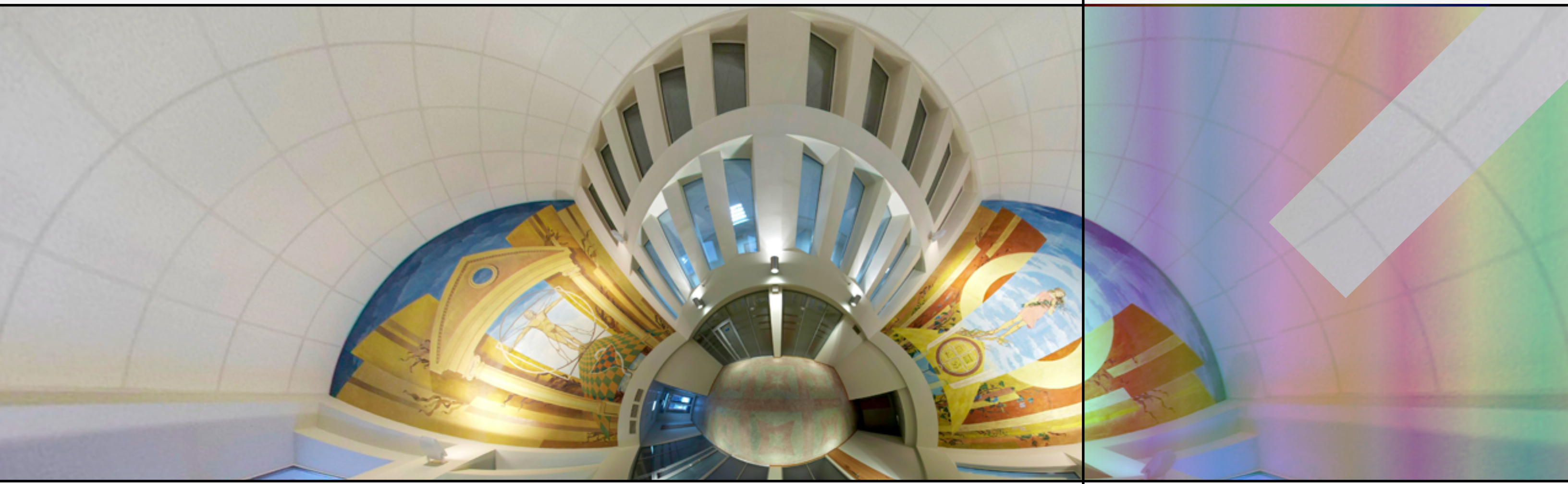
PORTA ONE

Porta Switch®



New Features Guide

Maintenance Release 24

www.portaone.com

Copyright Notice & Disclaimers

PortaSwitch: New Features Guide, September 2011
Maintenance Release 24
V1.24.5

Please address your comments and suggestions to: Sales Department, PortaOne, Inc. Suite #408, 2963 Glen Drive, Coquitlam BC V3B 2P7 Canada.

Table of Contents

Preface

This document describes new features found in PortaSwitch Maintenance Release 24.

Where to get the latest version of this guide

The hard copy of this guide is updated at major releases only and does not always contain the latest material on enhancements introduced between major releases. The online copy of this guide is always up-to-date and integrates the latest changes to the product. You can access the latest copy of this guide at: **www.portaone.com/support/documentation/**.

Conventions

This publication uses the following conventions:

- Commands and keywords are given in **boldface**
- Terminal sessions, console screens, or system file names are displayed in `fixed width font`

The **exclamation mark** draws your attention to important information or actions.

NOTE: Notes contain helpful suggestions about or references to materials not contained in this manual.

Timesaver means that you can save time by performing the action described in the paragraph.

Tips provide information that might help you solve a problem.

Trademarks and Copyrights

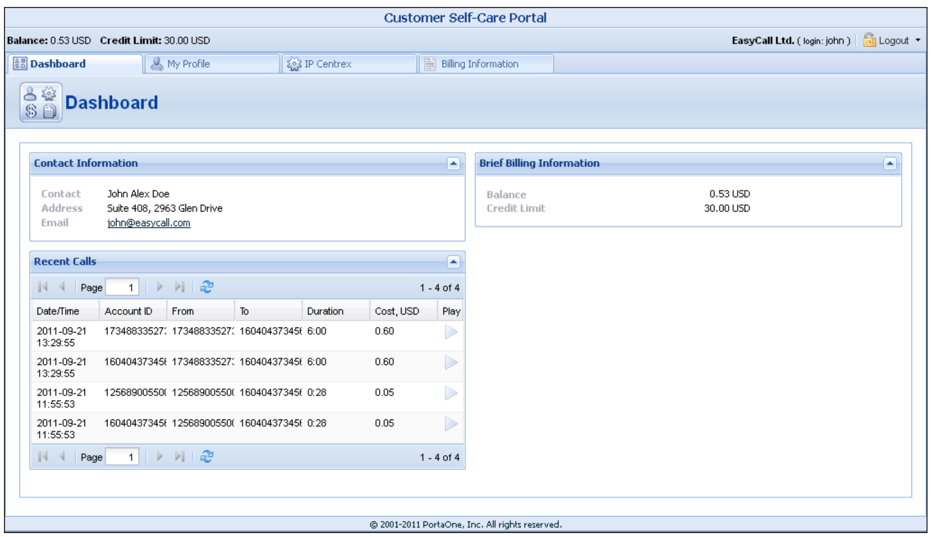
PortaBilling®, PortaSIP®, PortaUM® and PortaSwitch® are registered trademarks of PortaOne, Inc.

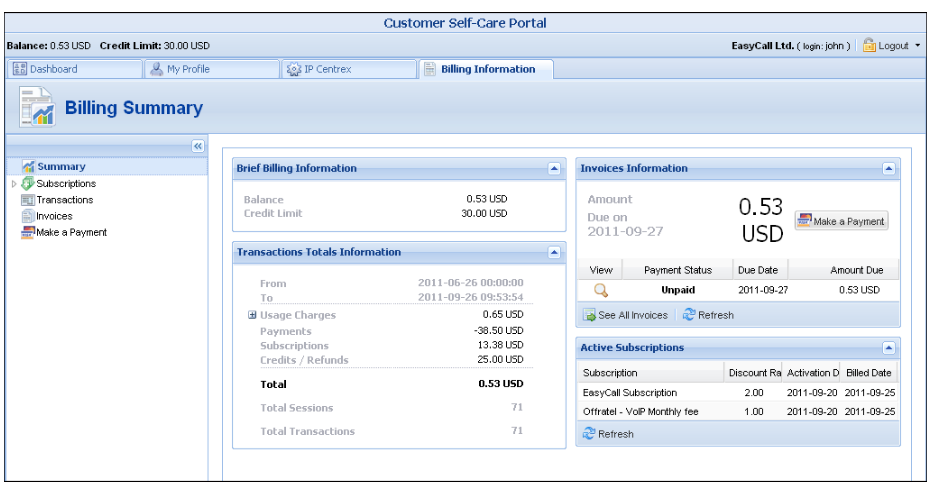
1. New Features in PortaSwitch

This section contains a description of features which involve more than one product and thus apply to PortaSwitch® as a whole.

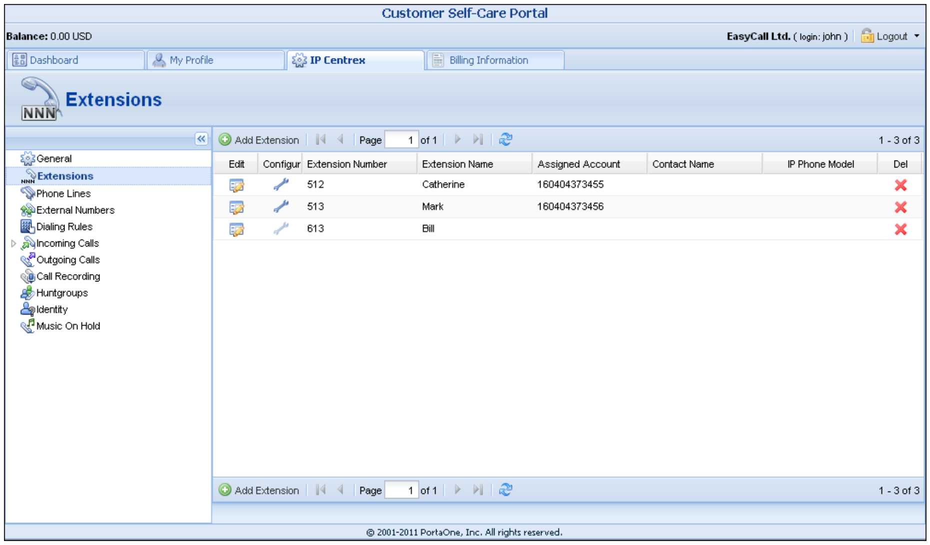
Redesigned Self-care Web Interface

Maintenance Release 24 includes a fully redesigned self-care portal for end-users to access their profile data, check billing information, download invoices and, most importantly, manage their IP Centrex settings. The main focus of this new interface redesign is simplicity and intuitive navigation for the end-user. This includes an easy-to-use structure of menus and controls, graphical icons and improved presentation of information, and a different approach for managing several important components (e.g. hunt-groups).





The new interface allows service providers to significantly reduce their helpdesk workload, since customers will be able to perform virtually all configuration tasks for their hosted IP Centrex environment themselves.

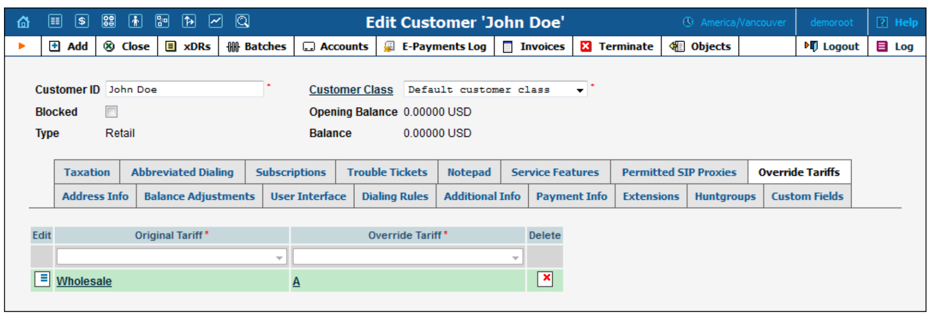

Customer Self-Care Portal
Balance: 0.00 USD
EasyCall Ltd. (login: john)
Logout
Dashboard
My Profile
IP Centrex
Billing Information
Extensions
General
Extensions
Phone Lines
External Numbers
Dialing Rules
Incoming Calls
Outgoing Calls
Call Recording
Huntgroups
Identity
Music On Hold
Add Extension
Page 1 of 1
1 - 3 of 3
Edit
Configur
Extension Number
Extension Name
Assigned Account
Contact Name
IP Phone Model
Del
512
Catherine
160404373455
513
Mark
160404373456
613
Bill
© 2001-2011 PortaOne, Inc. All rights reserved.

2. New Features in PortaBilling

Override Tariff

Often it happens that a particular customer negotiates an individual rate to a specific country, e.g. a company sending a lot of calls to UK mobile requests a reduced rate of $0.09/minute compared to your standard rate of $0.10, defined in the **Wholesale** tariff. A simple way to implement such preferential pricing is to create a volume discount for a specific destination group and assign it to the customer. The only disadvantage is that this discount is applied to the current price of UK mobile in the **Wholesale** tariff. So if the price there changes, the actual price for the customer will change as well. Of course, creating a dedicated tariff (containing rates for all destinations) just for this customer is also possible, but this significantly increases maintenance overhead.

In a situation where a customer requires a specific, pre-negotiated rate per minute for certain destinations, use the **Override Tariff** feature in PortaBilling.



An override tariff is just like a normal tariff, but it contains only a handful of destinations, i.e. only those where the price must differ from the standard one. In our example above, it would be a tariff **A**, containing rates for 447, 448 and so on. The **Override Tariff** feature is then enabled in the configuration for a particular customer, and the administrator can specify that the default tariff should be overridden by a specific tariff (in our case, the **Wholesale** tariff would be overridden by tariff **A**). Now, when calls arrive the billing engine will check for applicable rates in both the master tariff and the override tariff.

- If there is no match in the override tariff (i.e. there is no special price for this destination), then the rating is done as usual, using the rate in the master tariff.
- If a matching rate is found in the override tariff, and the master tariff does not contain a matching rate or the rate in the override tariff is more specific (the destination prefix is longer) than the rate in the master tariff, the rate from the override tariff is used. In our example, this would be $0.09/min from tariff A.

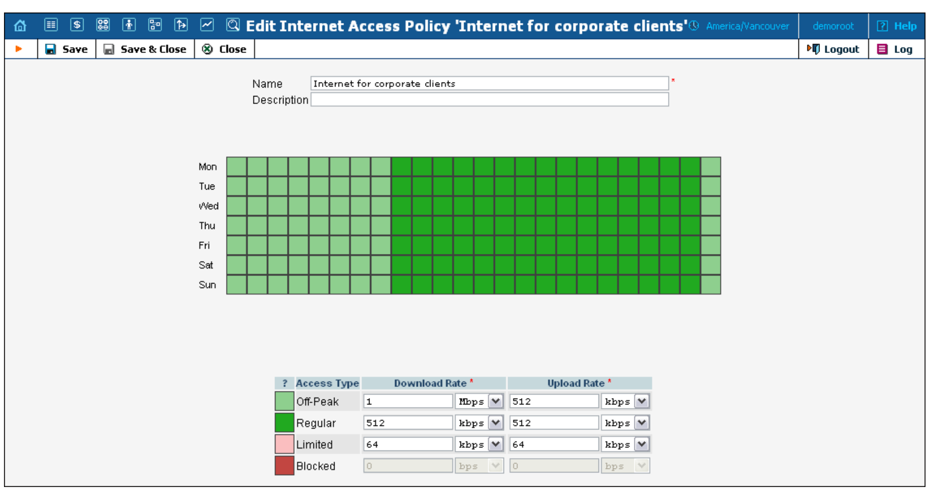
- Finally, if there are matching rates in both the master and override tariffs, but the rate in the master tariff is more specific (the destination prefix length is longer), the override rate is ignored. This is done to prevent a “too generic” rate in the override tariff from erroneously granting a special price for some underlying and more expensive prefixes. In our situation, for instance, this means that if the master tariff contains a rate for 4489 (mobile premium numbers) at $0.30, this is not a part of the special price arrangement, and the rate for 448 in the override tariff will not be applied.

Override tariffs allow the PortaBilling® administrator to easily create a custom rating for a specific customer (either adding new destinations not originally included in his tariff, or applying a special price for existing destinations). Also, it is possible to add multiple override definitions to the “override tariffs” table. This allows different override tariffs to be applied when a customer may be charged by several different master tariffs (e.g. the customer can alternate between the “Premium” and “Cheap” routing plans).

Time-based Internet Access Policies

In order to better oversee network utilization, it is very important for an ISP to control the bandwidth available for specific types of customers throughout the day. This includes, for instance, limiting the amount of bandwidth consumed by residential customers during business hours, thus ensuring that business customers receive adequate service; or creating special products for accessing the Internet at maximum speed during nighttime hours, when the network is underutilized.

PortaBilling allows you to define Internet access service policies, which are then applied to specific products or individual accounts.



There are four available types of Internet access:

- Normal access (default);
- Access during off-peak hours (often referred to as "turbo" access);
- Blocked service (the customer is not able to access the Internet at all);
- Limited access. Typically this is used as an alternative to "blocked" in situations where a customer does not have sufficient funds or failed to pay his last invoice on time. While the customer will not be able to surf the web or download normally, he can still send or receive emails and use the customer self-care portal to submit payment.

Each policy includes:

- A scheduling table, which designates what type of access is used during each time period;
- Specific values for the allowed upload / download speed for each type.

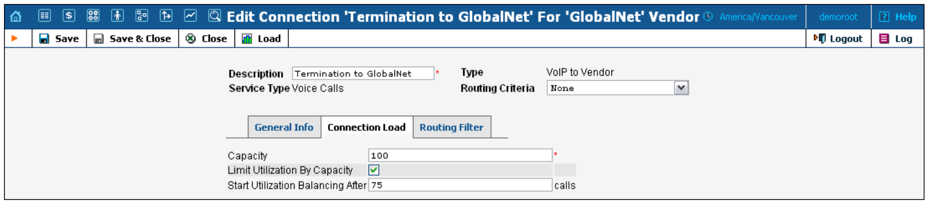
When a new period starts, PortaBilling sends a request to NAS to change the parameters of the current session. This is done using CoA, for equipment which supports it (in this case the bandwidth re-allocation is done transparently to the end user), or POD (packet of disconnect, which terminates the current session).

Routing and Connection Utilization

Aside from the price difference, another important consideration when arranging multiple routes is the number of already connected calls for a particular connection and the maximum number of calls it can potentially handle. Without this, a connection to a carrier may become overloaded, resulting in decreased sound quality and a higher number of failed call attempts, etc., while other carriers with just a slightly higher price may be underutilized.

This is why PortaBilling® now offers you the ability to factor in connection utilization when computing the routing list and perform load balancing among multiple connections accordingly. Load-balancing based on connection utilization may change route precedence even in cases where the price difference between vendors is too high for normal load-balancing (as described earlier).

For each connection the administrator defines two parameters: the **maximum connection capacity** and the **load threshold** (on PortaBilling® web interface it is the **Start Utilization Balancing After** attribute).



As long as the number of connected calls remains below the load threshold, the connection is considered to be operating in normal mode (sufficient available capacity), and calls are routed to this connection based solely on cost and assigned preferences (if the load-balancing threshold is defined, the shuffling of routes with similar prices may also take place).

If the number of simultaneous calls established via the connection exceeds the load threshold, the connection is considered to have surpassed the desired limit and therefore some of the load shifted to other carriers that possess an acceptable price difference. This is done using an algorithm similar to the one used for load-balancing (described in the previous chapter). The administrator sets the **overload handicap** parameter (defined per routing plan) that specifies how far above the desired limit a load may “weigh down” the connection. A portion of this value (proportional to the actual load on the connection) is added to the route cost during route sorting so that the vendor’s cost (whose route was the first ‘unconditional’ route due to its lower price) now becomes comparable with the other vendors’ costs, and load-balancing can commence.

If the connection load increases even further, the added “handicap” value increases as well. The calculated vendor’s cost (that is used during sorting) would rise higher and higher, leaving less chance for this vendor to be the first route.

Finally, when the quantity of calls in progress reaches the maximum connection capacity threshold, this connection becomes excluded from the routing process; no further calls will be routed there until some already connected calls are disconnected.

Let’s extend the example used for load-balancing to consider connection utilization. There are three vendors:

Vendor	Cost	Maximum Capacity	Load Threshold
A	$0.05	100	75
B	$0.07	250	150
C	$0.10	150	100

The routing plan set up is as follows:

Parameter	Value
Load-balancing threshold	0.01
Overload handicap	0.05

Initially, when the load across all connections is minimal, the routing order is determined only by the vendor's cost. Since the difference in cost is more than the allowed load-balancing threshold ($0.01), the routing order will always be Vendor A, Vendor B and then Vendor C.

How does the situation change when some connections surpass the desired load limit?

Vendor	Cost	Max. Capacity	Load Threshold	Current Load	Handicap	Adjusted Cost
A	0.05	100	75	83	0.016	0.066
B	0.07	250	150	30	0	0.07
C	0.10	150	100	5	0	0.10

As you can see, Vendor A has exceeded the desired load capacity (75, specified by the load threshold) by 32%. This is calculated as the ratio of total number of calls above the desired capacity against the difference between desired and maximum capacity, in this case (83 – 75) / (100 – 75). So 32% of the overload handicap ($0.05 * 32% = $0.016) is added to Vendor A's cost and the adjusted cost becomes $0.066. Since the difference between Vendor A's and B's costs is less than the load balancing threshold, load-balancing will commence and some percentage of calls will be routed to Vendor B. Let's continue the example and assume that Vendor A's load has increased even further:

Vendor	Cost	Max. Capacity	Load Threshold	Current Load	Handicap	Adjusted Cost
A	0.05	100	75	90	0.03	0.08
B	0.07	250	150	155	0.0025	0.0725
C	0.10	150	100	10	0	0.10

Since Vendor A's load is approaching the maximum allowed limit, they are penalized even further – now with an adjusted cost of $0.08. Vendor B's cost is also starting to be adjusted. Since it is only 5 calls above the load threshold, it is quite low compared to the maximum potential number of calls above the load threshold, which is 250 – 150 = 100 calls. So the overutilization is 5 / 100 = 5%, and thus the handicap effect is minimal ($0.05 * 5% = $0.0025). Now Vendor B will actually become the first route for the majority of calls, since its adjusted cost ($0.0725) is lower than Vendor A's ($0.08).

Vendor	Cost	Max. Capacity	Load Threshold	Current Load	Handicap	Adjusted Cost
A	0.05	100	75	98	0.046	0.096

B	0.07	250	150	200	0.025	0.0925
C	0.10	150	100	20	0	0.10

When the load on both Vendors A and B increases, their adjusted costs increase to the point that Vendor C also begins to participate in the load-balancing.

Vendor	Cost	Max. Capacity	Load Threshold	Current Load	Handicap	Adjusted Cost
A	0.05	100	75	100	0.046	N/A
B	0.07	250	150	230	0.04	0.11
C	0.10	150	100	20	0	0.10

And finally, when the load on Vendor A's connection reaches the maximum allowed limit, this connection will be excluded from call routing. All further call attempts will be routed to Vendors C and B (with C becoming the first route because of its lower adjusted cost).

3. New Features in PortaSIP

Routing Filters

In order for a voice call to be established, the two end-points (IP phones or media gateways) must not only be able to exchange IP packets which contain SIP messages, but also agree on a mutually acceptable way to encode the audio signal for transmission over the IP network (codec). There are many codecs available, with different features in terms of voice quality, compression rate and required processing resources. Some are free, while others require royalty payments. As a result, each device, such as an IP phone, is usually capable of supporting only the limited subset of codecs implemented by the manufacturer.

Normally, at the beginning of a call the calling party announces all the codecs it supports, and then the called party replies back with a list of codecs it is willing to accept, and so the decision is made by the two end-points only. This approach provides great flexibility and, since PortaSwitch does not have to interfere in the audio processing and utilize any codecs on its side, it allows PortaOne to provide you with an unlimited license, without your being responsible for any additional codec royalties. But since the two SIP end-points make the decision regarding the choice of the codec without any consideration of the network infrastructure or other important factors, in some situations their choice may be less than optimal. For instance, SIP phone A may have the G.711 codec as the first preference by default; and if that codec is supported by the other party, it will be chosen for the call. While this is great for a customer A, with high-speed broadband connectivity (G.711 provides sound quality identical to traditional "wired" telephony service, such as ISDN), if customer B attempts to use G.711 with limited bandwidth it will result in severely degraded voice quality and a negative customer experience. Such a customer B should always use a narrow-bandwidth codec such as G.729 to ensure good sound quality.

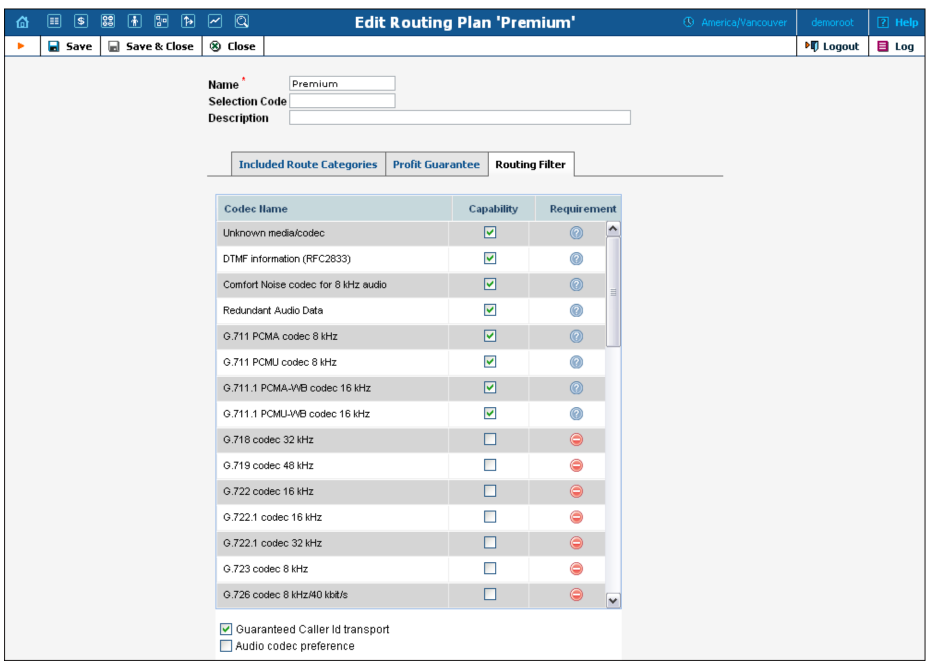
Thus it becomes increasingly important for the ITSP to actively control the choice of codecs used by the end-points, in order to optimize network performance and avoid negative customer experiences. How is it done?

Routing filters operate with the concept of **call media features**. A call media feature is a property of the call or call media, such as a specific codec, T.38 fax, or the ability to guarantee delivery of the correct CLI (caller identification) to the recipient of the call. Since the caller may have his own set of desired call media features, the main idea is to ensure proper "match-making" between the available carriers, while limiting the caller's choice if required (e.g. the caller may request a video call, but this will be prohibited if he is not authorized to do so).

Call Media Regulations

These describe the filters applied to call media features (such as a specific codec or T.38 fax capability), as requested by the calling party. For each feature the PortaSwitch administrator can specify that:

- It is "required" – meaning that the other SIP end-point must have this feature supported in order for the call to be completed. For instance, if the "G.729 codec" feature is marked as "required" for an account making a phone call, then only those vendors specifically marked as "guaranteed to support G.729" will be placed in the routing list.
- It is "suppressed" – meaning that PortaSwitch will prevent the use of this particular feature (e.g. G.722 codec) and will not even show the information about this codec in the SIP request when sending an outgoing call to the other end-point.
- It is "not required" – meaning that PortaSwitch does not do any special processing for this feature. It will be included in the outgoing SIP request, and may be used if the other party supports it. This is the default value for any feature.

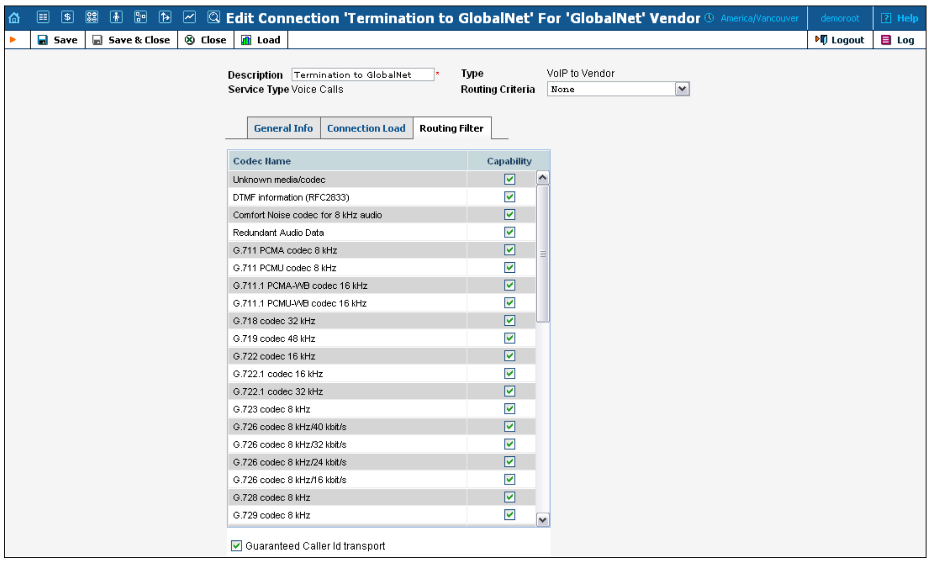


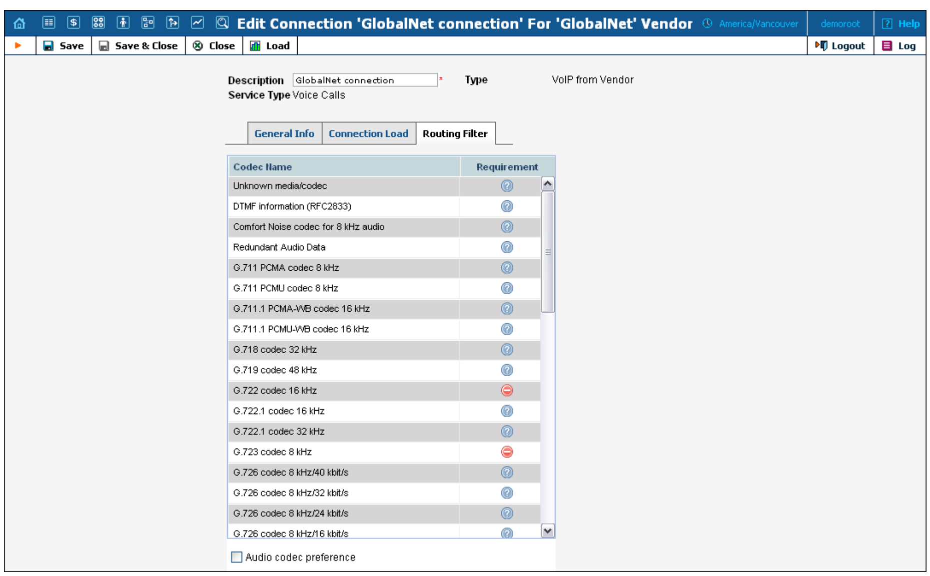
Call Media Capabilities

These describe the capabilities of the remote party (such as the gateway of a carrier) and our preferences on using them. For each feature it is specified whether it is:

- Supported – meaning that we know for sure that this equipment supports this feature and are willing to use it.

- Not supported – meaning that this equipment is unable to support this particular feature (e.g. G.723 codec). It could also be our administrator's decision to prohibit it. For example, although we do not know whether a vendor's gateway supports the G.722 codec, by marking it as "not supported" we will ensure that, even if the originating end-point shows this codec as available, it will be removed from the codec list sent to the carrier in the SIP call initiation request, and thus never used.





4. New Features in PortaUM

New Language Support

As of Maintenance Release 24, PortaUM supports German and Italian languages for all IVR applications such as voicemail management, account self-care and prepaid card calling.

ANI authentication of calls originating from the PSTN network

You may want to offer services such as voicemail and self-care for outside users calling your network. Usually you would configure a special access number that customers can call and then identify themselves to using their PIN. But there are several drawbacks to this method:

- ANI authentication for calls made from the PSTN network to your access number requires an account with an Account ID in E164 format (identical to the PSTN number). But you *should not* create such accounts in your system. The reason is that when someone in your network dials this number, the system will recognize that the call should be delivered to one of the IP phones connected to PortaSwitch® – and will therefore never be sent out to the PSTN.
- You may want to provide customers with alphanumeric accounts that cannot be entered as PIN numbers.

Since MR24, the system is now able to authenticate calls coming into your system from vendors by using an ANI number appended with a certain prefix, by default: "<ANI>@pstn" (e.g. 44203123123@pstn). To allow your customers to authenticate themselves by using an ANI while calling to UM access numbers, assign accounts (or aliases) to them by using the format described above.

Premium Numbers IVR

Providing prepaid calling card service implicates the management, printing and distribution of cards or top-up vouchers. The Premium Numbers IVR application enables ITSP to offer cheap international calls to local telecom customers without administrative overhead. It mainly provides a special access number that any customer can reach from PSTN (via landline or mobile phone). Local telecom then delivers calls made to this access number to ITSP's network and delivers the payment (i.e. the telecom collects payments from customers and delivers payment to ITSP).

Incoming calls are forwarded to PortaUM where the Premium Numbers IVR application is launched. IVR prompts for the destination and if allowed by the configuration, PortaBilling forwards the call to CLD according to the routing plan.

The Premium Numbers IVR should be configured to allow calls to profitable destinations only (where the price that the telecom pays to ITSP is higher than call termination costs). Several Premium Numbers applications can be configured and each will have its own access number with a different rate and allowable destinations, respectively.

If a telecom customer wants to call destinations that are not allowed by any Premium Numbers application he still can do this by registering an account (for ANI authentication) in the ITSP's system and transferring a part of his telecom balance to this account. This is done by sending an SMS with a certain command to a premium number. Once registration is completed, the client can use the transferred amount for international calls via a standard ANI-based calling card IVR. If the account's balance falls below a specified threshold, the ITSP system initiates an account recharge by transferring a preconfigured amount from the customer's telecom balance.

DISA functionality for One's own voice mailbox access application

When a customer accesses an IVR application to check his voicemail he may also want to make an outgoing call right away (e.g. to call back the person who left the voicemail). Now the customer can make an outgoing call from the voice mailbox by simply choosing that option from the application menu.