

Porta**Switch**

New Features Guide



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Preface

PortaSwitch® Maintenance Release 66 is the next long-life release which is mainly focused on improved system stability. It is supported with bug fixes, contains minor improvements and offers other software support for an extended period of time, thereby enabling customers to better plan the evolution of their PortaSwitch® systems.

Where to get the latest version of this guide

The hard copy of this guide is updated upon 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 actions that must be taken for proper configuration.

 $\ensuremath{\textbf{NOTE}}$: Notes contain additional information to supplement or accentuate important points in the text.

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



 \mathbf{X}

Archivist explains how the feature worked in previous releases.

Gear points out that this feature must be enabled on the Configuration server.



Tips provide information that might help you solve a problem.

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New PortaBilling® Web Interface

The first phase of PortaBilling®'s new web interface modification is now complete. Its newly redesigned layout facilitates navigation and makes PortaBilling® management more user-friendly and more intuitive.

The key features of the new web interface are:

- Simplified page navigation by organizing the contents into coherent groups.
- Redesigned ACL structure.
- Notifications about operations in progress and background tasks.
- Quick access to statistics information via dashboard widgets.
- Simplified troubleshooting.
- Permalinks for quick access to a given entity.
- Faster response time.

The **Dashboard** page is the point of entry to all services in PortaBilling®. Its redesigned layout provides the administrator with an overview of the service flow status within the system in the form of widgets. These widgets reflect customer and inventory status, a SIP status chart and traffic exchange statistics so that you can edit your dashboard view according to your needs.

The page navigation has been redesigned. Now pages display the general parameters of an entity and then the remaining information is organized into coherent groups containing multiple levels (e.g. customer information is grouped by Personal info, Accounts, Services, Finances, xDRs and Fraud protection sections). The panel-based page layout displays maximum information about a given entity. The sliding help panel provides additional guidelines on using a particular option in your configuration. All this therefore simplifies page orientation and makes it more intuitive.

The navigation menu to the left is divided into six expandable sections plus the **Favorites** section that serves for quick access to the most popular entities. The tree-like multi-level structure of the menu simplifies the search and enables quick access to the specific page. When on the page, the navigation menu is hidden though easily invoked by clicking the **Favorites** icon.

The web interface loading logic has been enhanced, too. Now instead of reloading a page with each change, the system sends an API request to load only the updated component. This results in faster response time (e.g. the Edit customer page on the new web interface loads 3 times faster than the same page on the old web interface).

For backward compatibility, the old web interface remains available and can be accessed with just a single mouse click. Pages not yet available on the new web interface can be accessed on the old one via the **Old UI** icon.

The new web interface is ready to use and is activated as the default one beginning with MR65-1 and MR66-1.

Improved Trace Session

With this release, PortaOne introduces a new approach for tracing user sessions that simplifies troubleshooting for administrators.

The advanced **Trace session** functionality has the following key features:

- A single page for session search and display;
- Merged SIP and billing logs to provide an entire overview of the session flow;
- General information about a session in the form of a diagram between the main participants (e.g. SIP phones and PortaSwitch®) with the issue description and troubleshooting instructions for the most common issues; and
- The ability to switch to the advanced log view if deeper investigation is required (e.g. to review the call flow among nodes within the PortaSwitch® cluster).

For instance, on a call with an administrator, John Doe (12103355787) wants to find out why he cannot call 18660333777. The administrator opens the trace session page and searches the session using the account number and time interval. On the log viewer page, he sees that John Doe is not allowed to call the destination because it is forbidden.



The new Trace session functionality is available on the new web interface, beginning from MR65-1 and MR66-0.

The improved trace session functionality drastically simplifies troubleshooting and reduces the load on your administrative staff.

Call Identification in Mobile Centrex

Mobile Centrex is a service that provides customary PBX features but allows the use of mobile phones instead of IP phones.

When end users answer calls on their mobile phones, they must be able to distinguish direct incoming calls from calls received via huntgroups / call queues so that they answer them differently.

For example, one may answer a call received via a huntgroup / call queue with, "Good morning! EasyCall sales, John Doe speaking. How may I help you?" But if a colleague dials directly to an extension, the answer might be, "Sales, John Doe speaking."

An administrator configures dialing rules that modify CLIs for calls that are passing through a huntgroup / call queue as follows: 00<CLI>. PortaBilling® sends modified CLIs to the mobile core. When end users

see 00, they understand that a call comes via a huntgroup or a call queue while direct calls are displayed as +<CLI>.

NOTE: If calls are received via a huntgroup / call queue and forwarded further (e.g. through Follow-me), their CLIs are not modified.

Missed calls can be distinguished the same way. End users can call back the originating number of a missed call by pressing the Redial key.

This allows Mobile Centrex users to see whether calls are either direct or routed from a huntgroup / call queue and treat those calls accordingly.

Call Disconnect upon Unsuccessful Blind Transfer

By default, if a blind transfer is unsuccessful (e.g. party A calls party B, B transfers the call to party C and C does not answer), PortaSIP® restores the call between A and B. However, some SIP phones do not restore calls between A and B when an unsuccessful blind transfer occurs (e.g. Gigaset), and some SIP phones do not disconnect B from the call unless C answers (e.g. Snom).

In cases where providers follow a BYOD (bring your own device) policy, a discrepancy may appear. For example, some calls are restored when an unsuccessful blind transfer occurs while others are not. This may lead to dissatisfaction among customers and will likely put an additional load on administrators when they troubleshoot an issue.

For that reason, administrators can now configure PortaSwitch® to disconnect calls when unsuccessful blind transfer occurs either for all of their customers or for only those customers who use a particular type of SIP phone.

For instance, let's say that A calls B and at some moment in the conversation B makes a blind transfer to C. When PortaSIP® starts to process the call, B is disconnected from the call and C hears ringing. C is busy so he rejects the call. A is disconnected and the call finishes.

To configure a call disconnect upon an unsuccessful blind transfer, create a dynamic service policy as follows:

- Specify the policy name and its ID.
- Specify the matching pattern (a % wildcard for all your customers or name% (e.g. **Snorm%**) for a particular type of device) in the **SIP End-point Pattern** field.

• Set the **Yes** value for the **transfer_disable_recovery** field and save the policy.

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Consequently, when B transfers the call to C, PortaSIP® checks for the service policy. If no policy is assigned for B's account or for the connection, PortaSIP® applies the dynamic service policy and disconnects B from the call.

This feature allows service providers to unify blind transfer flow for all their customers regardless of the equipment that their customers use.

Configure Auto Attendant to Disconnect Calls

Sometimes customers request that calls to their auto attendant be disconnected during non-working hours. For example: "TeenMuz," a local musical radio show for teenagers, has a separate line for callers to share their ideas or request a song while the show is live.

When the show is over and someone tries to get through, the caller hears: "The show is over. Please call back tomorrow from 10:00 a.m.–12:00 a.m." and then the call disconnects.

With this release, IP Centrex customers can configure their auto attendant to play a prompt and disconnect a call when the menu is inactive or when the caller presses / does not press a key.

To configure the auto attendant to disconnect calls when the menu is inactive, a customer adjusts the auto attendant configuration on his selfcare interface as follows:

- 1. On the Auto Attendant page of the IP Centrex tab go to the ROOT menu.
- 2. For the **When the menu is inactive, then:** option, select **Disconnect Call**.
- 3. Check the **Play Before Action** box.
- 4. Upload or record a prompt to be played for the **Upload/Record a prompt** option and save the changes.

Base Edit Phone Line: 12125557815				8 ×
Dashboard	🐣 My Profile	🙀 IP Centrex	Billing Information	
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	C General Promp	ts Actions Input Errors		
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NOTE: If a customer does not enable the **Play Before Action** option for the **Disconnect Call** action, the system will disconnect the call immediately after it has connected.

To configure the auto attendant to disconnect calls based on user input, a customer adds a new or adjusts the existing menu action. In our example we assume that the call must be disconnected if callers do not dial any key after they have listened to all menu options:

On the Actions tab of the necessary menu (e.g. ROOT) select:

- User Input No input
- Timeout -9
- Action Disconnect Call.
- Play Before Action checked
- **Play/Record Prompt** the prompt to be played.



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This feature provides additional flexibility for system management and increases customer satisfaction.

Diversion SIP Header Removal for Multiple Forwarding

When a call is forwarded multiple times (e.g. Alice forwards a call to Bob, Bob forwards the call to Carol), a new diversion header is added with each forward.

Some vendor equipment is unable to process INVITE requests correctly when they include multiple diversion headers. As a result, calls forwarded multiple times are not connected.

Therefore, with this release, PortaSwitch® allows the removal of all but the last diversion header from INVITE requests.

For example, an administrator configures a service policy for an outgoing connection that instructs PortaSwitch® to preserve only one (added last) diversion header. When multiple call forwarding occurs, PortaSwitch® removes all but the last diversion header from the INVITE requests sent to this connection.

This ensures that vendors properly handle multiple call forwarding scenarios.

Call Redirect via Call Control API

With this release, call control API has been extended with the ability to redirect a user's incoming calls. This permits the emulation of endpoint redirection functionality for applications, so that calls are redirected to a predefined number – as if from an IP phone.

This functionality can be of special interest to call centers that use external applications to monitor and control incoming calls to their agents. The external application can track an agent's status (e.g. available, busy, etc.) and send the redirect command if necessary. When PortaSwitch® receives the application's command, it redirects the agent's incoming calls to another phone number (e.g. a call queue number).

To configure call redirect, the administrator must enable the **Endpoint** redirection option for the customer on the Incoming calls configuration panel.

For instance, in the call center application, Mary Smith changes her agent status to "Occupied." Therefore, the application is configured to redirect all incoming calls (except from Mark Johnson, CTO) to the Sales department's call queue. When John Doe dials Mary Smith's direct number, his call is placed in the queue. But when Mark Johnson tries to reach her, Mary receives the call.

Charges for call redirection always apply to the called party – the call redirection initiator.

This functionality allows service providers to expand CTI (Computertelephony integration) technology and provide call centers with a solid application. Therefore, call centers can manage incoming calls with efficiency and improve their customers' satisfaction.

API for Notification about New Call Events

An operator must have a clear overview of all calls to efficiently manage a call queue. The operator, via an external application, must see the calls that are waiting and those that are already connected with agents or left the queue. With this release, the call control API includes new methods that enable an operator to receive notifications regarding call events.

When a new incoming call is placed in a queue, the system sends the **queued** event. An operator can then easily manage the queued call (e.g. answer it or transfer it to a free agent). Once the call is sent to an agent or the caller leaves the queue, the system sends a **dequeued** event. In case a caller leaves the queue (e.g. there are no free agents for a long time so the

caller hangs up), a **terminated** event follows the **dequeued** event immediately.

An operator must subscribe to notifications for the auto attendant in order to manage effectively.

Consider the following example:

John Doe (1212555123) wants to reach the sales department of Safari Telecom. He dials 1210502050 (an Auto Attendant number), listens to the company prompt and is informed that he is 2nd in the queue. A Safari Telecom operator then receives notification that 1212555123 has been placed in a queue with the **queued** event. Once a free sales manager takes the call, the operator receives another notification showing the **dequeued** event.

With this solution, an operator has an overall statistic for calls connected with agents and those waiting in a queue. This simplifies the management for call queues and brings CTI (Computer-telephony integration) technology to the next level.

RADIUS Proxy for Dual-Version PortaBilling®

To ensure uninterrupted service provisioning during gradual customer migration, with this release we are introducing the RADIUS proxy function.

The RADIUS proxy is deployed on the new system. It receives RADIUS requests for accounts to be charged for a session. It also detects to which system (the main or new one) accounts belong. The RADIUS proxy then sends the requests to that specific system.

When a RADIUS server processes the request, the response is sent back to the RADIUS proxy. The RADIUS proxy relays the response to the NAS.

This allows external RADIUS clients (e.g. NAS) to function seamlessly with dual-version PortaBilling[®]. It also allows the xDR mediation process to run on either system so it can perform file extraction and parsing yet put the charging data records into the system where the account currently resides.

Support of Number Porting Requests in Dual-Version PortaSwitch®

Number porting is widely used within the telecommunications market as it allows users to keep their numbers when switching service providers.

For service providers performing gradual and controlled customer migration within dual-version PortaSwitch®, it is essential that number porting requests are properly handled.

With this release, number porting requests are distributed within dualversion PortaSwitch® based on the current location of the customer record. Thus, all incoming requests from Neustar arrive to the main system, which either processes the requests itself or passes them to the new system. This ensures uninterrupted service provisioning and facilitates smooth customer migration.

Let's consider how request distribution works for port-in and port-out requests.

Port-out requests

Port-out requests are created automatically, without the administrator's involvement. So when a user wishes to port their number from PortaBilling® to another provider, the main system receives the port-out request from Neustar. The main system finds the customer's account record and to which system it belongs. If the customer has been moved to the new system, it passes the number porting request there. The new system terminates the customer's account record and informs Neustar about its number availability. Neustar confirms the request completion to the main system, which passes it to the new one.

If the customer's account is within the main system, the whole porting out procedure runs from there.

Port-in requests

When porting in a user's number, the administrator defines the system where this number will be provisioned. Then the respective system interacts with Neustar to complete the request.

Consider the following example:

John Doe, a former customer of EasyCall Ltd., signs up for HappyTraffic services and wants to keep his number 18185557852. The administrator creates the account 120522254200 for John in the new system and initiates a port-in request to Neustar in the new system's PortaBilling®.

After Neustar confirms the request, 120522254200 is replaced with 18185557852 and John starts using the services of HappyTraffic.

The distribution of number porting requests across systems within dualversion PortaSwitch® facilitates gradual customer migration and ensures the uninterrupted operation of number porting functionality for users, regardless of their location.

Load Balancing Within Sites Using Dispatching SBC

With this release, you can define the ratio by which the dispatching SBC will distribute requests across the sites of your geo-redundant PortaSwitch®. Thereby you gain advanced flexibility in balancing the load within your network.

For this, define the percentage of requests to be delivered to a particular PortaSIP® in the **load_distribution** option on the Configuration server.

Servers Sites Configurations () Environments	Users Update 9		🔝 • porta-support •
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Configuration '77_cloned_on_update_cloned_on_provision	-rev.2' information		×
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sp cluster-1@performance	sip_ports_tcp*	5060	0
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S Configurator	sip_ports_udp*	5060	0
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- iiii porta-billing-web-1@configurator	🐵 😐 vipaddr		
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Task monitor			

The default value "auto" means that the dispatching SBC will distribute requests equally among sites (e.g. the main and three secondary sites will each handle 25% of all requests).

During the system update, all requests to the site being maintained are evenly distributed among the remaining active sites.

Let's say the main site of your installation handles 50% of all requests and secondary sites A and B handle 30% and 20% of requests, respectively. When the main site is being updated the dispatching SBC will deliver 55% of requests to secondary site A and 45% of all requests to secondary site B. When secondary site A is being maintained, the request distribution is the following: the main site processes 65% of requests while 35% of requests will go to secondary site B.

This enhancement optimizes system management and enables administrators to utilize system capacity in the most efficient way.

Different Site Types for Redundancy Management

Geo-redundant PortaSwitch® provides a high-availability system so that services are uninterrupted, even in case of the main site's outage. In this configuration, every secondary site is fully redundant, i.e. hosts PortaSIP®, billing instances and database servers to operate in standalone mode.

With this release, you can optimize the network topology for georedundant installation in remote regions so that users can connect to the closest available PortaSIP®. To do this, add softswitch-only secondary sites to your system and then enable services failover for them. Your softswitch-only secondary sites will interact with PortaBilling® and the database of the fully redundant site while operating in standalone mode.

Consider the following example:

You want to start your business in Europe. You deploy the main site that hosts a standard PortaSwitch® Procinctus in Frankfurt. For high-availability, you deploy a secondary site which hosts a stand-by database server, PortaBilling® server and a PortaSIP® cluster in New York.

You also start a partnership with a reseller from Bulgaria who aims to serve thousands of customers there. Although in Bulgaria, internal network links are reliable and data transmission is cheap, outer connections are quite expensive. Therefore, to utilize the network topology to effectively handle Bulgarian traffic, you deploy a PortaSIP® cluster within another softswitch-only secondary site in Sofia. In normal mode, this site communicates with the main site in Frankfurt while in standalone mode, it initiates a failover to the secondary site in New York.

Thus, your sites configuration looks like this:

- Frankfurt the main site;
- New York the secondary site with redundancy mode enabled;
- Sofia the secondary site with the New York site defined as the failover one.

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	Create 🛛 🔂 Delete						
	Name T	Гуре	Redundancy mode	Failover site	Description		
1	Main F	Primary	Disabled				
2	New-York S	Secondary	Enabled	New-York			
3	Sofia S	Secondary	Enabled	New-York			
_							

This enhancement optimizes system management and ensures service availability for users on every secondary site of your geo-redundant installation.

New Version of Linux

PortaSwitch® Maintenance Release 66 utilizes Oracle Enterprise Linux version 7.4.

Oracle Enterprise Linux version 7.4 includes a new version of kernel and an updated collection of applications that offer the following benefits:

- Faster performance.
- Straightforward administration.
- Superior security introduced with cryptography enhancements.
- Advanced reliability.

Other Features and Enhancements

• Configure bundle promotions for a volume of consumed services – Now you can provide promotions to your customers not only for the amounts of money they spend on services but also for the volume of consumed services (e.g. number of SMSs sent). Thus, your promotion could state: "Get 15% off for DID usage after 100 minutes of international calls and 25% off after 150 minutes of international calls!"

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			0%	15%	25%				

Let's say EasyCall Ltd. has 10 DID numbers and pays \$5 monthly for each of those numbers. In September, EasyCall Ltd. staff makes 120 minutes' worth of international calls. When the billing period closes for EasyCall Ltd., PortaBilling® calculates the sum of recurring DID charges (e.g. \$50) and applies a 15% discount to it (\$50-\$50*15% = \$50 - \$7.5 = \$42.5).

Of course, you can combine different types of discounts in your promotional bundles, e.g. "Get \$10 off your invoice if you spend \$10 on SMSs and get a 5% discount on subscription fees for using 100 minutes' worth of calls!"

Note that promotional discounts for the volume of consumed services are available only for the new web GUI.

This enhancement provides you with additional flexibility in building your marketing strategy by encouraging customers to use more of the services you provide.

• Language support for CAD currency – With this release, the Canadian dollar (CAD) has been added for all IVR application languages as a supported currency. So now, when you charge your customers in CAD, they are notified about their balance in CAD.

For example: The default currency for the Portuguese language is EUR and it supports USD, BRL and CAD currencies.

Let's say that your residential customer, Juan Perez, who moved from Portugal to Canada, has listed Portuguese as his preferred IVR language. Since he currently lives in Canada, he is charged in CAD.

When Juan wants to check his balance, he calls the IVR access number (e.g. 4545), and hears, "You have 32 dollars and 79 cents" in Portuguese. This enhancement improves the system's usability and at the same time, customers' overall experience with the service provided. To get more information about language support for currencies, consult the **Supported currency** section of the **PortaSIP Media Applications** guide.

- Warning prompts for end users End users can now hear a 'limit reached' warning when:
 - o a permitted number of simultaneous calls is reached, and
 - o bandwidth allocated to their phone calls is used up.

Playing warning prompts to end users requires additional bandwidth. Its amount depends on a number of factors like the codec chosen, the transport protocol used, silence suppression, RTCP presence, etc.

Therefore, to prevent bandwidth shortage, set aside a certain amount of bandwidth when you define the bandwidth limit. For example, to play a prompt using the G.711 codec requires approximately 85 kbit/sec bandwidth. Thus, if you have 1 Mbps allocated for voice traffic, set the bandwidth limit to 915 Kbps (1 Mbps – 85 Kbps).

≡ ↑ Customer	SAVE	CHANGE STATUS				8
EasyCall Ltd. > Services > Voice c	alls					
EasyCall Ltd.	13 / 41	Services		Voice calls configuration		
Business model Hosted IP PBX		VOICE	>	Limit simultaneous calls	•	^
Customer class *		Abbreviated dialing	>	Type of calls		
IP Centrex	*	Bundles	>	All calls	*	
Balance control Postpaid				Simultaneous calls 100		
Currency USD - US Dollar				Forwarded calls		- 1
Balance, USD 93.24786				Incoming cells 20		
Credit limit, USD 1.000.00				Outgoing calls 30		
				Codec connectivity profile		- 1
Personal	~			UIKIIUWI		
Finances	\sim			Max bandwidth	_	
Services	>			915 KDps		
Fraud protection	~	*		Max incoming bandwidth kbps	~	

The 'limit reached' warnings help end users determine why a call is not connecting and what actions to take, and thereby reduces the administrators' workload.

• **Remaining balance for LTE services** – Administrators can now define whether end users must have available funds in order to use LTE services, regardless of whether the service is being provided for free.



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Usage charges	>				
Additional information	>				

By imposing a remaining balance, service providers can hold some reserves back for performing their day-to-day operations as well as cope with any unforeseen expenses.

• New version of Request Tracker (RT) – Beginning with this release, PortaBilling® is using RT version 4.4.2. RT is an issue-tracking system that allows service providers to keep track of issues reported by their customers.

RT version 4.4.2 includes interface updates, stability improvements, bug fixes and several important security updates. This provides improved protection against cross-site request forgery (CSRF), cross-site scripting (XSS) and denial-of-service (DoS) attacks, plus the application offers a better user experience.

• Roaming identification in xDRs / invoices for the LTE service – With this release, PortaBilling® can group xDRs for roaming LTE service usage separately. Thus, when subscribers log into their self-care interface, they can clearly differentiate Internet services consumed within their home network and those used in roaming. The detailed view provides the information about the roaming country and charges. Roaming xDRs are also reflected as a separate group on the invoices.

For instance, John Doe wants to review the costs for the Internet usage in roaming. He browses the billing statistics on his self-care web interface and easily finds the records for roaming services grouped together.



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Transactions								🕑 Help
	Filter							
Billing Summary Products and Services Outcas and Service Vallets Volume Discounts	From Date: 20 Till Date: 20 Service: Al	17-10-24 YYYY-J 17-10-25 YYYY-J Show Failed Transactions , Show Records	16:46:22 IM-DD 16:46:22 wmload	HH24:M HH24:M	11:55 11:55			
	Summary Service		Total Transactions		Charged Quantity	Charged Amount		Show Details
>	LTE (Roaming)			2	4 (megabytes)		0.80000 USD	EQ.
	LTE			2	4 (megabytes)		0.40000 USD	B
	Total Service Used:	2 Tota Tran	sactions:	4		Total Charged Amount:	1.20 USD	
	14 4 Page 1 of 1	I ▶ ₩ I &						1 - 2 of 1
	LTE, Roaming							
	Description	Con	lect time		Disconnect time	Data Volume, megabyte	Amour	nt, USD
	BELARUS, Mobile Digita	al Communications						۷
		2017-10	25 02 02 27	201	17-10-25 02:02:28	2		0.40000 USD
		2017-10	25 02:00:55	201	17-10-25 02:00:55	2	1	0.40000 USD
	4 Page 1 of 1	I > M 2						1 - 2 of 2
	Q Show Records							

To configure the system to group the roaming xDRs, an administrator must specify the home PLMN (Public Land Mobile Network) code in the Internet access policy (e.g. home-plmn-e212-match = 310).

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home-plmn-e	12-match				نـــــ×		
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This enhancement provides a clear understanding of all roaming charges in the xDR browser and on the invoices, and helps with troubleshooting.

Web Interface Changes

• **Renamed account role** – The **MSISDN** account role is used to configure mobile network subscribers in PortaSwitch®. It has recently been renamed **Mobile**, which gives a clearer idea of account intended use.

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44055556787@msisdn	18 / 64	Service configuration		Voice calls configuration
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lype Credit Customer				Override RTP proxy Use Default
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Important Upgrade Notes

- Changes in bundle promotions on new web interface The bundle promotions functionality has undergone the following changes when managed via the new web interface:
 - New format for bundle promotion types. Now an administrator can clearly distinguish between "credit" and "charge" type bundle promotions. They are represented as the promotional discounts for service usage and minimum service usage commitments, respectively. The wizard guides the administrator through the configuration steps, making the whole process very simple.
 - Both types of bundle promotions can be based either on the amount of funds spent on a service or on the volume of the consumed service.
 - Minimum service usage commitments can have only one, fixed-value threshold. Previously configured bundle promotions of this type having multiple thresholds will work as they are. However, they are read-only and can only be deleted.
 - Minimum service usage commitments apply either to the same service as the one selected for analysis or to the whole bill. The previously configured bundle promotions that apply to different services are read-only and only can be deleted.
 - The administrator no longer needs to define a charge amount explicitly when configuring money-based minimum service usage commitments. Charges are calculated automatically at the end of the customer's billing period as the difference between their actual service usage and the defined minimum. Bundle promotions with explicitly defined charges will work as they are. If you edit

this promotion, the charge value will be adjusted automatically to equal the defined service usage minimum.

• Deprecated Charged Amount Rounding option in volume discount plans – With this release, PortaBilling® uses the rounding settings defined for a customer tariff to round off the amount of monetary-based discounts, quotas and service wallets consumed. Accordingly, the Charged Amount Rounding option is now discontinued and has been removed from the Edit Discount Plan page.

So now if an account's charges for voice calls are rounded off using the XX.XX000 pattern, the same pattern will be used to round off the amount deducted from the volume discount plan's counter. That is, if John makes a call worth \$1.247, the charge will be rounded up to \$1.25 and John's volume discount counter will be adjusted accordingly.

The same logic applies to multi-service wallets: their counters are adjusted based on the service used within the service pool and rounding pattern defined within the tariff applicable for this service.

If you have previously and separately defined the rounding for volume discount plans, make sure to configure it for your tariffs after the software update.

- New SureTax API As of this release, PortaBilling® operates with SureTax via the updated API. Changes introduced by SureTax include the following:
 - The new URL for the API server access in this format: https://<API_server_domain>/Services/Telecom/V01/S ureTax.asmx/PostRequest
 - The discontinued Federal TRS (Federal Regulatory Assessment Fee) Exempt tax.

Consequently, verify your taxation configuration before the software update and adjust it, if necessary, to ensure proper taxation.

• Release of DID numbers from customers upon account termination – As of this release, DID numbers do not remain assigned to a customer either upon an account ID change or termination. These DID numbers return to the DID pool and / or remain within their respective pricing batches. This enhancement prevents excessive charges for DID users. • New format for IVR-related rating entries – Beginning with this release, rating entries associated with an IVR application must be added to the relevant product with an access code that matches the IVR access number. Note that rating entries with the access code ANY no longer work for such kinds of calls.

Therefore, reconfigure all IVR-related rating entries for all the necessary products before performing a software upgrade.

Alternatively, the support team can temporarily enable backward compatibility mode for you to ensure safe migration before a software upgrade. When you complete reconfiguring all the necessary products, contact the support team to disable backward compatibility mode.