



## **JVS Softswitch System Administration Guide**

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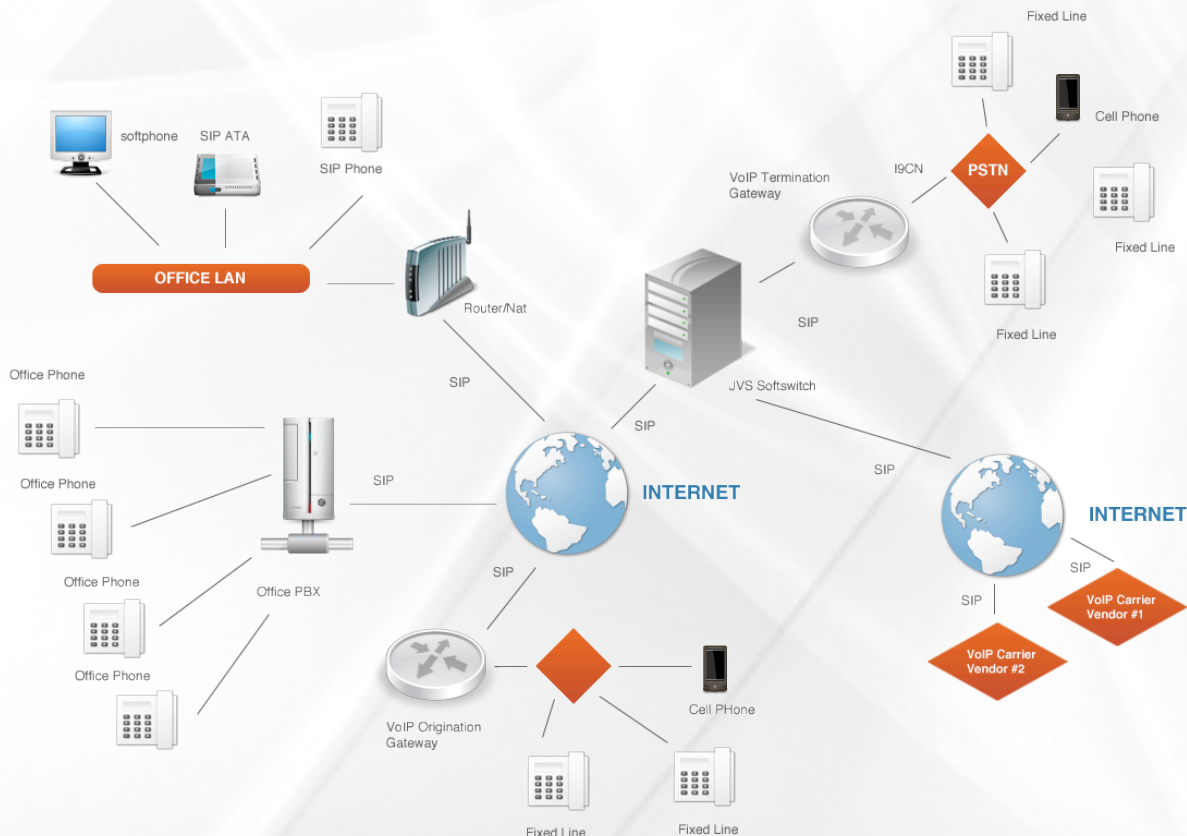
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## Product Description

The JOOD-VS Softswitch (“Softswitch”) is a carrier-grade session controller and billing solution that provides a single interface between multiple IP networks for delivery of end-to-end VoIP communications.

It enables safe and secure interconnection with customers and peering partners, protects your network from unauthorized access and DoS attacks, and provides for reliable and cost-effective control of VoIP calls that go through your network.



The Softswitch is a call control software package enabling service providers to build scalable and reliable VoIP networks. Based natively on the Session Initiation Protocol (SIP),

The Softswitch provides a full array of call routing capabilities to maximize performance for both small and large packet voice networks.

By combining advanced VoIP billing, smart routing, network protection and border control mechanisms all on a single platform the Softswitch delivers complete network management and billing solution for VoIP carriers.

The Softswitch allows IP Telephony Service Providers (“ITSPs”) to deliver communication services at unusually low initial and operating costs that cannot be matched by yesterday’s circuit-switched and narrowband service provider PSTN networks.

In addition to conventional IP-telephony services, the Softswitch provides a complete solution to the NAT traversal problem and enhances ITSP network management capabilities.



## System Concepts

There are four main entities in the system, around which all of the Softswitch activity revolves.

Those entities are: Account, Customer, Connection and Vendor. This chapter will describe their meaning and purpose in detail. It will also describe the call process and how those entities are used during it.

### Accounts

Account is a logical entity usually representing some logical or physical end point in the VoIP network. Example of physical endpoint is SIP phone or individual port in a multi-port device, example of logical end point is single DID number in a pool of DID numbers received from PSTN-SIP gateway.

**Each call going through the system is associated with one and only one account.**

The Softswitch establishes association between the new incoming call and some particular Account during the Authentication phase and once established this association then remains unchanged during all lifetime of the call.

The Authentication could either be performed based on VoIP Login and VoIP Password parameters, using SIP secure digest authentication method, or based on Authentication Rules associated with each account. Please note that the SIP registration with the Softswitch is only allowed using SIP secure digest authentication.

### Customers

Customers are logical entities in the system for which invoices will be generated. They may be termination partners, resellers of a service, or a virtual Customer for keeping track of pre-paid cards distributed by the company. A Customer can have several Accounts associated with it.

These Accounts may represent individual users of a product, such as pre-paid cards, or may identify different SIP UAs (e.g. phones or individual ports in multi-port devices) belonging to the same Customer.

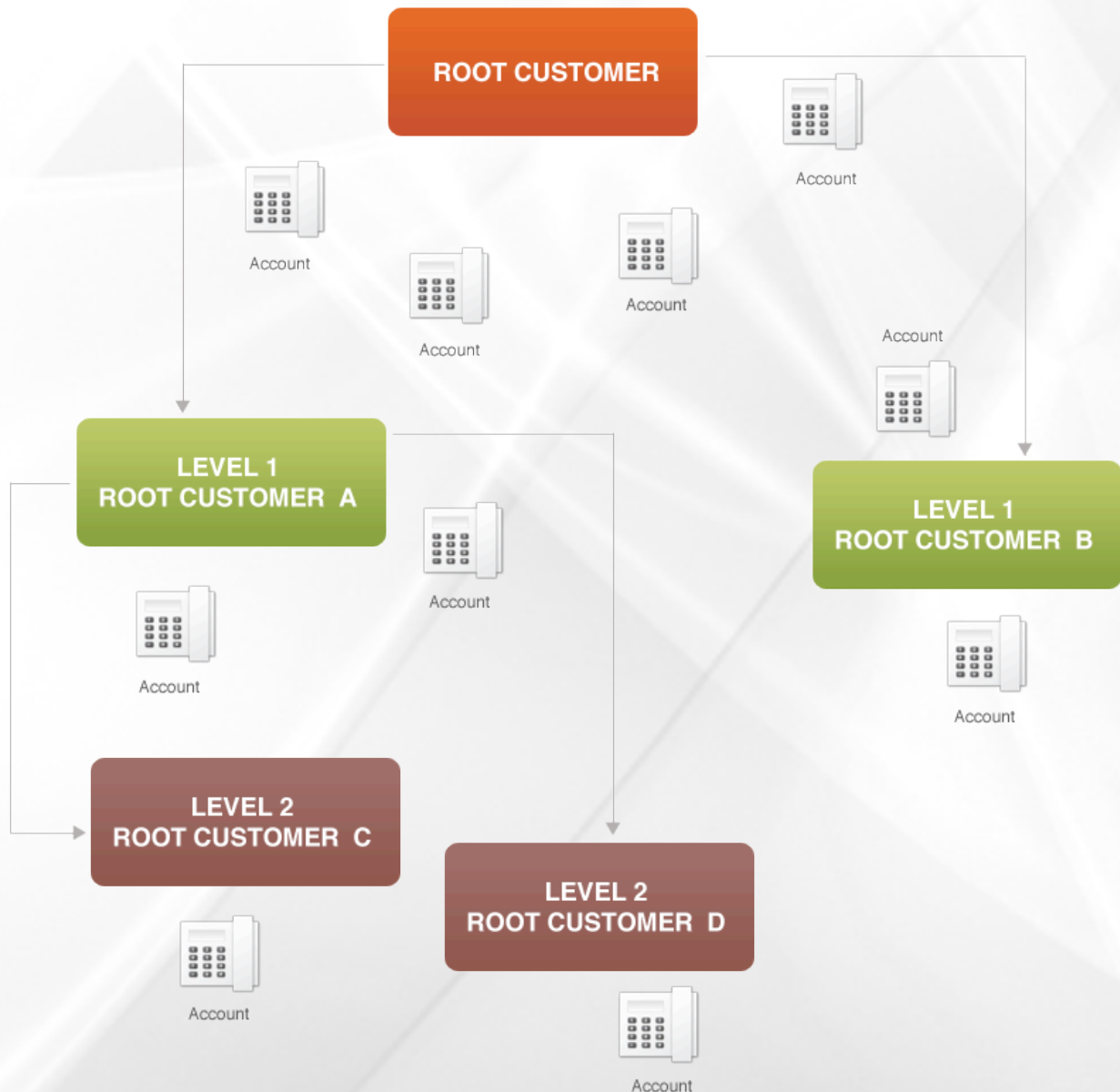
Customers are organized hierarchically so that one customer may “own” one or more sub-customers.

There are two basic types of customers: Root Customer and all other customers. They differ by the fact that Root Customer doesn't have any another upper-level Customers who “owns” him.

Also, Root Customer either directly or via his sub-customers bills other customers in the system. It usually represents owner of the switch. Unlike any other Customer in the system the Root Customer isn't billed for the calls his accounts or sub- customers make. Also, by default only Root Customer has a right to view and/or modify system-wide parameters, such as for example routing configuration, though he/she can delegate some of those rights to selected sub-Customers via access control lists (ACLs).

Customer could be, for example, a company with an account for each employee. The company will be invoiced for all usage, and the Softswitch will charge the company for the total, but the company will be able to see details of the usage for each account and bill them if necessary.

The following figure illustrates this concept:



Here you can see the system with 5 Customers, and 3 hierarchy levels. It should be noted that the system doesn't allow a Customer to see any other customers above it, so that there is no way that the Customer D for example will be able to tell that there are some other levels above the Customer A.

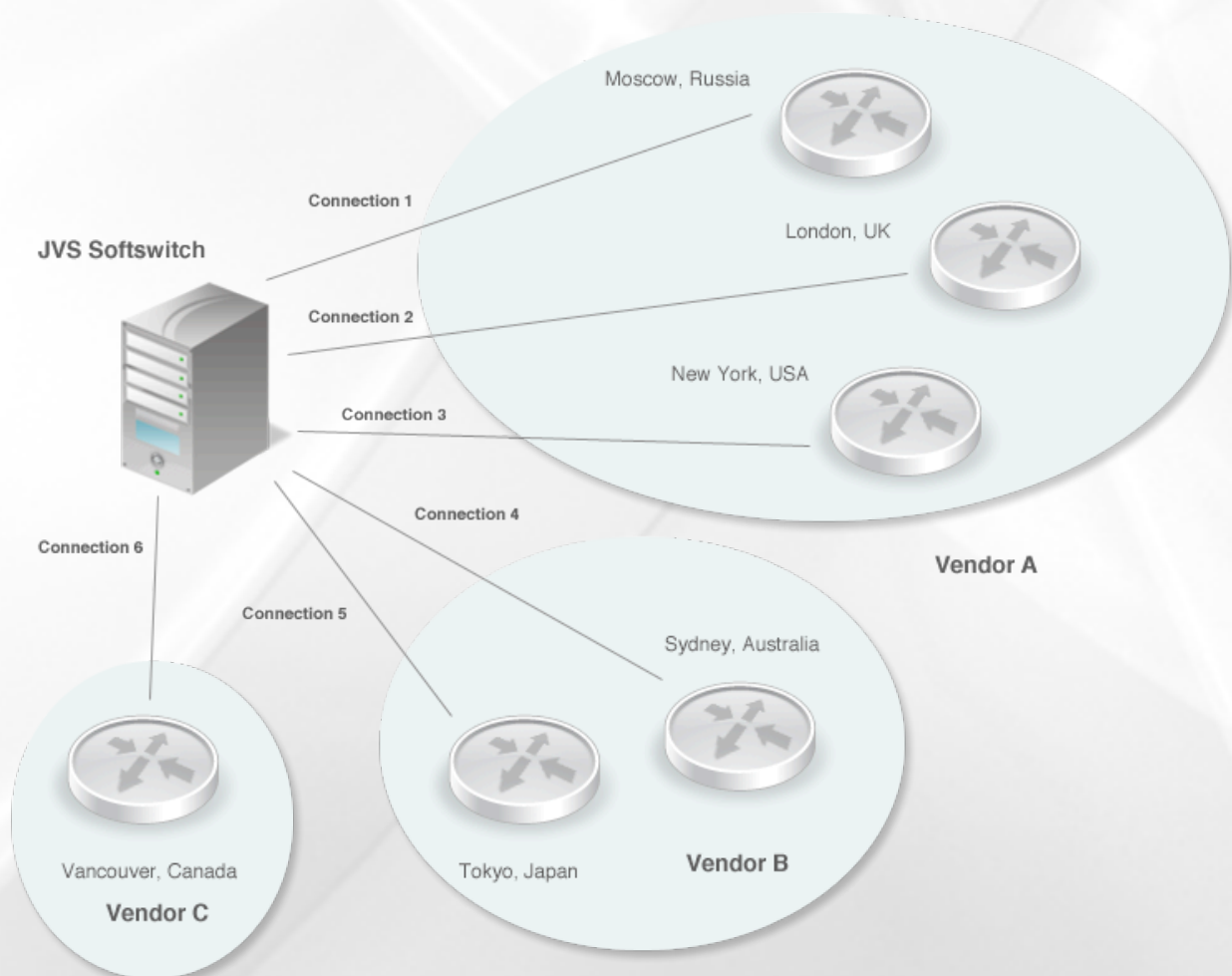
## Connections

Connections are logical entries in the system representing destinations for sending calls to. For example connection can represent specific SIP-PSTN gateway in your network, another SIP softswitch that belong to your termination provider, some SIP IVR or conference server, etc.

## Vendors

Vendors are logical entries in the system representing carries to which calls will be sent for termination. Each Connection belongs to one and only one Vendor, while Vendor can have many Connections that belong to it.

The following figure illustrates this concept:



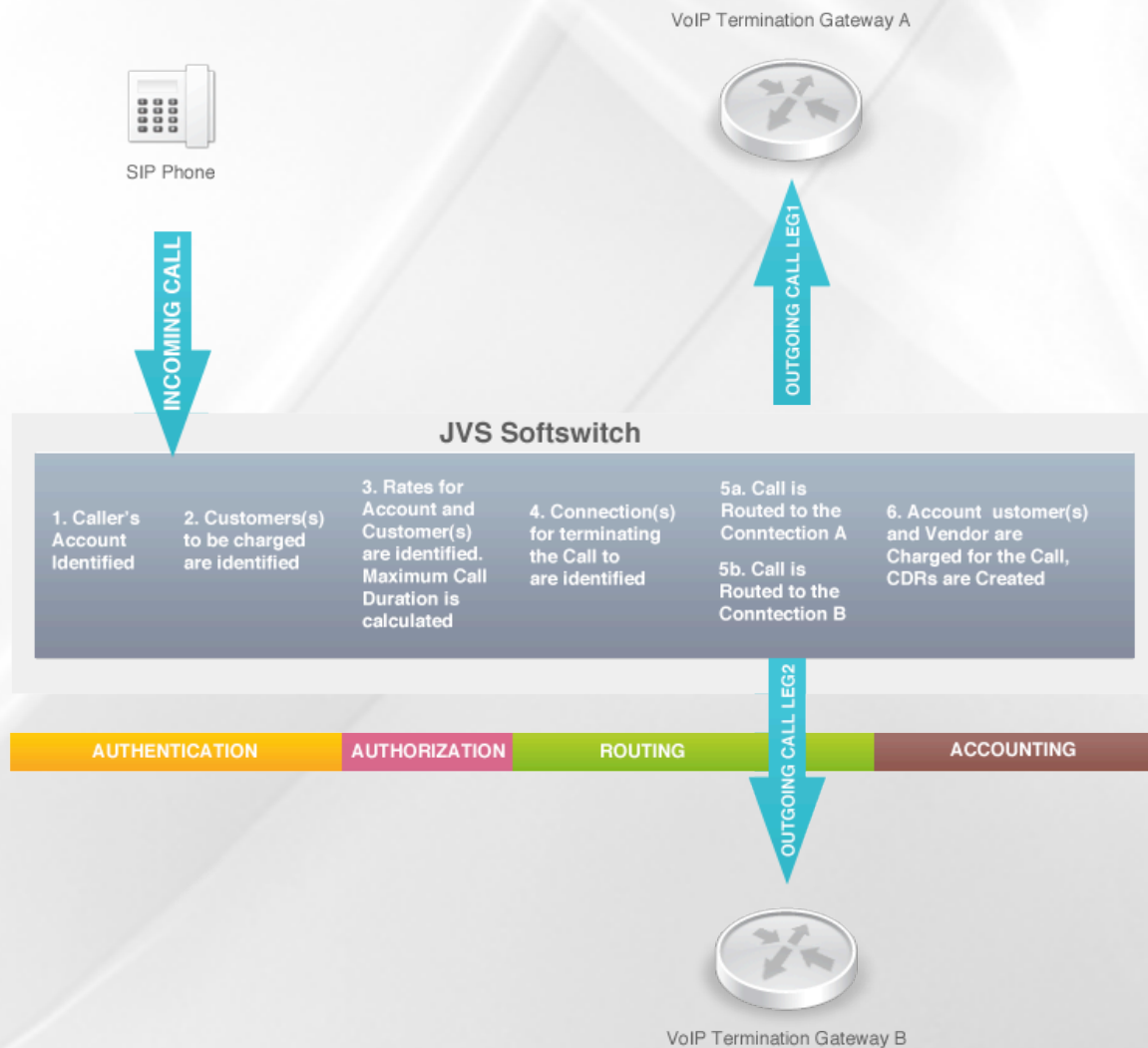
Here you can see system which has 6 connections to 3 different vendors. Each connection represents separate SIP-PSTN gateway located in different places all around the world.

## Call Process

The process starts with the Softswitch receiving incoming call from the originating gateway or any other SIP endpoint. The switch then identifies the caller's Account (1).

This process is called Authentication. If the account cannot be found the call is rejected. Once the Account has been identified the switch enumerates all Customers that need to be charged for the call (2). For Account and each Customer in the list the switch then locates associated Tariff and Rate based on called number and calculates the maximum duration for the call (3). If one or more Rates could not be found or if balance of Account or one of Customers is not sufficient the call is rejected. The next step is Routing – based on called number and routing configuration the switch enumerates one or more Connections for terminating the call to (4). Based on that information it performs actual overflow routing (5) until the call is either connected or there are no more connections to try.

Once the call has been completed, either successfully or unsuccessfully the switch charges Account, Customer(s) and Vendor associated with the call and creates the appropriate CDRs





## Understanding Authentication

Authentication is the process of establishing association between the new incoming call and some particular account in the system. There are two basic methods for performing it in the Softswitch: using secure SIP digest and using Authentication Rules. Those methods will be described in details below. Please note that both methods could be combined together to provide advanced features (see DID **assigning** example below).

### Secure SIP Digest-based Authentication

For Authentication of this type the new call is mapped into particular Account by performing so-called secure SIP digest authentication. In practice that means that the device or software that originates the call has username and password configured in it and those parameters are matched against VoIP Login and VoIP Password parameters of all Accounts until the match is found. Only this type of authentication allows SIP device to register itself with the Softswitch, providing facility necessary for receiving on-net calls. Main application of this Authentication method is providing support for connecting various SIP phones and ATAs to the Softswitch.

### Rules-based Authentication

For Authentication of this type the new call is mapped into particular Account by matching the following three parameters of the new incoming call against one or more Authentication Rules that can be associated with each Account:

- IP address of the device or software that originates the call;
- Calling Number (CLI);
- Destination Number (CLD).

In order for the call to match the rule each of the three parameters of the call should match with appropriate parameters of least one Authentication Rule. When one or more parameters or Rule is empty only remaining parameters are matched against incoming calls.

For example the first rule will match any call originated from the IP address of 1.2.3.4 and having CLI 567890 and any CLD, while the second rule will match calls with CLD of 123456789 and any CLI coming from any IP.

#	Remote IP Address	Incoming CLI/DNIS	Incoming CLD/ANI
1	1.2.3.4	567890	
2			123456789

In addition, wildcard form could be used in the Incoming CLI and Incoming CLD fields by putting asterisk (\*) after the number, which will result in prefix matching.

For example the following modification of the first rule above will match any call originated from the IP address of 1.2.3.4 and having CLI starting with 567890 (e.g. 567890123).

#	Remote IP Address	Incoming CLI/DNIS	Incoming CLD/ANI
3	1.2.3.4	567890*	

When performing the match against several concurrent Rules the Softswitch also calculates so-called weights for each matching Rule in order to determine the best match.

The formula is as follows:

- Matching IP address adds 1.0 to the resulting weight;
- Matching non-wildcard CLD or CLI adds 1.5 to the resulting weight;
- Matching wildcard CLD or CLI adds 1.0 to the resulting weight;
- For any matching wildcard CLI or CLD resulting weight is increased by the length of the matching part divided by the 100.

For the example if the call from IP address of 1.2.3.4, CLI of 567890 and CLD of 123456789 comes the weights would be calculated as follows:

#	Remote IP Address	Incoming CLI	Incoming CLD	Weight
1	1.2.3.4	567890		$1.0+1.5=2.5$
2			123456789	1.5
3	1.2.3.4	567890*		$1.0+1.0+(6/100)=2.06$

As a result, the rule 1 will be selected as the best match.

Please note that when there are multiple rules with the same weight matching and they are belong to different accounts, the call will be rejected due to conflicting authentication rules.

## Typical Applications for Rules-based Authentication

The Rules-based Authentication can accommodate different real-world scenarios. Some of them are listed below.

VoIP Peering. Put IP address of peer's softswitch into the Remote IP Address field.

PSTN origination. Put IP address of gateway into the Remote IP Address field. If there are multiple gateways several rules with different IP addresses could be added. Assigning DID numbers. The DID (Direct Inward Dialing) service could be

implemented by combining secure SIP digest authentication with one or more Authentication Rule that contain DID number coming from the origination gateway as the Incoming CLD field. Optionally, Remote IP

Address could be used to only allow calls coming from a specific gateway to be considered as DID calls. Also, it might be necessary to apply translation rule to ensure proper routing of the call to the registered account.

## Understanding Rating and Billing

Ability to rate calls and bill Accounts, Customers and Vendors for them is a key functionality of the system, so that full understanding of how it works is really important. This chapter explains this subject in depth.

### Tariffs and Rates

Tariff is logical entity in the system, which represents rules for rating and billing calls made by Account or Customer to which this Tariff has been assigned. With each Tariff there can be set of Rates, which define prices to be charged for calls to different destinations. Tariffs

are created by Customers and each Customer has its own set of Tariffs which it can assign to his own sub-Customers or Accounts, invisible to other Customers.

There can be more than one Rate matching destination number, in this case Rate with longest prefix is used (e.g. when destination number is 1234567890, while there are two Rates with prefixes 1 and 12, Rate with prefix 12 will be selected for rating/billing purposes). It is also important to understand that Account or Customer will not be able to make a call to destination for which no Rate exists.

### Balance and Credit Limit

In addition to Tariff, each Customer or Account, except Root Customer, has those two basic rating/billing parameters associated with it – Balance and Credit Limit. Negative Balance represents amount of money Account or Customer “owns” to its owner, while positive one represents amount deposited with the owner. Therefore, with each successful call the Balance goes down by the price of the call. Credit Limit set the maximum negative value the Balance can reach. After its Balance has reached its Credit Limit an Account or Customer will be blocked and not allowed to make any more calls.

### How Calls are Rated and Billed

When the call goes through the Softswitch there are at least 2 entities in the system to be billed for it: Account and one or more Customers. All relevant Accounts and Customers are billed according to rules defined by the Tariff associated with each of them.

Algorithm works as follows:

When the system needs to rate or bill the call first it locates which Account is making it.

After that it finds Tariff for that Account and selects the best matching Rate for the destination number as described in the 3.1. Since there may be nested customers it checks if Customer to which that Account belongs is not a Root Customer and select the best matching Rate according to the respective Tariff for that Customer if not.

It iterates over the customers chain until Root Customer has been reached. This may sound as a complicated process, but in fact it is not. Rates are indexed by the DB server, while in practice really deep nesting is uncommon, so that selection of a best matching one is really quick in majority of cases with only 1-2 iterations.

**NOTE:** In the case when one of links in the chain from the Account to the Root Customer doesn't have a Rate matching the destination number the call will not be allowed to go through, since this would mean that somebody in the chain not pays for this call. When traversing the chain the Softswitch also records the difference between Balance and Credit Limit for each entity and also rejects the call if it finds that for some of them Balance already exceeds respective Credit Limit, thus not allowing customers to overdraw their credits.

This procedure brings up several sets of distinct billing parameters for Account and one or more Customers. Those parameters and their meaning are described in details below.

From Tariff the system obtains the following parameters:

- FreeSeconds - in seconds
- ConnectFee - in money units
- PostCallSurcharge - in percents (0.01 means 1%)
- GracePeriod - in seconds

From Rate the system obtains:

- Price1 - in money units per minute
- PriceN - in money units per minute
- Interval1 - in seconds
- IntervalN - in seconds

The simplest parameters are ConnectFee, which is fixed amount of money charged for each successful call regardless of its duration, and PostCallSurcharge which is additional charge applied, which is calculated as percentage of the amount charged, that is if the call cost is 1 dollar and the PostCallSurcharge is 0.01, the actual amount charged will be 1.01 dollar.

The following picture illustrates how the calls are charged. The process starts with comparing value of the GracePeriod parameter with the duration of the call. The

GracePeriod parameter determines the minimum duration of the call that will be subject of charge. Calls with durations of less than this value are not charged at all. GracePeriod value of 0 second and 1 second provide almost the same behavior except that when the

GracePeriod is 1 second connected calls with zero duration won't be charged a connection fee.

A ConnectFee is charged immediately upon connection, and all calls shorter



than Interval1 will be rounded to Interval1 seconds. FreeSeconds are granted after the Interval1, so this part of the call is not charged, therefore calls shorter than (Interval1 +FreeSeconds) will be

rounded to Interval1 seconds. If call is longer than (Interval1 +FreeSeconds) remaining portion will be rounded up to multiple IntervalN seconds. Afterthat, the PostCallSurcharge is applied to the total amount charged.

The call illustrated in the figure will be charged using the following formula:

## Understanding Routing

Just like Rating and Billing, Routing is another important function of the Softswitch that needs to be well understood. Routing refers to the ability of the system to send calls to different destinations (prefixes) to different Connections. It is also possible to configure the system in such a way that several Connections will be tried in some order until the call is either connected or there are no more Connections to try.

## Connection Parameters

Each connection has several parameters associated with it:

- Destination – IP address or DNS name of the remote SIP device/software for sending outgoing calls to;
- Username – username for the SIP digest authentication if required by the Connection destination;
- Password – password for the SIP digest authentication if required by the Connection destination;
- CLD Translation Rule – translation rule to be applied to the CLD before sending the outgoing call to the Connection destination;
- CLI Translation Rule – translation rule to be applied to the CLI before sending the outgoing call to the Connection destination;
- Capacity – maximum number of simultaneous calls accepted by the Connection destination. Value of -1 means that there is no limit;
- Enforce Capacity – if enabled instructs Softswitch to limit number of simultaneous calls establishes to this particular Connection by the value of the Capacity parameter;
- Routing Class – routing class this Connection belongs to. Only calls from Customer accounts with matching Routing Class parameter will be directed to this connection.

**NOTE:** Routing Class for the Root Customer is assumed to be 0;

- Use Media Relay – policy for using media relay (RTP Proxy) when sending calls to this Connection.

## Routes

Each Connection can have set of prefixes which it can terminate associated with it. All those prefixes are called Routes.

For example if you have service agreement with carrier FooTel, who provides you with access to two his SIP-to-PSTN terminating gateways: one in the Moscow, Russia and the second one in London, UK there will be Vendor in the system representing that carrier and two Connections associated with that vendor. One connection will have Route with prefix 795

and the second with one with 44207 prefix. In addition, if those gateways can terminate calls to other destinations within the same area (mobile, nearby areas etc) there can be other prefixes.

## How the Calls are Routed

When the system needs to route a call it should find set of Connections which can accept it. It does it in two stages: first list of Connections that have at least one Route matching destination number associated with them is built, then the list is filtered so that only one Route, with longest prefix, to each Connection is left. For example when destination number is 1234567890, while there are two Connections having Routes with prefixes 1, 12 and 123, 1234 respectively, the system will select Routes with prefixes 12 and 1234.

**NOTE:** unlike with Rates, where only one best matching Rate has to be selected, the system will select several Routes for the same destination number, but pointing to the different Connections and try them one-by-one until the call is either connected or all possible Connections have been tried.

After the list of possible Routes has been built Softswitch will try to send call to Connections associated with each of those Routes one by one until the call is either successfully connected or there are no more Connections to try. The relative order in which connections associated with routes with the same prefix are tried depends on the parameter Preference associated with each Route. Connections associated with routes with higher Preference will be tried first.

**NOTE:** In the case when the system has selected several Routes with the same Preference the order in which they will be tried is undefined.

Consider the following example: you have two terminating providers, A and B. A can terminate prefixes 1234 and 1345, while B prefixes 1345 and 1456, and prices for termination to prefix 1345 provided by B are lower than prices offered by A. In this case your configuration will look like following:

Connection A Routes		Connection B Routes	
Prefix	Preference	Prefix	Preference
1234	5		
1345	5	1345	6
		1456	5

There are also boolean Huntstop and two different numeric Timeout parameters associated with each Route.

The Huntstop if set instructs the Softswitch to stop hunting routes when Connection associated with such Route has been reached in the process. The failure code will be returned to the caller if attempt to send the call into such Connection will fail.

Two timeout parameters, allow limiting amount of time the Softswitch waits for the call to be answered before sending the call over to the next available Connection.

First timeout, called 1xx timeout, limits amount of time the switch waits for any session progress response from the Connection (typically 180 Ringing or 183 Session Progress), it allows for a detection of non- functional or unreachable Connections. The second timeout, 2xx timeout, limits amount of time the switch will be waiting for the call to be answered before trying to send it to the next available Connection. The timeouts work concurrently, so that whichever happens first “wins”.

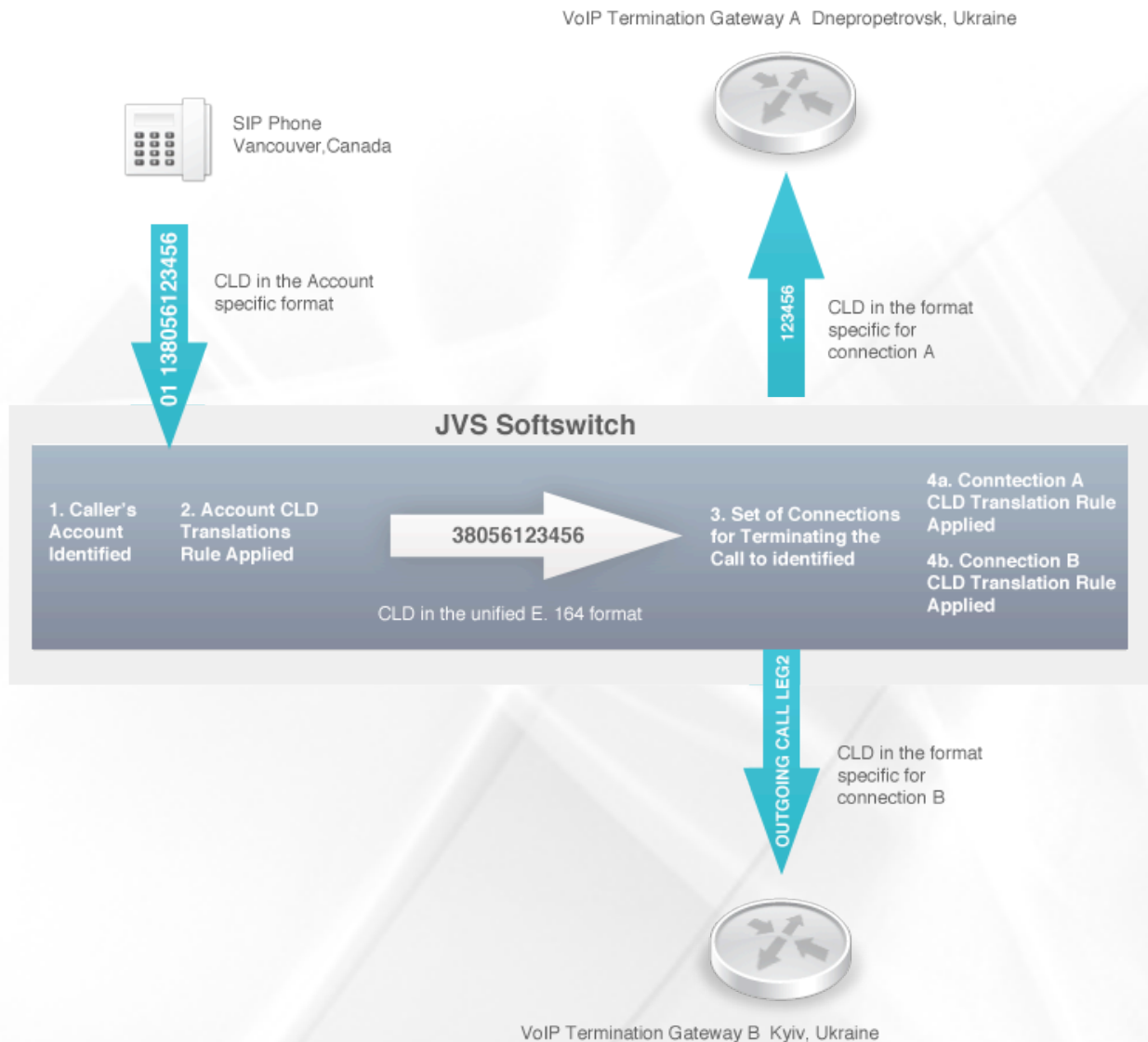
## Practical Role of Number Translation

Number translation plays very important role in today's VoIP world by allowing network with different and often conflicting numbering conventions to communicate. In ideal world there would be only one numbering standard, and everybody would adhere to it, but unfortunately this is not the case in real world.

Even as long as plain old PSTN is considered, almost each country has its own incompatible standards for dialing local, long distance and international numbers. Over the years people got used to it, so that any VoIP provider who wants to provide services to customers in different countries will face the need to somehow compensate for those differences and provide consistent numbering plan in the system, still letting users dial numbers in the way they have used to.

Just like users, many VoIP carriers for various reasons require from their customers to send calls to them using some “weird” numbering plans (country-specific plans, technical prefixes, etc).

To allow dealing with all those real-world scenarios, the Softswitch provides flexible number translation mechanism, whose task is to compensate for difference in dialing habits of individual users and allow adhering with Vendor's requirements. The diagram below illustrates how different CLD number translations are applied during a typical call.



1. The customer dials number in his SIP Phone. He enters the phone number in the same format he uses on conventional phone. The number is delivered to the Softswitch, which performs authentication as a result identifying corresponding Account in the system.
2. In the example above, customer who is physically located in Vancouver, Canada wants to call number 123456 in Dnepropetrovsk, Ukraine.
3. After identifying the calling Account the Softswitch translates number using CLD Translation Rule specific for this account. In the example above this process results in the international dialing prefix 011 to be removed. The CLD becomes 380- 56-123456 – number in the E.164 format.
1. The Softswitch then passes resulting number into routing logic, which builds list of possible destinations (Connections) for terminating the call to. In the example above, two such Connections has been identified – the Connection A and the Connection B.
2. The called number in E.164 format is passed to the routing engine along with the set of Connections that can accept this call along with CLD Translation Rules corresponding to each of those Connections. Routing engine tries to send call to each of those Connections until it is either connected or there are no more Connections to try.



In the example above, in the process of trying to reach the final destination for the call the Softswitch first tries to send it to Connection A, which corresponds to gateway connected to the PSTN in the same area where the destination number is. Therefore, applying CLD Translation Rule for Connection A removes country code and local area code from the E.164 number and it becomes simply 123456 (4a). For some reason, the call is not going through, so that the Softswitch picks up the next Connection, Connection B, from the routing list. The Connection B corresponds to the gateway connected to the PSTN in the same country, but in different area code. Again, before sending the call to that gateway the Softswitch takes CLD in E.164 format and applies corresponding CLD Translation Rule, which in this case removes country code (380) and replaces it with long-distance access code (80). The CLD becomes 80-56-123456.

## How Number Translation Works

The Softswitch is built with assumption that all numbers inside the system are in the E.164 format, that is, country code followed by the area code followed by the local number followed by the optional extension number. However, as previous section explains in the real world the calls will be coming in some user-specific, which needs to be converted into E.164 and also Vendors are likely to expect some different format of CLDs sent to them. For the purpose of doing Account-specific to E.164 and then E.164 to Vendor-specific conversions the system provides three different features allowing translating called or calling number at different stages of the call propagation through the switch. All those rules are specified either as the replacement numbers or as a set of regular expressions with Python syntax in the following two forms:

- s/<match what>/<replace with>/
- s/<match what>/<replace with>/g

The former will match/replace a single matched occurrence or pattern, while the latter – all of them. Several expressions may be used in one translation rule; in this case they need to be separated by semicolon (“;”).

There are several “special” symbols in <match what> which allows to define matching rules, for example ^ matches beginning of the string, while \$ - its end, so that for example ^1 will only match 1 as the first character of the string, while 9\$ will only match 9 as a last character.

For example the following translation rule can be used to strip leading 011 prefix from the number:

- s/^011// While the next rule will result in the number replaced with 123456:
- 123456

For more details on Python regular expressions syntax please check the following link:  
<http://www.python.org/doc/2.4/lib/re-syntax.html>

There are 4 basic types of number translation available in the Softswitch:

- Incoming Called Number Translation

Configured at per-account basis, applied to called number before any call rating and routing decisions are made. Configured via per-Account CLD Translation Rule Account parameter.

- Incoming Calling Number Translation

Configured at per-account basis, applied to calling number before any call rating and routing decisions are made. Configured via two per-Account parameters: CLI and Force CLI. If Force CLI is ON then CLI of all incoming calls matched to that Account will be replaced with value of the CLI parameter.

- Outgoing Called Number Translation

Configured at per-connection basis, applied to called number obtained after applying incoming called number translation rule and after the rating and routing decisions are made but before a call is send to this particular connection. Configured via CLD Translation Rule parameter for each Connection.

- Outgoing Calling Number Translation

Configured at per-connection basis, applied to an original calling number before a call is send to this particular connection. Configured via per-Connection CLI Translation Rule parameter.

## Enabling Account-to-Account Calls (On-Net Calls)

From the point of view of the Softswitch, calls between different Accounts in the system (on-net calls) are not different from any other types of calls, both from the routing and from rating/billing point of view. Therefore, in order to enable such calls it is necessary to create special connection pointing to the system itself and associate one or more routes with it. Then, a call to destination matching those routes will be routed to registered account with matching Account Name. It is also necessary to have a matching rate in the Tariff assigned to caller's Account/Customer, which allows to charge Accounts for on-net calls.

For example, if IP address of the Softswitch system is 1.2.3.4 and there are Accounts in the system with usernames in the form 01XXXXXX, then in order to enable on net calls between them the Connection with destination of 1.2.3.4 has to be created and route with prefix of 01 associated with it. Also, Rate with prefix of 01 should exist in caller's Tariff.

## Miscellaneous System Features

### Interoperability with non-SIP VoIP Networks

The main VoIP protocol for the Softswitch is SIP, but the system provides built-in features allowing to be integrated with legacy (H.323) VoIP networks and proprietary (IAX2) networks if necessary.

The protocol converters are built into the system allowing incoming non-SIP calls to be translated into SIP automatically and then processed by the system as usually.

The support works as follows: for all incoming calls in supported protocols (H.323 and IAX2 currently) the appropriate converter makes signaling/media conversion and sends call converted into SIP to the Softswitch. To allow the Softswitch to authenticate the call, before sending the call to the Softswitch the protocol converter prepends CLD with the IP address of the originating device, so that CLD received by the Softswitch will be: <IP address of the originator>-<Original CLD>.

It is possible then to allow those calls by creating IP Auth Account in the

Softswitch. In order for this account to be accounted for those calls VoIP



Password has to be the IP address of the Softswitch system itself and VoIP Login to be “<IP address of the originator>-” (without quotes). The following figure illustrates the process of receiving H.323 call:

In order to allow the Softswitch to be fully integrated into legacy and proprietary VoIP networks not only should it allow receiving incoming calls in different protocols, but also it should be able sending calls in those protocols as well. For this purpose each protocol converter built into the system is bi-directional, allowing it receive SIP calls from the Softswitch, perform protocol conversion and send it further.

Such feature should be configured as follows: outgoing SIP calls have to be routed to the IP address of the Softswitch system itself, but to the port 5062 and outgoing CLD should be in the form “<protocol name>-<IP address of the target device>-<outgoing CLD>” (without quotes).

Currently, the <protocol name> could be either h323 or iax2. Such outgoing CLD can be created by applying appropriate Connection’s CLD translation rule.

For example, if we want to send some outgoing calls to IAX2 device with IP address of 4.3.2.1, and the IP address of the Softswitch system itself is 5.6.7.8, the Connection has to be added with the following parameters:

- Destination: 5.6.7.8:5062
- CLD translation rule: s/^/iax2-4.3.2.1-/

### Automatic Database Backup

The Softswitch includes the system which allows performing automatic backup of the database periodically. Parameters of this backup, such as schedule, start time,

compression and number of generations to keep, can be configured via the Backup section of the System Parameters interface. The care should be taken when selecting start time for the backup, since even though the backup procedure is on-line one, which means that it doesn't take service offline for its duration, but due to its very nature it creates significant load on the system, which can interfere with call processing. Therefore, off-peak time has to be selected. It should be noted that in a weekly mode the backup will run on the same day of the week as it run the for first time after being enabled. So that for example if you have enabled it on Friday afternoon to run weekly at 4:30am then it will run every Saturday at 4:30am. The similar rule applies for the monthly mode – the backup will run on the same day every month as it run the for first time after being enabled.

By default backup copies of the database are stored in the `/var/backups` directory of the Softswitch system itself. It is advised to configure the server to mount separate physical hard drive into that directory, so that if the main drive fails, the system can be restored from the backup one.

For even better protection, there is an optional mechanism allowing uploading backup files to external remote server over the network automatically as part of the backup procedure. In order to use this mechanism, one or two scripts have to be placed in the `/var/db/pgsql` directory: `ecopy` and `eremove`, each of them taking backup filename as a single argument. The first one is invoked when backup file has been created and needs to be uploaded to the remote server, while the second one – when the backup file has to be removed as part of the rotation procedure. The `ecopy` script should return its error status – exit code 0 means that the upload has completed successfully, while any other code indicates error. Sample `ecopy` script that uploads file to the remote TFTP server is available in the `/home/ssp/scripts/ecopy_tftp.sh` file. The `eremove` script can be absent, in which case some external mechanism for remote backup aging and rotation has to be implemented instead.

### Voicemail Support (optional)

The Softswitch comes with integrated Voicemail module, which provides basic Voicemail functionality for Accounts in the system. There are parts of that feature: one allowing leaving messages to the Account which is not currently available and another one allowing checking own messages.

In order to enable this functionality, two additional Connections have to be added, pointing to the IP address of the Softswitch system itself, but to the port 5062 and outgoing CLD has to be set to `"vm-u<account username>"` for leaving messages to `<account username>` or just `"vm"` for checking own messages.

For the "leave message" Connection the same Routes need to be added as for Account-to-Account calls (see 4.6), but with lower priority to ensure that the system tries to route call to destination Account first and takes the call to the VM if that fails.

For the "check messages" Connection, some easy to remember number(s) has to be added as the Route.

For example, if IP address of the Softswitch system is 1.2.3.4 and there are Accounts in the system with usernames in the form 01XXXXXX and we want to enable Account-to-Account calls with VM using \*98 as a number to check own messages, the following configuration has to be created:



Connection Account-Account		Connection Account-VM		Connection VM-Check	
Destination 1.2.3.4		Destination 1.2.3.4:5062 CLD Tr.: s/^/vm-u/		Destination 1.2.3.4:5062 CLD Tr.: s/.*/vm/	
Routes		Routes		Routes	
Prefix	Prefix	Prefix	Prefix	Prefix	Prefix
1	5	1	4		
				*98	5

### Bulk Account Generation

One of the common tasks facing operators of retail service is generation of large amounts of SIP accounts with secure passwords and optionally PIN codes. In order to help with this task the JOOD-VS Softswitch has a very flexible account generation facility. The account generation is performed in several easy steps:

1. Create template account via Customer's web management interface. Parameters of this account will be shared by all generated account
2. Export template account into the Microsoft Excel document by clicking "Download" button in the account management section.
3. Open document in the Microsoft Excel, locate template record and edit its parameters to indicate which parameters should be auto-generated and set desired ranges for those parameters using special directives. The following directives are supported:

- **\$(N:xxx)** - auto-generate sequence of random digits. The xxx here should be in the format [range1][range2]...[rangeN], where rangeX is replaced with the single digit after generation and should be either list of allowed digits for this position (i.e. [0,4,9]) or range of such digits (i.e. [0-6]). For example: the directive **\$(N: [2,3,4][0- 8][0-9][0-9])** will instruct switch generate numbers in the ranges 2000- 2899, 3000- 3899 and 4000-4899.

- **\$(C:xxx)** - fill in column with values from another column. The xxx here should be number of another column (starting with 1) to copy the value from. For example, if Account Name

column is the second one and VoIP Login is the forth, placing **\$(N:xxx)** directive into the Account Name column and **\$(C:2)** into the VoIP Login will result in the same auto-generated value to be placed into both columns.

- **\$(P:xxx)** - auto-generate secure alphanumeric password phrase. The xxx here should be the required length of the generated password phrase.

The total number of accounts to be generated is defined by placing Gxxx into the Action column, where xxx would be total number of accounts to be generated.

For example the following specification will instruct switch to generate 100 accounts with Account Name in the range 0100-0999 and PIN code in the range 00-99, generate Web Password to be a random alphanumeric string with length of 8 symbols, generate VoIP Login to be the same as Account Name, generate VoIP Password to be a random alphanumeric string with length of 16 symbols and set Balance and Lifetime to be 10.0 and 30 respectively for all generated accounts.

1	2	3	4	5	6	7	8	9
Id	Account Name	Web Password	VoIP Login	VoIP Password	IVR Pin	Balance	Lifetime	Action
	0\${N:[1-9][0-9][0-9]}	\${P:8}	\${C:2}	\${P:16}	\${N:[0-9][0-9]}	10.0	30	G100

**NOTE:** the most of the template account parameters are omitted in the table above for clarity purposes, but they should be present when uploading real specification.

4. Save the Microsoft Excel document and upload it into the accounts management interface.
5. When generation is complete the system prompts you to download Microsoft Excel document that contains parameters for all generated accounts. Please note that generation of large number of accounts could take considerable amount of time, so that it is suggested doing it in batches of 1,000-3,000 accounts to avoid problems due to network outage and/or TCP/IP timeouts.

## System Provisioning

The Softswitch's main configuration method is via its Web provisioning interface. This interface allows setting and changing majority of system parameters. Also, it provides basic CC (customer care) and CRM (customer relations management) functionality for Customers as well as self-care functionality for Customers, Accounts and Vendors, such as for example exploring CDRs (call data records) in real-time.

When user first opens the Softswitch Web provisioning interface she is present with login screen, where she should select the level of access (Account, Customer or Vendor) and enter her credentials (login and password). After user Login button the system checks her credentials and either allows or denies the access. If the access is allowed, the user is presented with the appropriate interface.

There are three different of access and basically different interface for each level. However, the system provides unified layout for all of them, which is illustrated at the diagram below:

The interface at any level of access consists of three basic elements:

1. Status bar at the very top of the window. It displays some basic

information about the currently logged in entity, such as current system time, login level, login name, etc.

2. Menu of available items at the left. User can select appropriate menu by clicking on its name.

3. Currently active menu item in the centre.

In the following sections detailed description of each menu item is provided.

### **Customer Interface**

The purpose of this interface is to allow Customer to do self-provisioning as well as provisioning and monitoring of his Accounts. In addition, if he is Root Customer or has appropriate rights assigned to him by his upper-level customer, he is also allowed to view/modify global system properties, such as routing rules for example.

### **Account Interface**

The purpose of this interface is to allow Account to do self-provisioning.

### **Vendor Interface**

The purpose of this interface is to allow Vendor to do self-provisioning.



