displays the list of all the calls made in a given period. For each call there is detailed information on the duration of call, cost, name of the client, route the connection utilized and many other useful information. By using filters only the desired records can be displayed e.g. in order to generate CDR for a particular client. "Export" button enables saving records to csv, xls, html or xml files.



Below the list there is information on total cost of all calls, average cost of a call, total time of calls and average time of a call.

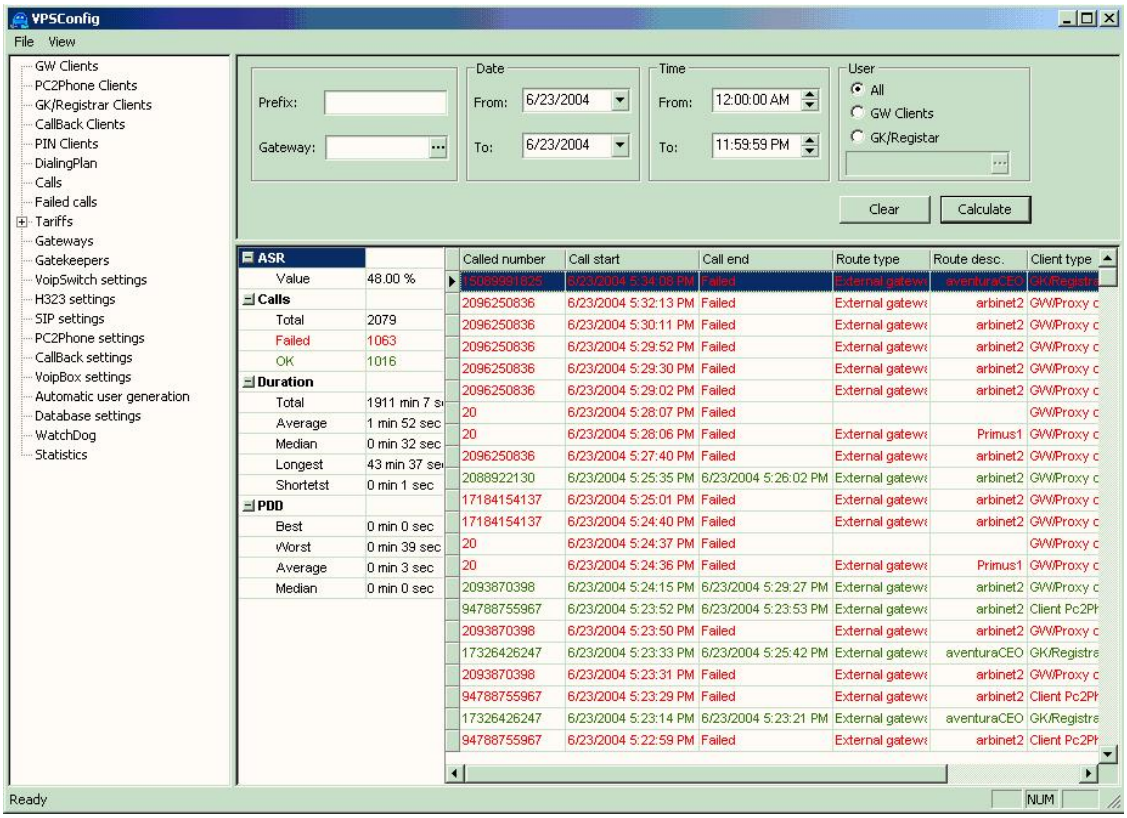


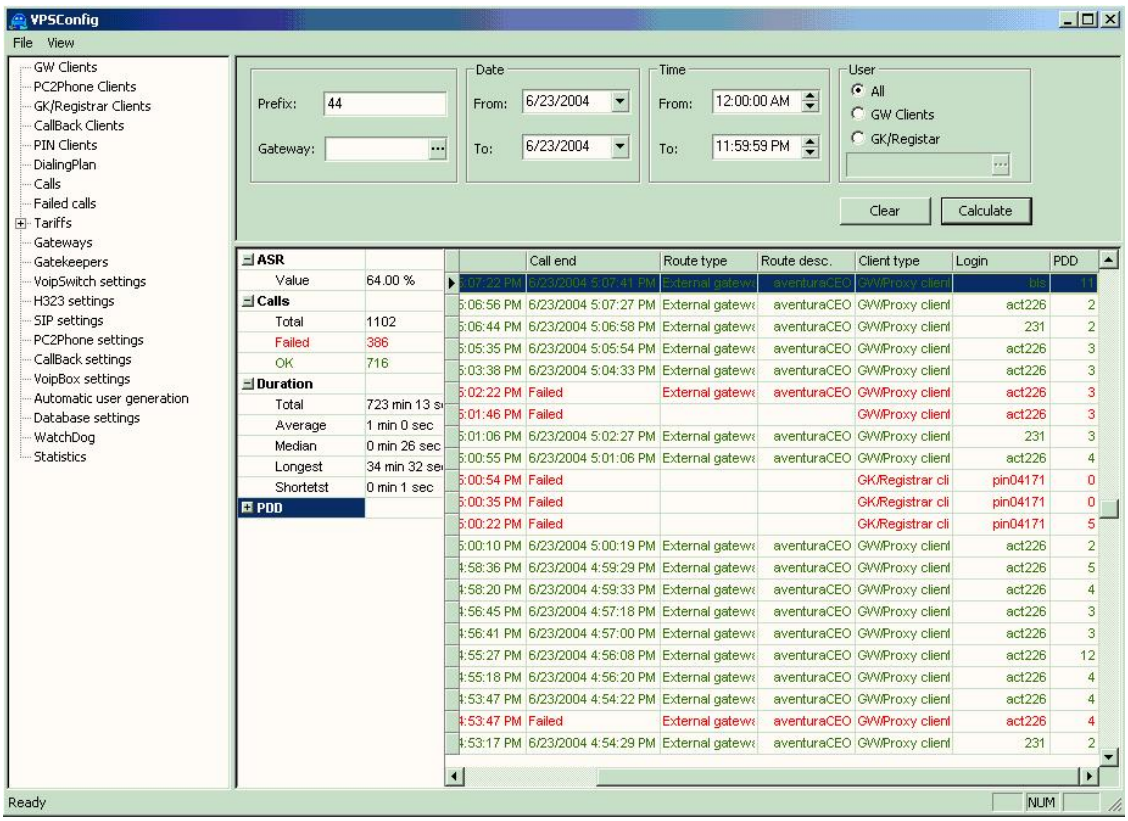
Failed calls section is organized similarly with the exception that it presents different set of information. Connections that have failed, for various reasons, are displayed here. Description of errors and release reasons helps identifying the potential problems



Statistics

In the ***Statistics*** section all the connections are displayed both successful and failed in a given period of time along with ASR, ACD and PDD indexes. It is possible to limit the statistics to the selected, desired information e.g. calls sent only to the particular gateway, particular prefix or statistics referring only to one client.

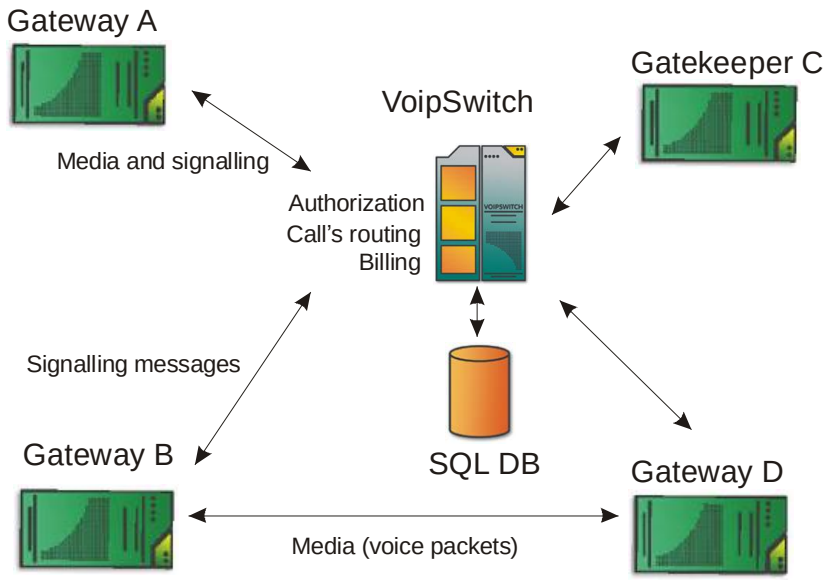

VPSConfig
File View
GW Clients
PC2Phone Clients
GK/Registrar Clients
CallBack Clients
PIN Clients
DialingPlan
Calls
Failed calls
Tariffs
Gateways
Gatekeepers
VoipSwitch settings
H323 settings
SIP settings
PC2Phone settings
CallBack settings
VoipBox settings
Automatic user generation
Database settings
WatchDog
Statistics
Prefix:
Gateway:
Date
From: 6/23/2004
To: 6/23/2004
Time
From: 12:00:00 AM
To: 11:59:59 PM
User
All
GW Clients
GK/Registar
Clear
Calculate
ASR
Value 48.00 %
Calls
Total 2079
Failed 1063
OK 1016
Duration
Total 1911 min 7 s
Average 1 min 52 sec
Median 0 min 32 sec
Longest 43 min 37 se
Shortetst 0 min 1 sec
PDD
Best 0 min 0 sec
Worst 0 min 39 sec
Average 0 min 3 sec
Median 0 min 0 sec
Called number | Call start | Call end | Route type | Route desc. | Client type
2096250836 | 6/23/2004 5:32:13 PM | Failed | External gatew | arbinet2 | GW/Proxy c
2096250836 | 6/23/2004 5:30:11 PM | Failed | External gatew | arbinet2 | GW/Proxy c
2096250836 | 6/23/2004 5:29:52 PM | Failed | External gatew | arbinet2 | GW/Proxy c
2096250836 | 6/23/2004 5:29:30 PM | Failed | External gatew | arbinet2 | GW/Proxy c
2096250836 | 6/23/2004 5:29:02 PM | Failed | External gatew | arbinet2 | GW/Proxy c
20 | 6/23/2004 5:28:07 PM | Failed | | | GW/Proxy c
20 | 6/23/2004 5:28:06 PM | Failed | External gatew | Primus1 | GW/Proxy c
2096250836 | 6/23/2004 5:27:40 PM | Failed | External gatew | arbinet2 | GW/Proxy c
2088922130 | 6/23/2004 5:25:35 PM | 6/23/2004 5:26:02 PM | External gatew | arbinet2 | GW/Proxy c
17184154137 | 6/23/2004 5:25:01 PM | Failed | External gatew | arbinet2 | GW/Proxy c
17184154137 | 6/23/2004 5:24:40 PM | Failed | External gatew | arbinet2 | GW/Proxy c
20 | 6/23/2004 5:24:37 PM | Failed | | | GW/Proxy c
20 | 6/23/2004 5:24:36 PM | Failed | External gatew | Primus1 | GW/Proxy c
2093870398 | 6/23/2004 5:24:15 PM | 6/23/2004 5:29:27 PM | External gatew | arbinet2 | GW/Proxy c
94788755967 | 6/23/2004 5:23:52 PM | 6/23/2004 5:23:53 PM | External gatew | arbinet2 | Client Pc2Ph
2093870398 | 6/23/2004 5:23:50 PM | Failed | External gatew | arbinet2 | GW/Proxy c
17326426247 | 6/23/2004 5:23:33 PM | 6/23/2004 5:25:42 PM | External gatew | aventuraCEO | GK/Registra
2093870398 | 6/23/2004 5:23:31 PM | Failed | External gatew | arbinet2 | GW/Proxy c
94788755967 | 6/23/2004 5:23:29 PM | Failed | External gatew | arbinet2 | Client Pc2Ph
17326426247 | 6/23/2004 5:23:14 PM | 6/23/2004 5:23:21 PM | External gatew | aventuraCEO | GK/Registra
94788755967 | 6/23/2004 5:22:59 PM | Failed | External gatew | arbinet2 | Client Pc2Ph
Ready
NUM


VPSConfig
File View
GW Clients
PC2Phone Clients
GK/Registrar Clients
CallBack Clients
PIN Clients
DialingPlan
Calls
Failed calls
Tariffs
Gateways
Gatekeepers
VoipSwitch settings
H323 settings
SIP settings
PC2Phone settings
CallBack settings
VoipBox settings
Automatic user generation
Database settings
WatchDog
Statistics
Prefix: 44
Gateway:
Date
From: 6/23/2004
To: 6/23/2004
Time
From: 12:00:00 AM
To: 11:59:59 PM
User
All
GW Clients
GK/Registar
Clear
Calculate
ASR
Value 64.00 %
Calls
Total 1102
Failed 386
OK 716
Duration
Total 723 min 13 s
Average 1 min 0 sec
Median 0 min 26 sec
Longest 34 min 32 se
Shortetst 0 min 1 sec
PDD
Call end | Route type | Route desc. | Client type | Login | PDD
5:07:22 PM | 6/23/2004 5:07:41 PM | External gatew | aventuraCEO | GW/Proxy client | bis | 11
5:06:56 PM | 6/23/2004 5:07:27 PM | External gatew | aventuraCEO | GW/Proxy client | act226 | 2
5:06:44 PM | 6/23/2004 5:06:58 PM | External gatew | aventuraCEO | GW/Proxy client | 231 | 2
5:05:35 PM | 6/23/2004 5:05:54 PM | External gatew | aventuraCEO | GW/Proxy client | act226 | 3
5:03:38 PM | 6/23/2004 5:04:33 PM | External gatew | aventuraCEO | GW/Proxy client | act226 | 3
5:02:22 PM | Failed | External gatew | aventuraCEO | GW/Proxy client | act226 | 3
5:01:46 PM | Failed | | | GW/Proxy client | act226 | 3
5:01:06 PM | 6/23/2004 5:02:27 PM | External gatew | aventuraCEO | GW/Proxy client | 231 | 3
5:00:55 PM | 6/23/2004 5:01:06 PM | External gatew | aventuraCEO | GW/Proxy client | act226 | 4
5:00:54 PM | Failed | | | GK/Registrar cli | pin04171 | 0
5:00:35 PM | Failed | | | GK/Registrar cli | pin04171 | 0
5:00:22 PM | Failed | | | GK/Registrar cli | pin04171 | 5
5:00:10 PM | 6/23/2004 5:00:19 PM | External gatew | aventuraCEO | GW/Proxy client | act226 | 2
4:58:36 PM | 6/23/2004 4:59:29 PM | External gatew | aventuraCEO | GW/Proxy client | act226 | 5
4:58:20 PM | 6/23/2004 4:59:33 PM | External gatew | aventuraCEO | GW/Proxy client | act226 | 4
4:56:45 PM | 6/23/2004 4:57:18 PM | External gatew | aventuraCEO | GW/Proxy client | act226 | 3
4:56:41 PM | 6/23/2004 4:57:00 PM | External gatew | aventuraCEO | GW/Proxy client | act226 | 3
4:55:27 PM | 6/23/2004 4:56:08 PM | External gatew | aventuraCEO | GW/Proxy client | act226 | 12
4:55:18 PM | 6/23/2004 4:56:20 PM | External gatew | aventuraCEO | GW/Proxy client | act226 | 4
4:53:47 PM | 6/23/2004 4:54:22 PM | External gatew | aventuraCEO | GW/Proxy client | act226 | 4
4:53:47 PM | Failed | External gatew | aventuraCEO | GW/Proxy client | act226 | 4
4:53:17 PM | 6/23/2004 4:54:29 PM | External gatew | aventuraCEO | GW/Proxy client | 231 | 2
Ready
NUM

Exemplary scenarios

Wholesale termination

VoipSwitch allows managing inbound and outbound traffic in the wholesale operations. In this scenario, VoipSwitch is a central point in the VoipSwitch's network architecture that accepts connections coming from the Voip's gateways and gatekeepers, authorizes them and sends further to gateways or gatekeepers that terminate calls. VoipSwitch keeps the account of connection charges and controls clients' accounts of prepaid or credit type.
Information on all the connections is saved in a SQL database. The chart below shows the exemplary VoipSwitch's implementation.



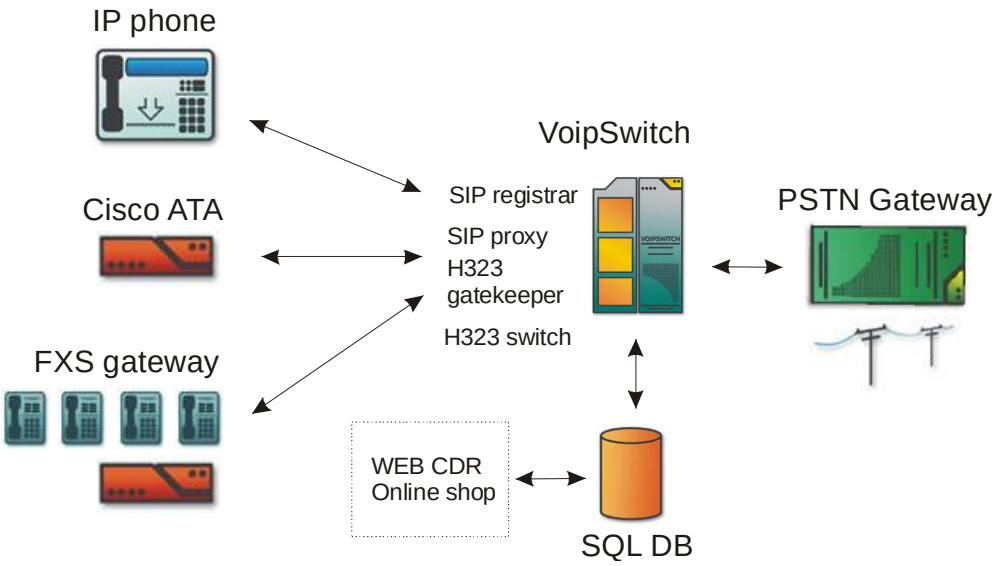
Picture 2 Exemplary scenario

Calls coming from the gateway A are authorized, based on IP address, and sent further, according to the dialing plan, to gatekeeper C or gateway D. The connection is carried out in "proxy all" mode. Both media and the signalling messages are transferred through switch. VoipSwitch calculates the cost of each call based on the tariff assigned to a client.

The gateway B connects with the gateway D in “signaling proxy” mode; only the control information goes through switch while media are sent directly to the target gateway.
Clients can make calls either from h323 or SIP devices.
VoipSwitch, acting as an h323 gateway, can register to multiple external gatekeepers simultaneously.
In the example above, gateway A and gateway B are defined in “GWclients” menu.
Login and password are not used during the authorization process, however they allow clients to access website with the information on the connections made and the current account status. Gateways’ IP addresses are used for the authorization purposes. There are two accounts types: prepaid and credit. Connection is automatically terminated by switch if the set limit is exceeded.
Gateway D and gatekeeper C, to which the connections are sent from the VoipSwitch, are defined in “gateways” and “gatekeepers” menus respectively.
Proxy modes are set in “dialing plan” menu.

Accepting connections from SOHO clients.

VoipSwitch allows authorizing the calls, coming from small Voip gateways or Ipphones, based on the login and password (both for h323 and SIP). That enables providing services to clients that cannot be authorized based on IP address (dynamic IP). Clients are able to send calls either in peer-to-peer mode or register to VoipSwitch as to the gatekeeper. VoipSwitch supports connections coming from both h323 clients and SIP.
For SIP clients VoipSwitch can be treated as a SIP proxy, SIP gateway or SIP registrar.
It is possible to carry out connections between H323 devices and SIP. Switch performs conversion from h323 to SIP and from SIP to H323.



In the above example ipphone registers to VoipSwitch as to SIP registrar. Cisco ATA is set up to send the calls directly to SIP Proxy port, four ports gateway is registered to VoipSwitch as to the h323 gatekeeper. All the clients that are registered can initiate calls between each other and carry out outbound calls to Voip Gateway. Clients, calling in peer-to-peer mode (regardless if they are h323 or SIP clients), are defined in "GWclients" menu; authorization is conducted based on login/password pair. In case of h323 device, login@password string needs to be entered in h323ID field in particular device's settings ("@" acts as a separating character and can be set up in VoipSwitch's options).

Ipphones A that register to SIP are defined in "GK/Registrar clients" menu. Authorization is based on login/password pair.

The H323 gateway, as in the example, registers to VoipSwitch as to the h323 gatekeeper.

Authorization is conducted based on the login@password string defined as an alias in the gateway's settings.

Calling from the H323 equipment

Voipswitch system works with the different type of VOIP equipment compatible with the h323 protocol. VOIP gateways, or IP phones can connect with the system in two different modes:

Direct mode – gateway to gateway – in this mode switch functions as a gateway.
Gatekeeper mode – registering to the system as to the gatekeeper.

Direct mode

Direct mode is strongly recommended in the case of sending calls from the VOIP gateway that uses many channels simultaneously. In this case switch functions as a h323 gateway and the connection should be send to its IP address, to the 1720 port. Clients account in this type of service should be created in the "GW clients" menu.

Authorization for the calls make in Direct Mode

- by the IP address of the equipment the call originated from
- by the equipment's h323ID

In the case of authorization by the equipment's h323ID, option "authorize by h323id" has to be activated for the particular account in the "GW clients" menu. The string login@password has to be entered to the equipment that the calls originate from (@ serves as a separating character). The same pair of login and password can be used to access the web interface with the information on the account status and calls made.

In the case of authorization by the IP login and password are not important (they can be used to log in to the web interface from CDR). It is allowed to enter more than one IP address for each client.

Gatekeeper mode

In this mode VOIP gate or IP phone registers to the switch the same way it registers to the gatekeeper. User account for this type of clients should be set up in the "GK/Registrar" menu. String login@password has to be entered in the alias field (h323id, e164 alias) in the equipment that is to be registered.

Calling from the SIP equipment

Equipment based on the SIP protocol can connect through the switch in the two following modes:

SIP proxy

SIP registrar

For the clients calling in the SIP proxy mode, accounts should be set up in the "GW Clients" menu. Connection can be authorized based on the equipment's IP or login/password pair.

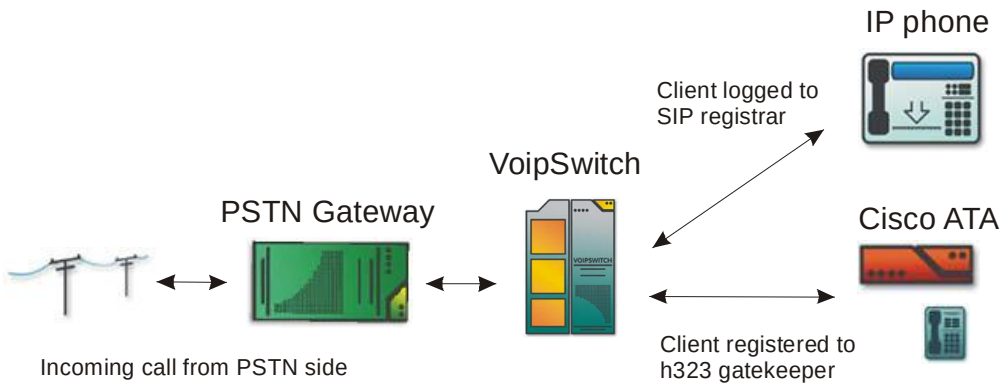
For the clients that register to SIP registrar, accounts should be set up in "GK/Registrar clients" menu. Authorization takes place through entering login/password pair in the proper fields in the equipment's config.

Assigning telephone numbers to the VoipSwitch's clients

Each client that is logged to the gatekeeper or SIP registrar can have his/her unique telephone number assigned. In order to accomplish that particular number has to be defined in the "dialing plan" menu, "internal gatekeeper" option has to be selected and the user, that incoming calls are to be directed to, has to be chosen from the clients list.

VoipSwitch will transmit the calls only if the user is logged.

In this way DID numbers can be assigned to the Voip's clients, which is illustrated on the picture below.



Calls, coming form the PSTN network to the gateway, are redirected to VoipSwitch, which in turn, based on the “dialing plan”, transmits them to the Voip’s clients (e.g. Cisco ATA). Using this method the group of local numbers that we possess, e.g. in the USA, can be directed to the clients’ devices regardless of their location.

Also, it is possible to use an option “Map DNIS to username” which allows for mapping a group of phone numbers to the gatekeeper/regsitrar’s users. In this case the user’s login represents its phone number.

The same features are also available for pc2phone users.

VoipSwitch supports calling to the SIP devices behind NAT.