



New Features Guide

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Preface

PortaSwitch® Maintenance Release 93 is the next leap-forward release, consistent with our “fast releases, precisely on time” ideology.

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.

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.



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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Empower users to control how they accept PBX calls with extension modes (business hours, non-working hours)

PBX users can now use modes to control when and how calls come through to their line, and quickly change the way incoming calls are handled. For example, they can set their mode to accept calls during business hours, send all calls to voicemail at once during non-working hours, and forward calls to another colleague when on vacation. PBX users can change the mode for their extension by calling the IVR or on their self-care interface.

The PBX administrator can quickly change the way incoming calls are handled for all PBX extensions at once. Say the entire staff has to leave for a fire drill – the PBX administrator calls the IVR, and changes the mode from “Business hours” to “Emergency”. Now, all incoming calls will be forwarded directly to mobile phones, and everyone who phones the office still receives assistance.

This is how it works:

Let’s say ABC company has three modes configured: “Business hours”, “Non-working hours”, and “Emergency”. Each mode has a unique Dual-Tone Multi-Frequency (DTMF) code that is used for switching. Mary, a sales agent, goes out for business lunch (during business hours) and doesn’t want any calls to disturb her colleagues in the office. Thus, to forward all calls to voicemail at once, Mary dials *61 on her phone and specifies the DTMF code for “Non-working hours” mode. Once the mode is changed, Mary stops receiving calls, both from clients and other agents. All the calls are forwarded to voicemail. In two hours, Mary comes back to the office. She dials *61 on her phone again and changes the mode to business hours, meaning now she can receive calls.

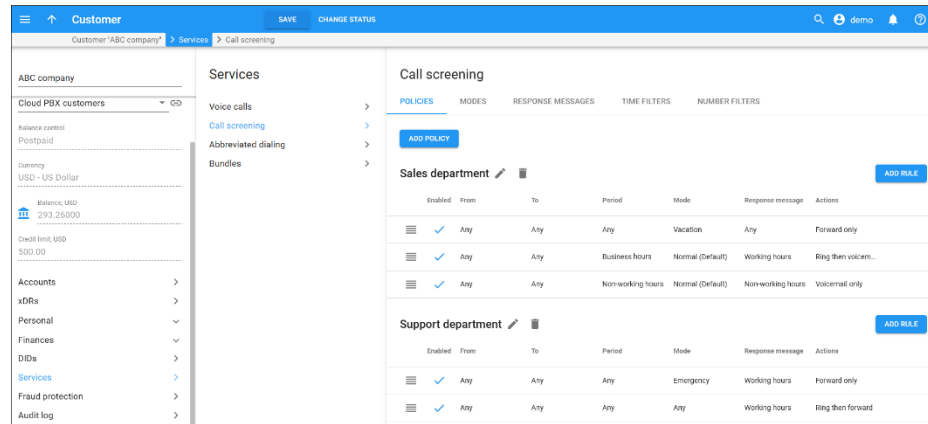
Benefits

- PBX users have more control over the incoming calls so that they can quickly change the modes to control what calls to accept.
- PBX administrators gain control over all incoming PBX calls. This allows for managing who can reach PBX users and at what time by quickly changing the way all incoming PBX calls are handled via IVR.

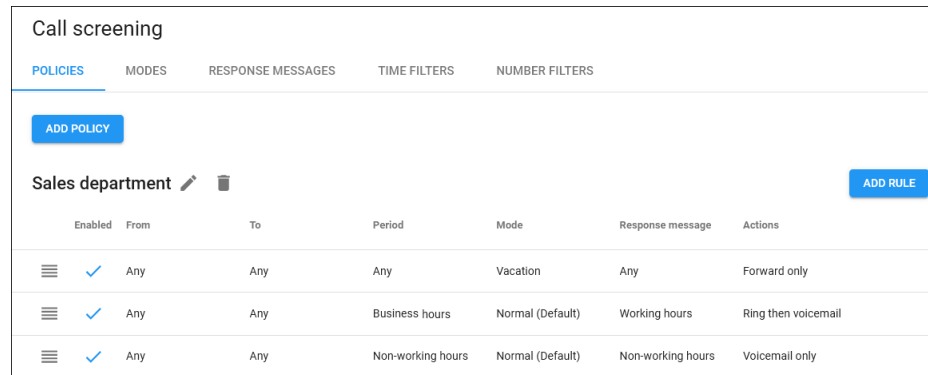
Policies

Policy contains a set of rules that define whether to play the response message (personalized audio message to play to callers before the call is answered) and what call action to apply (e.g., ring, forward, voicemail). A PBX customer may have several policies, e.g., a policy for each company department.

The PBX administrator can assign a policy to a specific account.



The order of rules matters. When the call arrives at PBX, PortaBilling® checks the rules within the policy from top to bottom. The first rule that matches the call is applied, and the other rules are ignored. PortaBilling® checks the rules with these parameters: caller/callee numbers (it can be a pattern, e.g., 2233%), period (time interval), mode of the called extension.



Modes

The Normal (Default) mode is present in the system. The DTMF code of the default mode is always zero (0). This DTMF code is used to switch to this mode. Users can switch from individual sticky mode to Normal (Default) mode to continue receiving calls according to the rules defined for the Normal (Default) mode.

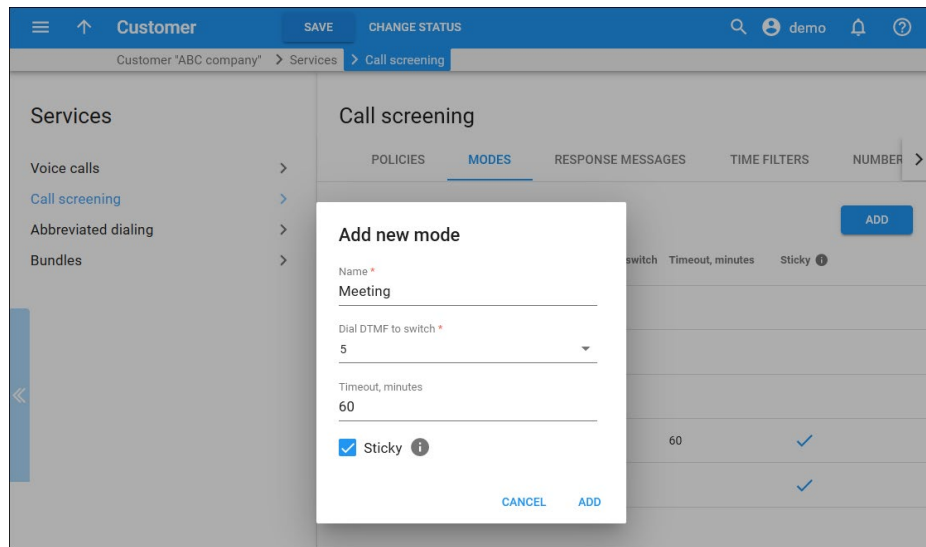
Call screening				
POLICIES	MODES	RESPONSE MESSAGES	TIME FILTERS	NUMBER FILTERS
List of modes ADD				
Name	Dial DTMF to switch	Timeout, minutes	Sticky	
Normal (Default)		0		

The PBX administrator can set the following parameters for a mode:

- **Dial DTMF to switch** – this is a unique DTMF code of the mode. Users dial it to switch to this mode. Digits from 1 to 9 are available.
- **Timeout, minutes** – this is mode duration in minutes, after which the mode automatically switches to Normal (Default) mode. For example, there is a “Meeting” mode with 60 min timeout. If Mary switches to “Meeting” mode at 3 p.m, the mode switches to Normal (Default) mode at 4 p.m.
- **Sticky mode for individual use** – if the mode is marked sticky, only extensions can set this mode. If the PBX administrator switches all the extensions to the other mode, this change doesn’t influence the extensions with sticky mode. These extensions remain in this mode until they change it via IVR or their self-care interface, or sticky mode’s timeout ends.

Let’s say John Doe changes the mode from Normal (Default) to “Vacation” (all calls are forwarded to the other PBX extension) on their self-care interface. Later, the PBX administrator changes the mode for all PBX extensions to “Emergency” (calls are forwarded to the mobile phone). Despite this change, all incoming calls to John are handled according to vacation mode. Thus, when Mary calls John, the following happens:

- first the call is forwarded to the extension of John’s colleague, Ann, (according to “Vacation” mode), and
- then the call is forwarded to Ann’s mobile phone (according to “Emergency” mode).

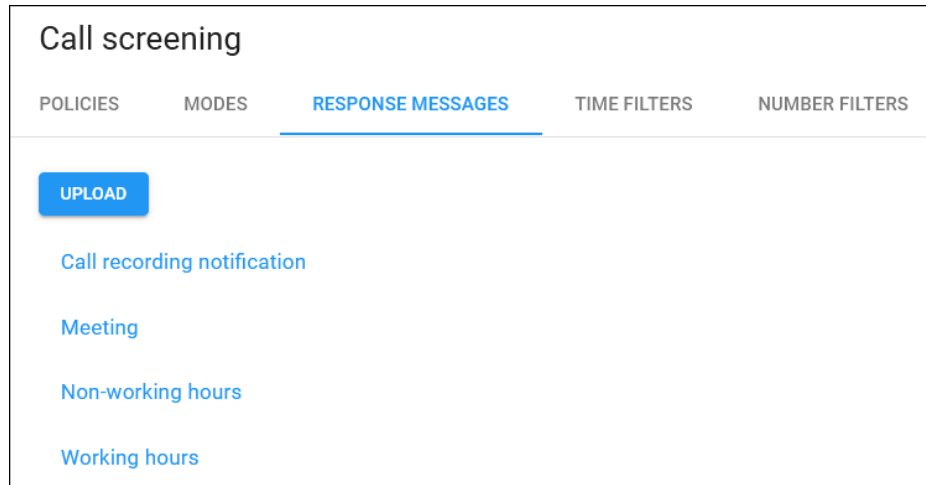


The PBX administrator can see all the modes and their configuration.

Call screening			
POLICIES	MODES	RESPONSE MESSAGES	TIME FILTERS
List of modes			ADD
Name	Dial DTMF to switch	Timeout, minutes	Sticky ⓘ
Normal (Default)	0		
Business hours	1		
Non-working hours	2		
Emergency	3		
Meeting	5	60	✓
Vacation	9		✓

Personalized response messages for incoming calls

PBX users and administrators can record a personalized audio message to play to their callers before the call is answered, forwarded, or redirected to voicemail. This gives a caller additional information, such as the user is on vacation. For more information, see the Empower users to control how they accept PBX calls with extension modes [Personalized response messages for incoming calls](#).



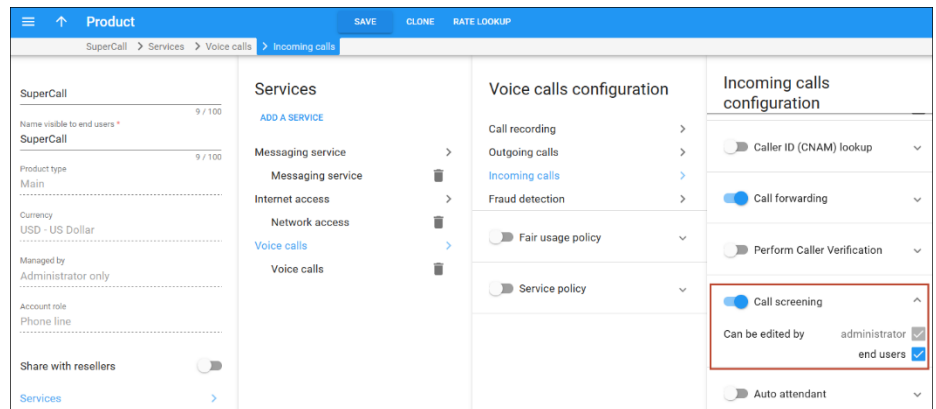
Configuration

Let's say ABC company wants to configure three modes for their PBX extensions:

- Business hours – to accept all calls from 9 a.m. till 6 p.m.
- Non-working hours – to forward calls to voicemail.
- Vacation – this is a sticky mode per individual use only. Users can enable it during vacation. All calls will be forwarded to the other PBX extension/phone number.

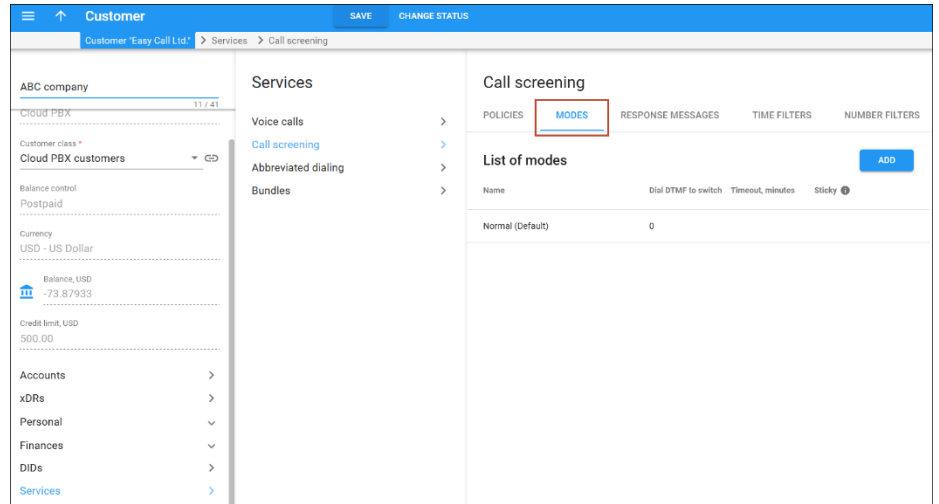
To configure the call handling for ABC company, the administrator performs the following steps:

1. Enable **Call screening** feature on the product:
 - Open **Product > Services > Voice calls > Incoming calls**.
 - Turn on the **Call screening** toggle switch.
 - Select the **end users** checkbox to enable users to edit call screening rules on their self-care interface.



2. Configure a policy to handle incoming calls:

- Open **Customer > Services > Call screening > Modes** tab.



- Click **Add** to create a new mode:
 - Business hours – select 1 as a DTMF code.
 - Non-working hours – select 2 as a DTMF code.
 - Vacation – select 9 as a DTMF code, select the **Sticky** checkbox so that only PBX users can use this mode.

Edit mode

Name *

Vacation

Dial DTMF to switch *

9

Timeout, minutes

Sticky i

CANCEL
EDIT

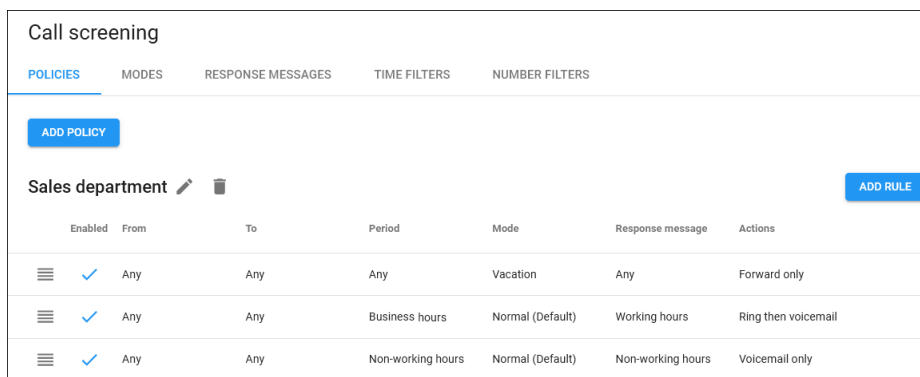
All the modes appear on the interface.

Name	Dial DTMF to switch	Timeout, minutes	Sticky ⓘ
Normal (Default)		0	
Business hours		1	
Non-working hours		2	
Vacation		9	✓

- Go to **Time filters** tab, click **Add** to create a new filter:
 - Business hours – from 9:00 till 18:00 on Monday-Friday of every month.
 - Non-working hours – from 18:01 till 8:59 on Monday-Friday of every month and the whole day on Saturday-Sunday.

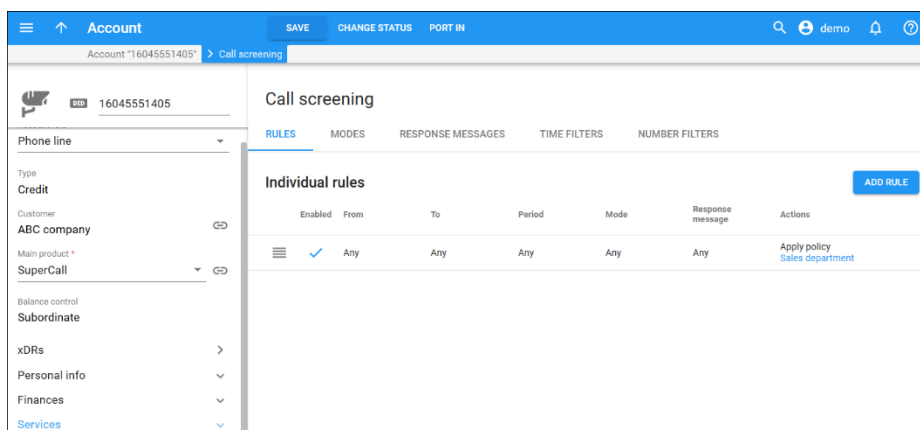
Name	Calendar Icon	Filter Description	Action
Business hours	📅	From 09:00 till 18:00 on Mon-Fri of every month	Hide +
Non-working hours	📅	From 20:01 till 08:59 on Mon-Fri of every month	+ Hide
	📅	Whole day on Sun, Sat of every month	

- Go to **Policies** tab > **Add Policy** > **Add rule**.
- Create three rules specifying a period, response message and actions such as ring, forward, voicemail, etc.
- Rearrange the rules so that “Vacation” mode becomes the first rule.



3. Apply policy to every account so that they can use the predefined modes:

- Open **Account > Services > Call screening**.
- Click **Add rule** and select a policy.



By default, *61 is the prefix for changing mode for a specific extension and *62 for changing the mode for all PBX extensions at once (PBX manager option must be enabled for the account). It's possible to change the default prefixes by creating a new dialing rule.

Personalized response messages for incoming calls

PBX users can record a personalized audio message to play to their callers before the call is answered, forwarded, or redirected to voicemail. This allows PBX users to give a caller additional information, for example, that the call is being recorded.

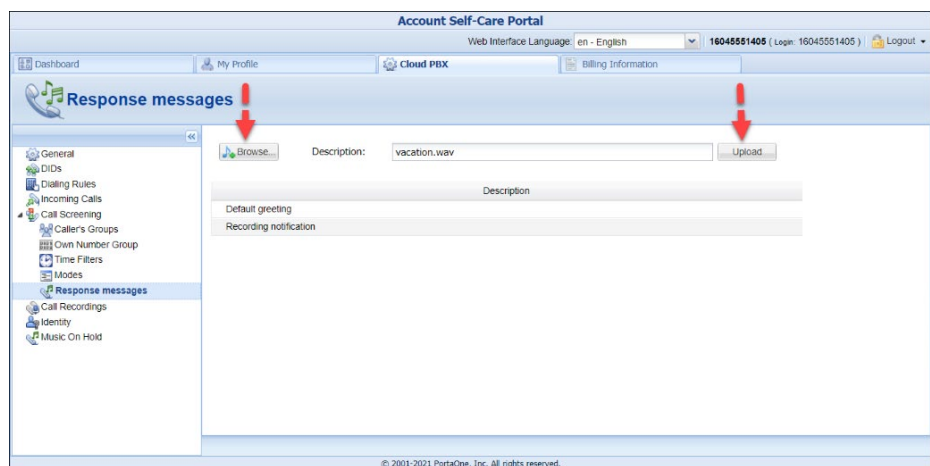
Let's say Mary Smith goes on vacation from August 1 and wants to record a personalized message to play when somebody calls her. Beforehand, the PBX administrator has to set several call processing modes: “Business hours”, “Non-working hours” modes, and a “Vacation” mode for the users who go on vacation. You can read more about operation modes in the [Empower users to control how they accept PBX calls with extension modes \(business hours, non-working hours\)](#) chapter.

To switch from “Business hours” to “Vacation” mode, Mary dials *61 on her IP phone. Then she follows the voice-recorded instructions to record a personalized message: “I’m on vacation till August 5. Your call will be forwarded to my colleague, Ann”. When John Doe, a client, calls Mary on August 3, he hears the message in full, and only then the call is forwarded to Ann’s phone. Ann picks up the call and talks to John. When Mary comes back to work on August 6, she calls the IVR (*61) again and dials a code that switches her phone line back to “Business hours” mode.

The message, added by the user, always overrides the default message added by the PBX administrator, no matter whether it is recorded via the IVR or uploaded via the self-care interface. The audio message can be in .wav, .mp3, .og, or .au format and don’t exceed the size of 3 MB.

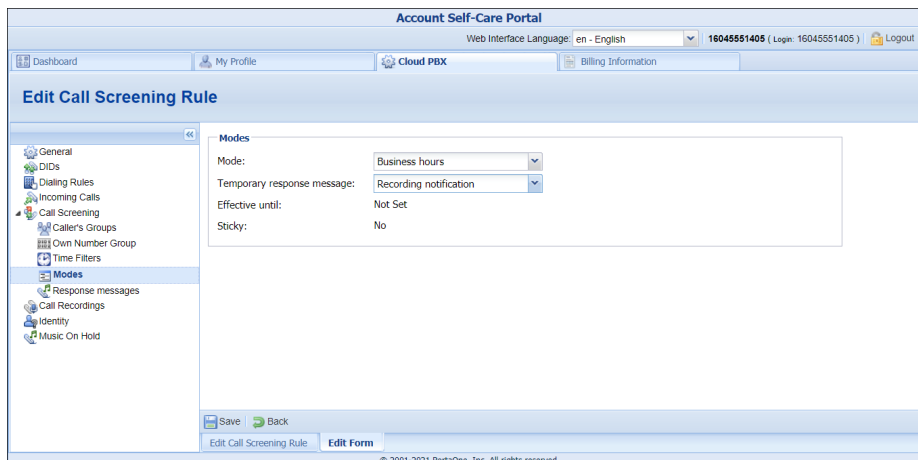
To add the personalized message to a specific mode via self-care interface, the PBX users have to:

1. Upload an audio file to the self-care interface:
 - Go to the **account self-care interface > Cloud PBX > Call screening > Response messages**.
 - Click **Browse**, choose the file from PC and click **Upload**. The name of the file automatically appears in the **Description** field.

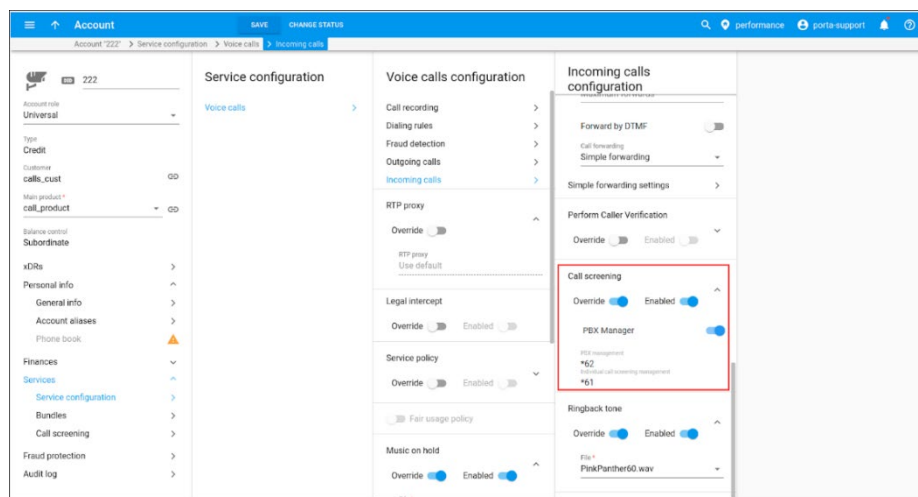


2. Add an audio file to the mode:

- Open **Cloud PBX > Call screening > Modes**.
- Click **Edit**, select a specific mode from the drop-down menu, and afterward select the **Temporary response message**.



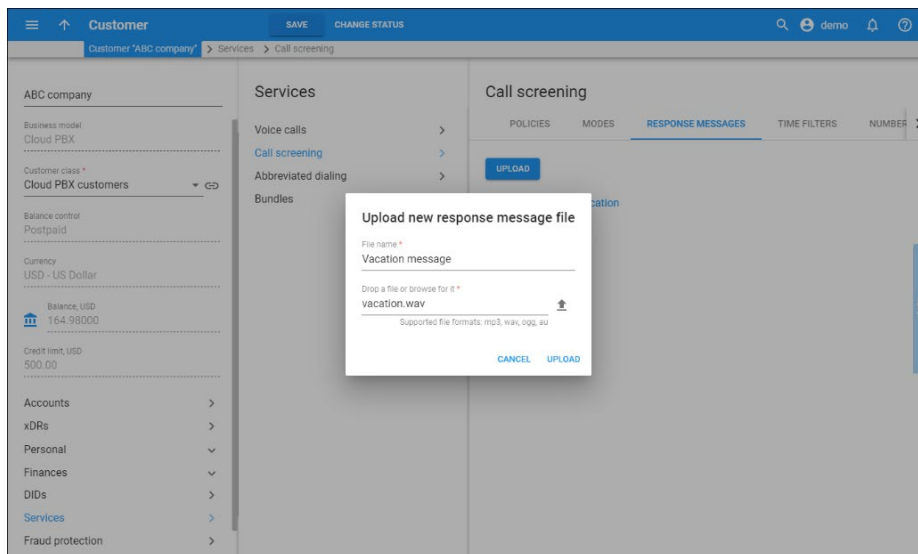
PBX administrators can also add a default message that applies to all PBX extensions, both via a customer self-care interface and by calling the IVR. Make sure that the **PBX management** feature is enabled for this extension. To do it, go to the **PortaBilling® interface**, open the **Account > Services > Service configuration > Voice calls > Incoming calls > Call screening**, and enable **PBX manager** toggle switch.



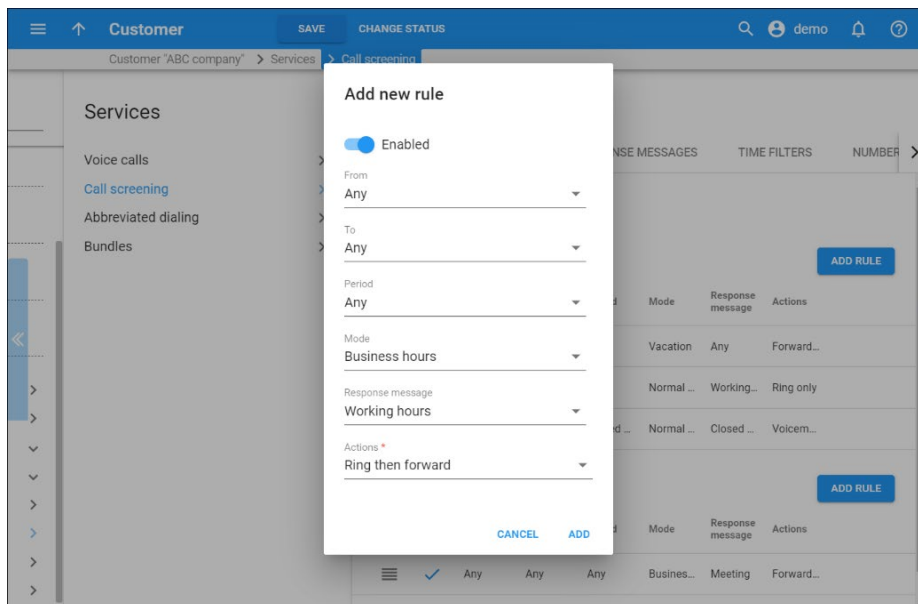
PortaBilling® administrator can upload a default message on the PortaBilling® web interface:

- **For the customer:** The administrator can upload a default message that will apply to all PBX extensions that have a specific mode, except those extensions that set their own personalized message.

To add a default message, the administrator goes to the **Response messages** tab and uploads a file.

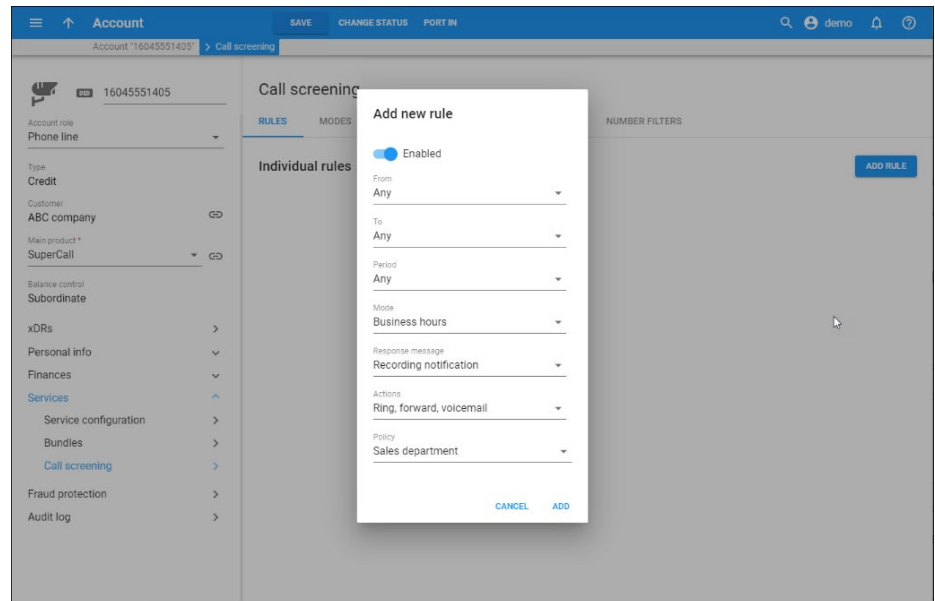


Afterward, to add the default message to the mode, the administrator goes to the **Policies** tab, adds a rule to the specific policy, selects the mode, and chooses the audio file in the **Response message** drop-down list.



- **For a specific account:** Personalized messages can be applied to each of the accounts separately. The administrator has to upload the file in the **Response messages** tab and afterward add it to the mode via the rule in the **Rules** tab. The file uploaded to the

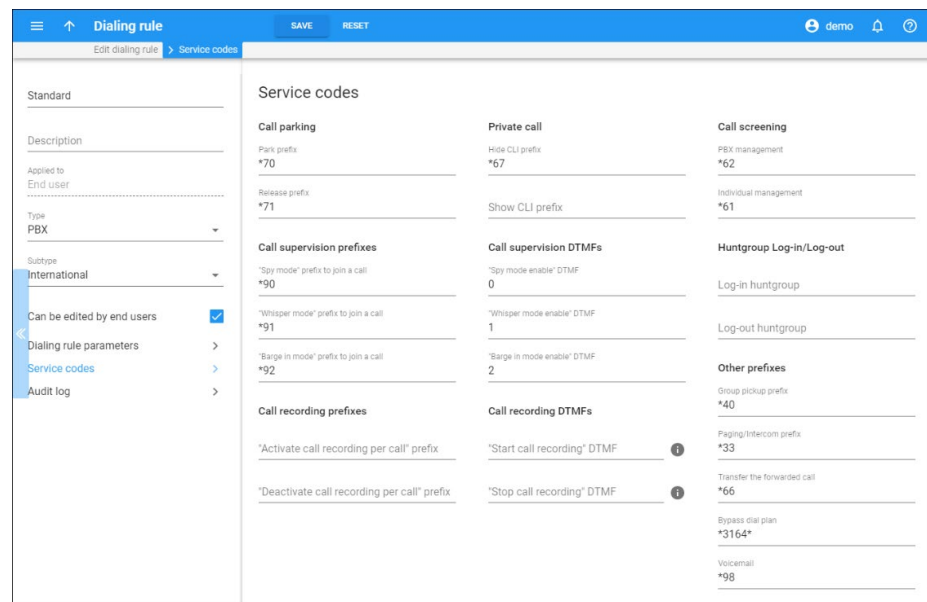
account **Response messages** list will appear on the account self-care portal.



Default service codes to change response messages:

- ***62** - PBX management default code. When called, the response message for all accounts of a certain customer changes.
- ***61** - individual management default code. When called, the response message for a single account changes.

You can change the default service codes by creating a new dialing rule for the customer.



Benefit

PBX users can provide better service to their clients with informative audio messages.

Call recording announcement

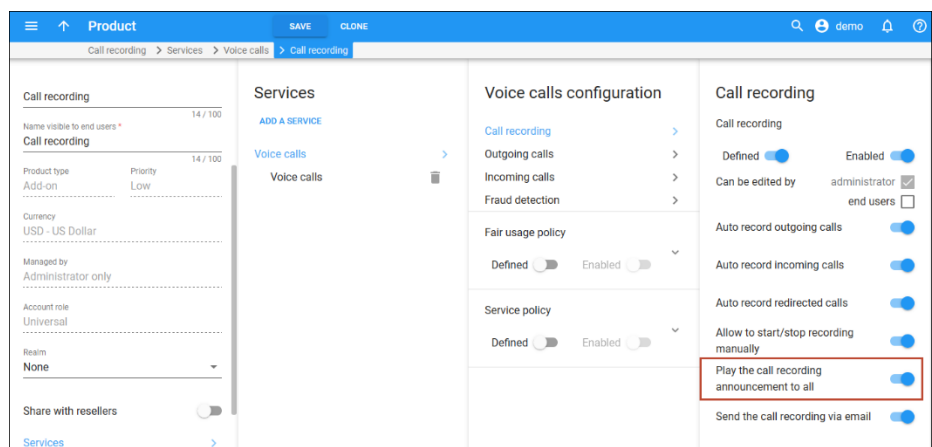
Now, businesses can notify everyone on the call that their conversation is being recorded, allowing them to comply with call recording regulations. When call recording starts, all parties hear the “Call recording started” announcement. This ensures that everyone on the call is aware that the call is being recorded.

For example, Mary calls ABC company, where all the calls are recorded automatically. The call is received by the auto attendant, and then it is transferred to the sales huntgroup. Peter, the sales agent, picks up the call, and both Mary and Peter hear the “Call recording started” announcement. Later, Peter manually stops the recording during a call since Mary is going to provide some sensitive information such as credit card details. In this case, the “Call recording stopped” announcement is played.

The recording always contains the announcement, ensuring that the call parties are notified of the call recording.

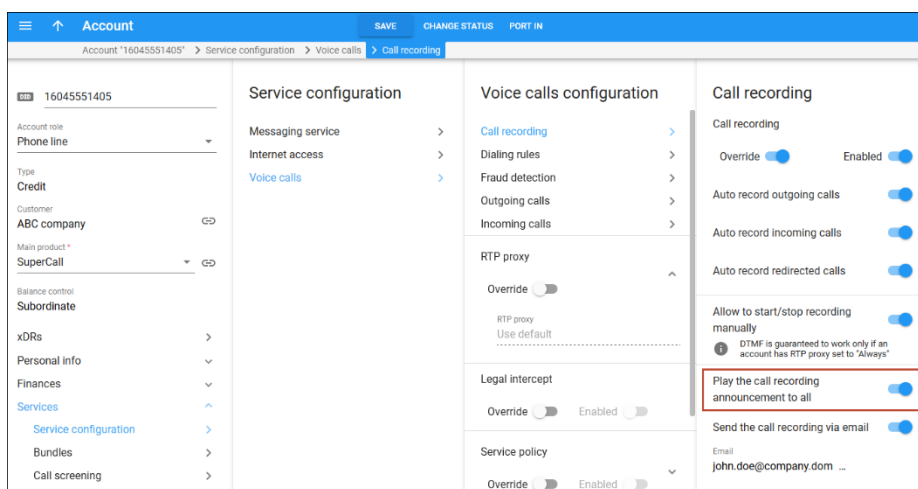
To enable the call recording announcement option for all PBX users, the administrator needs to:

- Open a specific product/add-on product with call recording feature enabled.
- Go to **Services > Voice calls > Call recording**.
- Turn on the **Play the call recording announcement to all** toggle switch.

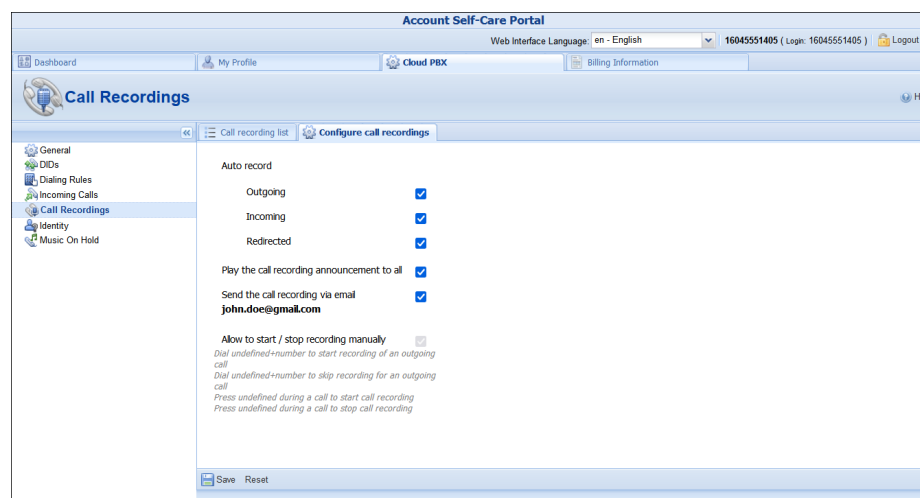


The administrator can enable the call recording announcement option for a specific account. To do this, the administrator follows these steps:

- Open a specific account.
- Go to **Services > Service configuration > Voice calls > Call recording**.
- Turn on the **Play the call recording announcement to all** toggle switch.



End users can manage whether to play the call recording announcement on their self-care interface (if the administrator enabled it for end users beforehand).



Note that the call recording announcement is available only in English. If you need the announcement in another language, contact [PortaOne® support team](#).

Benefit

Businesses comply with call recording regulations by playing a call recording announcement to all call parties.

Allow hunt group members to pause receiving hunt group calls

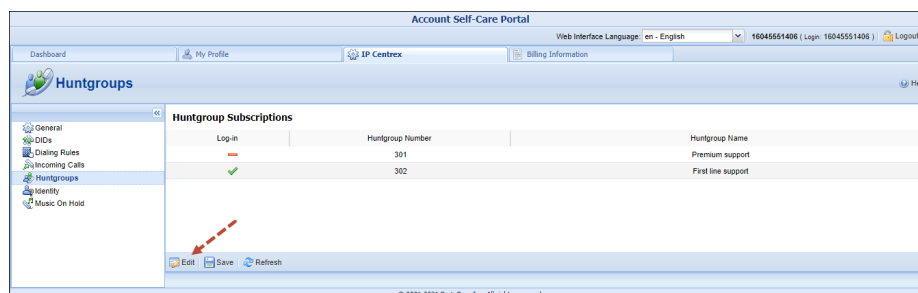
Hunt group members can stop receiving hunt group calls by logging out of a hunt group. To resume receiving the calls, they log in back to the hunt group.

Let's say John, a support agent, receives calls sent to the “First line support” hunt group.

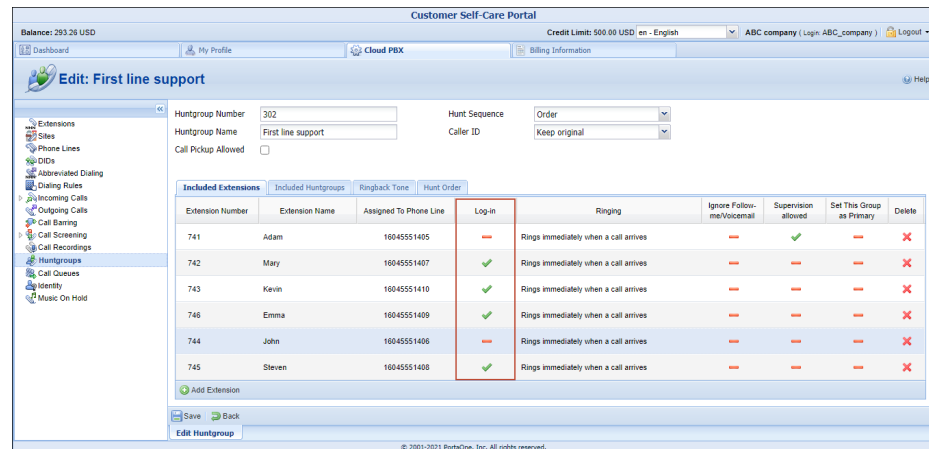
John needs to write an email to a client. While working on this task, he will be unable to take support calls. Thus, to log out of the “First line support” hunt group, John dials *42302 on his phone, where *42 is the prefix to log out and 302 is the hunt group number. Now, John doesn't participate in handling calls sent to this hunt group but can still receive direct calls. He starts writing the email. John's manager, Adam, calls John to ask whether the email has been already sent to the client. John discusses a few email details with Adam and finishes it. When John sends the email to a client, he logs back in to the “First line support” hunt group. To log in, John dials *41302, where *41 is the prefix to log in and 302 is the hunt group number. Once John is logged in again, he receives another support call.

Note that to log out/log in, John must specify the number of a hunt group he is a member of.

Also, agents can use their account self-care interface to log out/log in: they open the **Cloud PBX** tab > **Hunt groups** page > see the list of hunt groups they are members of > click **Edit** > clear/select the checkbox for a specific hunt group, e.g., “First line support” > click **Save**.



On the customer self-care interface, the PBX administrator can monitor whether the hunt group members are logged-in or logged-out at the moment.

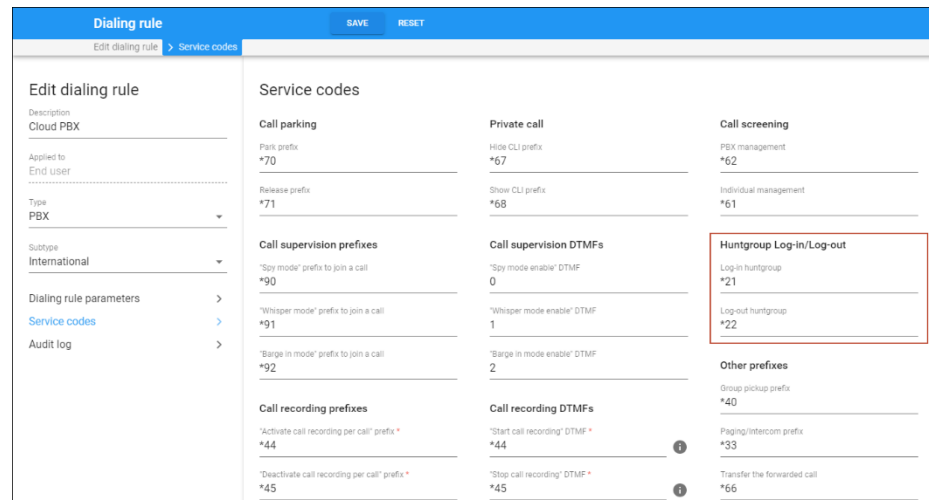


Specifics

By default, *41 is the prefix for logging in and *42 for logging out. It's possible to change the default prefixes by creating a new dialing rule.

For example, a customer migrates to cloud PBX from their legacy PBX. The customer's agents are used to stop/resume receiving hunt group calls by dialing the prefixes *22 and *21. To enable logging out/logging in with these prefixes in cloud PBX, the administrator:

- creates a customer record with the dialing rules option enabled;
- creates a new dialing rule of PBX type; and
- changes the default prefixes in the dialing rule.



PBX customers can edit the default prefixes on their customer self-care interface: they need to open the **Cloud PBX** page > **Dialing rules**,

switch to the **Custom rule** and change the prefixes for hunt group logging in/logging out.

Benefits

- Agents can control whether to receive their hunt group calls.
- PBX customers can monitor the availability of their agents.
- PBX customers can provide better service by delivering calls to available agents only.

Control how charges are rounded when billing clients

In certain countries, such as Lithuania, there are strict regulations for charging customers. Service providers can be fined if the charged amount is rounded using a wrong method. Thus, service providers need the ability to select what rounding method to apply to customers' charges and invoice amounts.

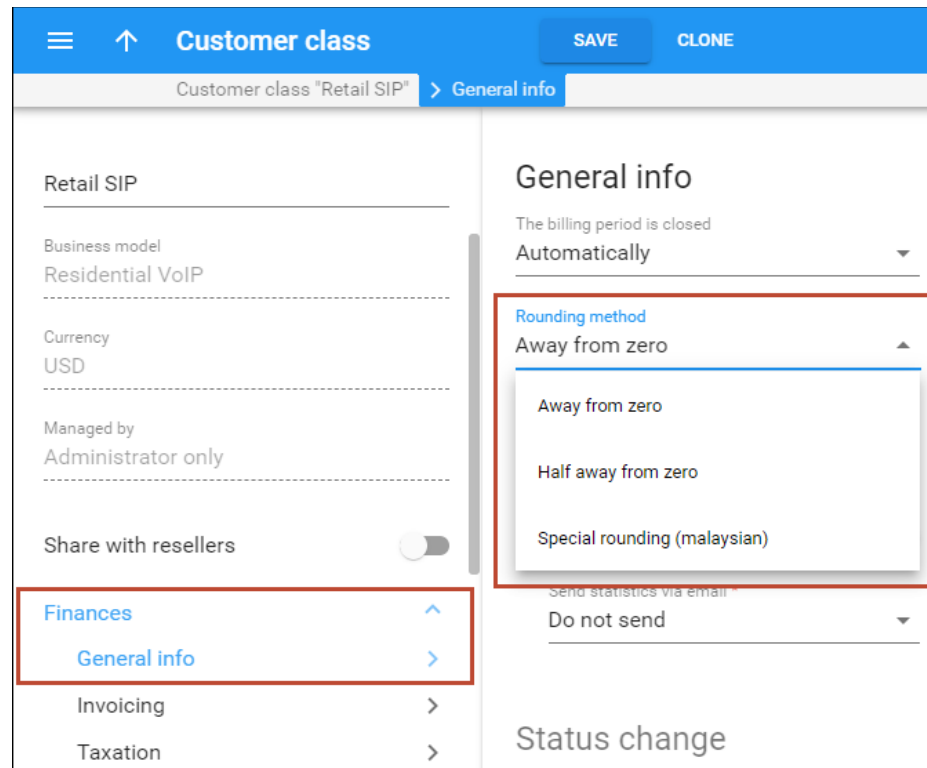
With this release, the administrator can change the rounding method globally in the customer class. The selected rounding method will be applied to:

- invoice amounts, and
- charged amounts for subscriptions, bundle promotions, measured services, and DID charges in an individual xDR.

The administrator can select one of the following rounding methods:

- **Away from zero** – this rounding method is selected by default. It works similar to rounding up but differs when rounding negative values. Positive and negative values round symmetrically. For example, if the rounding precision is set to two decimals, then:
 - 1.214, 1.215 and 1.216 all round up to 1.22.
 - - 1.214, - 1.215 and - 1.216 all round to - 1.22.
- **Half away from zero** – this rounding method works similar to arithmetic rounding but differs when rounding negative values. Positive and negative values round symmetrically. For example, if the rounding precision is set to two decimals, then:
 - 1.214 rounds to 1.21, 1.215 and 1.216 all round to 1.22.
 - - 1.214 rounds to - 1.21, - 1.215 and - 1.216 all round to - 1.22.
- **Special rounding (malaysian)** – formerly known as custom rounding. This type of rounding depends on the last decimal at precision point. For example, if the rounding precision is set to two decimals, then:

- If the last decimal at precision point is [0...2], it is set to zero. For example, 1.204, 1.215 and 1.226 all round to 1.20.
- If the last decimal at precision point is [3...7], it is set to 5. For example, 1.234, 1.255 and 1.276 all round to 1.25.
- If the last decimal at precision point is [8...9], it is set to 0 and the previous decimal increases by 1. For example, 1.284, 1.296 all round to 1.30.



Also, the administrator can choose the number of decimals to round the charged amount in the **Rounding precision** option. For invoice amounts and charged amounts for measured services the **Rounding precision** can be configured in the customer class. The **Rounding precision** for a tariff, subscription, bundle promotion and DID charges can be configured individually per entity.

The screenshot displays the 'Subscription plan' configuration interface. The top navigation bar includes a menu icon, an up arrow, the title 'Subscription plan', and a 'SAVE' button. Below the navigation bar, the breadcrumb trail shows 'Subscription plan "Basic calls" > Fees'. The main content area is divided into two columns. The left column contains settings for 'Basic calls', including 'Subscription plan name visible to end users', 'Basic calls', 'Subscription charges applied In advance', 'Periods in advance' (set to 1), 'Activation mode' (set to 'At the given start date'), and a 'Share with resellers' toggle. The right column is titled 'Fees' and contains a list of rounding precision options: '0.01 (2 decimals)', '0.00001 (5 decimals)', '0.0001 (4 decimals)', '0.001 (3 decimals)', and '0.01 (2 decimals)'. The '0.01 (2 decimals)' option is selected and highlighted with a red box. Below the list, there are two checked options for 'Prorate fee for the partial billing period': 'For the first billing period' and 'For the last billing period'.

For example, a service provider starts operating in a new country. To comply with local regulations, a service provider wants to apply the **Half away from zero** rounding method to all charged and invoice amounts of their customer John Doe. Also, the service provider wants to round the charged amounts for subscriptions, to two decimals in an individual xDR. So, an administrator assigns a customer class to John Doe where the “Half away from zero” rounding method is selected. Then the administrator creates a “Basic calls” subscription plan for Jon Doe, goes to the **Fees** panel and selects “**0.01 (2 decimals)**” in the **Rounding precision** option.

Benefit

Service providers can comply with local regulations by applying a specific rounding method to their customers' charges and invoice amounts.

Add wildcard SIP domains

When an incoming call comes to PortaSwitch®, PortaSIP® validates the domain name contained in the incoming Invite request to prevent the traffic that is not intended for the service provider's servers. PortaSIP® checks that the domain name is specified in the list of allowed domains owned by a service provider to authenticate the caller.

For example:

```
INVITE sip:27510230043@greentelecom.sbc.8ic.co SIP/2.0
```

The list of allowed domains is specified in the **sip_domains** option on the Configuration server.

Each customer who connects their Microsoft Teams to PortaSwitch® has to have their own domain. Previously to route calls between PortaSwitch® and the customer's Microsoft Teams environment a service provider added the domains manually for each new customer. To add each new SIP domain, you needed to apply the configuration with a service restart that led to service downtime.

With this release, you can add a single wildcard SIP domain that matches all new subdomains.

This is how it works:

For example, a service provider adds *.sbc.8ic.co domain name where "*" is a wildcard that matches all new subdomains. Thus, if they add *.sbc.8ic.co to the list of allowed domains, incoming calls from **abctelco.sbc.8ic.co**, **intertelecom.sbc.8ic.co**, and **xxxxxxxxx.sbc.8ic.co** will be allowed.

Benefits

- Service providers save time on adding new domains for customers who have Microsoft Teams by adding a wildcard SIP domain.
- End users don't experience service downtime due to adding new domains.

Match events related to the same call via the call control API

External applications that use call control API can match call events related to the same call by the **tracking_id** attribute. In complex call scenarios involving attended transfer, call pickup, call parking, etc. where a few calls get connected, the **tracking_id** value changes. In this case, PortaSwitch® sends both the new and the previous **tracking_id** values, so an application can track the whole call session consisting of multiple connected calls. Thus, application developers don't need to write and maintain additional code to match the related call events in an application.

For example, ABC company uses a web-based switchboard that allows PBX users to control and monitor calls. When John, ABC's sales manager, calls a client, the application receives notifications for each call status change ("trying", "ringing", "connected", "on-hold", etc.). All these notifications contain the **tracking_id** attribute. Since these call event notifications include the same **tracking_id**, the application identifies them as a part of the specific call. When the client tells John they have a technical issue, John transfers the call to the support department. During the attended transfer, the **tracking_id** changes. The application receives a notification that includes a new value (**tracking_id**) and the previous value (**previous_tracking_id**). The application uses these values to properly track the whole call session consisting of connected calls. As a result, Mary, ABC's receptionist, can see on the switchboard whether John's phone line is busy before transferring a call to him.

Tracking_id is included in all "sip.call_control_notifications" notifications, such as "call_info", "conference_info", "participant_info", etc.

Here's an example of an event notification sent to an application:

```
{
  "call": {
    "id": "ZGZiODJkZmQ2ZmJjNDJjYzQyMTQ1OWQyZGFjYWRhNWE.",
    "tag": "7147342c"
  },
  "previous_tracking_id": "c67e10bd-c35c-41e7-a3d9-e40e7fb96767",
  "tracking_id": "e2dd7af5-7c0d-4e75-a9e2-043bd83c5e37"
}
```

Benefits

- Fewer implementation efforts for external application developers.
- Service providers can reduce costs and delays associated with building their external applications.

Simplified implementation of click-to-call service using the call control API

Service providers may want to implement a simple integration using call control API, e.g., add a possibility to initiate calls from an external application and track the status of these calls. Now, call control API allows applications to receive call statuses via HTTP/HTTPS protocol with no need to connect via WebSocket.

Let's say, a service provider wants to embed click-to-call service in their CRM application so the CRM users can initiate calls from the CRM with one click. A user clicks a number and receives a callback on their IP phone. They pick up the call and get connected with the desired destination. To implement this service, the call control API method **originate_advanced_call** is used. With this release, the statuses of calls initiated with this API method can be sent to the application in HTTP POST requests. This allows the application to display the current call statuses to users. Thus, the call status tracking can be implemented by developers without WebSocket experience. Also, it's possible to specify which statuses to receive and deal with the needed notifications only.

To subscribe an application to call status notifications, the **originate_advanced_call** API method is extended with two optional parameters: **state_callback** and **state_callback_events**.

When a user initiates a call from an application, the application sends **originate_advanced_call** request with **state_callback** and **state_callback_events** parameters to PortaSwitch®:

- **state_callback** – contains the URL (HTTP/HTTPS) to send the call status to an application when one of the events specified in **state_callback_events** happens;
- **state_callback_events** – contains the statuses (call progress events) that must be sent to the **state_callback** URL. It's possible to specify multiple events:
 - trying – a call is initiated, and an outgoing request is sent;
 - ringing – a phone is ringing;
 - early – early media is played;
 - connected – a call is answered/taken from hold and the remote side is connected;
 - terminated – a call is disconnected; this status is sent by default;
 - held – a call party is connected and is put on hold (this status is returned to the party placed on hold);
 - holding – a call party is connected and is put on hold (this status is returned to the party placing the call on hold).
 - queued – a caller is placed on a call queue;

- dequeued – a caller is removed from the call queue.

If a call status matches the status specified in **state_callback_events**, the application receives a POST request from PortaSwitch® with the status event details.

Benefits

- Easier implementation for external application developers.
- Service providers can implement click-to-call service in a shorter time.